# Session 5aAA

## Architectural Acoustics: General Topics in Architectural Acoustics

Vincent Valeau, Chair
*Univ. of La Rochelle, LEPTAB, Av. M. Crepeau, La Rochelle 17042, Cedex 01, France*

### *Contributed Papers*

**8:30**

**5aAA1. An investigation of Columbia College Chicago reverberation chamber.** Ryan P. Shannon, Kevin A. Fowler, Jeremy J. Hill, Sean D. Terry (Dept. of Audio Arts & Acoust., Columbia College Chicago, 33 E. Congress Pkwy Ste 601, Chicago, IL 60605, ryanshannon8@hotmail.com), Melinda J. Carney, and Dominique J. Cheenne (Columbia College Chicago, Chicago, IL 60605)

The Audio Arts and Acoustics department at Columbia College in Chicago acquired a new building in 2003. The facility, a former bank, contained an old steel and concrete vault that was converted to a reverberation chamber. The acoustic properties of the space, including reverberation time, modal density, and early reflection maps were analyzed and compared to a computer model. Reflectograms were predicted at various locations and compared to test data acquired with Time Delay Spectrometry (TDS). Polar Energy Time (PET) techniques were also used to identify the location of a single $4\times4$ sample of foam absorber and the results of the test were compared to the predicted value from the computer model. The results of the tests show that, under its current configuration, the room is usable as a reverberation chamber down to 300 Hz, and that the computer model was able to accurately predict the results from the PET methodology.

**8:45**

**5aAA2. Chronological analysis of architectural and acoustical indices in halls for music performance.** Youngmin Kwon and Gary W. Siebein (Architecture Technol. Res. Ctr., School of Architecture, Univ. of Florida, 231 Arch, P.O. Box 115702, Gainesville, FL 32611)

The objective of this study was to identify the changes in architectural and acoustical indices in halls for music performance built in the 18th through the 20th centuries. Seventy one halls were classified in five specific periods from the classical period (1750–1820) to the last two-decade contemporary period (1981–2000) based on chronology in music and architectural acoustics. Architectural indices such as room shape, seating capacity, room volume, balcony configuration, and the like as well as acoustical indices such as RT, EDT, G, C80, IACC, and the like for the halls found in the literature were chronologically tabulated and statistically analyzed to identify trends in architectural and acoustical design for each of the historical periods identified. For example, seating capacity increased from the eighteenth through the early twentieth century. It has decreased in the twentieth century while the room volume per seat has increased. This results in longer occupied reverberation times, higher G values and lower C80 values for contemporary halls compared to those built earlier in the century. Acoustical indices were in general found to be lower during the early twentieth century relative to other periods and to have increased since then.

**9:00**

**5aAA3. Acoustic design for Teatro Sesc Pinheiros.** Jose Augusto Nepomuceno and Anthony McCreery (Acustica & Sonica, Fradique Coutinho, 955 sala 01, Sao Paulo, Brasil)

Sesc is a non-governmental social organization to promote culture, sport and education in Brasil. Among its facilities there is the new Sesc-Pinheiros center with a multipurpose theater seating 1000. Due to changes in the planned use of the theater during construction, Acustica & Sonica was hired by the contractor Mendes Junior to adjust and improve the room acoustics and noise control design of the theater. The original design was modified by adding adjustable acoustical systems, changing in the ceiling and balcony shaping, and revising the specification of all finish materials. The room acoustics modifications followed the recommendations of Akustiks, acoustical consultants from South Norwalk, Connecticut, USA. A concert shell was designed and fabricated to allow the use of orchestral music. An open shell concept was used. The new hall has received excellent critics from pianists, conductors and players, and is rated among the best in the country. Acoustical measurements were performed. The measured values have been compared with the results of a computer model carried out by Acustica & Sonica for this project and they are consistent.
[The authors would like to thank Mendes Junior, Sesc, Bengt-Inge Dalenbck and Christopher Blair of Akustiks for their support.]

**9:15**

**5aAA4. Characterization and prediction of low frequency sound in workrooms.** Galen Wong and Murray Hodgson (UBC Acoust. and Noise Res. Group, 3rd Fl., 2206 East Mall, Vancouver, BC, Canada V6T 1Z3)

Low-frequency noise is of concern in industrial settings since workers are subjected to long-term exposures of low-frequency noise caused by machinery which can adversely affect human health and well-being. Passive noise-control methods are usually not cost-effective. Active noise control, on the other hand, is most effective at low frequencies. As part of a project investigating active noise control of low-frequency noise in these settings, and how to predict its benefit, measurements were performed on the propagation of low-frequency noise in a workroom, with and without fittings the obstacles in the room to understand their effects and importance in prediction models. Measurements performed in a real-world workroom, as well as in a scale model workroom show that fittings alter the sound field in an unpredictable manner. With low-frequency octave band filtered noise output, changes in the sound pressure level of 10 dB or more were measured in both cases. Prediction of low-frequency noise in workrooms was performed using an image-phase and a modal model—which do not account for the effects of fittings. Predictions were compared to measurements in full-scale and scale-model workrooms with pure tone outputs. So far it has proven difficult to accurately model low-frequency effects in large workrooms.

**9:30**

**5aAA5. Evaluation and optimization of acoustical environments in eating establishments.** Zohreh Razavi and Murray Hodgson (UBC Acoust. & Noise Res. Group, SOEH, 3rd Fl., 2206 East Mall, Vancouver, BC, Canada V6T 1Z3)

Complaints from hard-of-hearing faculty members at UBC about communicating verbally in one of the upscale restaurants on the campus led to a study of acoustical environments in eating establishments and how to optimize them. Speech intelligibility and speech privacy are important considerations in the design of eating establishments; however, they are commonly overlooked. In this preliminary research, three bistros were considered. Physical measurements were done and associated acceptability criteria applied to evaluate the environments. The noise exposures of em-

ployees and customers were measured and compared to maximum permitted occupational limits. Worker noise exposures were in the range 80–84 dBA. Customers were often exposed to levels over 75 dBA; especially at lunchtime it exceeded 80 dBA. The CATT room-acoustical prediction model was used to study the physical and acoustical factors that affect the acoustical environments in eating establishments and how optimize them. The effect of facility modifications, including the use of sound-absorbing wall panels, lowered and improved acoustical ceilings, and partial barriers between tables were predicted.

**9:45**

**5aAA6. A new method for measuring the speech security of meeting rooms.** John S. Bradley and Bradford N. Gover (Inst. for Res. in Construction, Natl. Res. Council, Montreal Rd., Ottawa, Canada K1A 0R6)

This paper describes a proposed new method for measuring the speech security of offices and meeting rooms. It is intended to assess the risk of someone who is talking in the meeting room being overheard at points outside the room, close to the outside boundaries of the room. Previous work has derived frequency-weighted speech-to-noise type measures that are well related to the perceived intelligibility and audibility of the transmitted speech. The proposed new method measures the attenuation between average levels in the meeting room and received levels at points 0.25 m from the outside boundaries of the meeting room in adjacent spaces. The average source room levels are representative of all possible talker location. The receiving points, 0.25 m from the room boundaries, can assess expected privacy at critical listening locations with minimal effect of the acoustical properties of the adjacent spaces. New measurements in 11 meeting rooms have been used to evaluate the influence of several parameters on the accuracy of the measured degree of speech security. These include: the type and number of sound sources used, and the number and location of microphones in the meeting room and the adjacent spaces. Details of the analyses will be presented.

**10:00–10:15 Break**

**10:15**

**5aAA7. A simplified method to estimate the free path length variance.** Yan Zhang, Godfried Augenbroe, Ruchi Choudhary (College of Architecture), and Brani Vidakovic (Georgia Inst. of Technol., Atlanta, GA 30332)

Kutruff derived an analytical solution for the prediction of reverberation time in spaces with arbitrary shape. It is the most popular analytical equation that contains shape factors. However, in this equation the free path length variance remains as unknown parameter. Kutruff suggested the implementation of Monte-Carlo simulation to determine this value for each unique shape. In practice this method is not significantly simpler than a full ray tracing simulation. Fortunately, using methods from the probability field, the free path length variance does have an analytical solution, but this takes such a complicated form that it is not convenient to use. This article treats the development of a simplified method to estimate the free path length variance without losing accuracy when applied to concert halls. A simplified regression model is developed for rectangular shapes based on the analytical solution. This simple model explains 99.8% variance for any rectangular shape up to 1:10:10. Secondly, for arbitrary shapes, a simplified model is proposed by including several additional simple variables. This model is validated against simulation results. It will be shown that this simplified method not only can be used in Kutruff's equation, but also reveals the significant shape-related parameters that influence the reverberation time.

**10:30**

**5aAA8. Acoustic modeling in rectangular rooms with impedance using the finite element method with the Dirichlet-to-Neumann map.** Yusuke Naka (Dept. of Aerosp. and Mech. Eng., Boston Univ., 677 Beacon St., Boston, MA 02215, ynaka@bu.edu), Assad A. Oberai and Barbara G. Shinn-Cunningham (Boston Univ., Boston, MA 02215)

Computational models of head-related transfer functions (HRTFs) are useful in investigating the effects of echoes and reverberations in enclosures. These models may be computed at relatively low cost by geometric methods, such as the image source method. However, geometric methods typically ignore several important physical effects, such as diffraction, which effect the fidelity of the resulting HRTF. On the other hand, methods based on solving the wave equation, such as the finite element method, include these effects but tend to be computationally expensive. This study represents a Dirichlet-to-Neumann (DtN) map which significantly reduces the costs associated with using the the finite element method for computing HRTFs in a rectangular room, by analytically eliminating empty regions of the room. The DtN map for rooms with realistic impedance boundary conditions is developed. This work represents an extension of our previous approach for sound-hard rooms [Y. Naka, A.A. Oberai, and B.G. Shinn-Cunningham, Proc. 18th International Congress on Acoustics, Vol. IV, pp. 2477–2480 (2004)]. [Work supported by AFOSR.]

**10:45**

**5aAA9. Analysis of diffuse broadband sound fields in enclosures by decomposition in powers of an absorption parameter.** Donald B. Bliss, Jerry W. Rouse, and Linda P. Franzoni (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708, dbb@duke.edu)

A novel analysis for steady-state or time-dependent broadband diffuse sound fields in enclosures is developed. Beginning with a formulation utilizing broadband intensity boundary sources, the strength of these wall sources is expanded in a series in powers of an absorption parameter, thereby giving a separate boundary integral problem for each power. The first problem has a uniform interior field level proportional to the reciprocal of the absorption parameter, as expected. The second problem gives a mean-square pressure distribution that is independent of the absorption parameter and is primarily responsible for the spatial variation of the reverberant field. This problem depends on the location of sources and the spatial distribution of absorption, but not absorption level. Additional problems proceed at integer powers of the absorption parameter, but are essentially higher order corrections to the spatial variation. The important second problem and the higher order problems are easily solved numerically, but closed form approximate solutions based on one iteration step are simple and accurate. Solutions obtained by retaining only the first couple of terms in the series expansion are compared to complete solutions obtained by another approach, and good agreement is shown between the two methods.

**11:00**

**5aAA10. Finite-sized sources in the presence of boundaries.** John R. MacGillivray (Red Tail Hawk Corp., 111 E. Seneca St., Ithaca, NY 14850) and Victor W. Sparrow (Penn State, University Park, PA 16802)

Many methods of auralization convolve a source signal (e.g., cello recorded in an anechoic room) with a room's impulse response (which has been computed using method of images, ray tracing, etc.). Many instruments are finite-sized sources because they produce music having frequencies where the product of the wavenumber and the instrument's characteristic length is not small. Sound produced by a finite-sized source in the presence of boundaries can include scattering and diffraction, resulting from the presence of the source in its own field. These effects are not accounted for by the auralization types mentioned above. A geometrically simple example of a finite-sized pulsating sphere in the presence of a rigid infinite boundary is solved using the translational addition theorem for spherical wave functions (TATSWF). Using TATSWF, the original problem is solved by replacing the rigid infinite wall with an image of the finite-sized sphere. This is a surprisingly complicated problem to solve,

given the simple geometry, and serves to illustrate how a source can perturb its field when near a boundary. Examples are presented for which significant changes in the pressure magnitude occur. [Work supported by the Applied Research Laboratory, Penn State.]

**11:15**

**5aAA11. Subjective impression of differences in realism, source width, and orientation between auralizations created from multichannel anechoic recordings.** Michelle C. Vigeant, Lily M. Wang (Architectural Eng. Prog., Univ. of Nebraska–Lincoln, Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0681, mvigeant@unlnotes.unl.edu), and Jens Holger Rindel (Tech. Univ. of Denmark, DK-2800 Kgs. Lyngby, Denmark)

Auralizations can be very useful in the design of performing arts spaces. One of the fundamental modeling inputs to create auralizations is the source directivity. Standard methods involve inputting the measured source directivity, calculating the impulse response (IR) and convolving it with a single channel anechoic recording. This paper focuses on an alternative method of modeling source directivity which involves multichannel anechoic recordings to create auralizations. Subjective tests were conducted comparing auralizations made with one, four and thirteen channels for differences in realism and source width. Auralizations were made using three different types of musical instruments: woodwinds (flute), brass (trombone) and strings (violin). Subjects were asked to rate each musical track on a seven-point scale for the degree of realism and source width. An analysis of variance (ANOVA) was carried out to determine the differences between the number of channels and the effect of instrument. A second test was conducted to assess the degree of difficulty in detecting source orientation (facing the audience or facing the stage wall) depending on the number of channels (one, four or thirteen) and the amount of absorption in the room. [Work supported by the National Science Foundation.]

**11:30**

**5aAA12. Sampling methods for decay time evaluation in acoustically coupled spaces.** Tomislav Jasa, Ning Xiang, and Mendel Kleiner (School of Architecture and Dept. of Electrical, Computer, and Systems Eng., Rensselaer Polytechnic Inst., Troy, NY 12180)

This paper applies the methods of Bayesian inference to the estimation of decay times in coupled rooms. Previous papers [N. Xiang and P. M. Goggans, J. Acoust. Soc. Am. **110**, 1415–1424 (2001); **113**, 2685–2697 (2003)] developed a solution method to estimate the decay times and the number of decay modes in terms of measured Schroeder's decay functions. This paper extends the previous work by using statistical sampling methods to efficiently determine the decay times along with error estimates and evaluate the ''Bayesian evidence'' term used in determining the number of decay modes. This paper discusses the implemented methods together with the previous work to solve the problem of decay time estimation as well as determining the number of decay modes in acoustically coupled rooms.

**11:45**

**5aAA13. Experimental validation of a diffusion equation-based modeling of the sound field in coupled rooms.** Alexis Billon, Vincent Valeau (LEPTAB Univ. of La Rochelle, Av. M. Crepeau 17042, La Rochelle Cedex 01, France, abillon@univ-lr.fr), Judicael Picaut (LCPC Nantes ESAR Rte. de Bouaye–BP 4129 44341, Bouguenais Cedex, France), and Anas Sakout (LEPTAB, La Rochelle, France)

Sound modeling in coupled rooms (i.e., two acoustically coupled rooms separated by an open area) has attracted considerable attention in the past. However accurate and operational models are still needed, principally when three or more rooms are coupled. In recent papers, a diffusion equation-based model has been applied to unusual room shapes. For the coupled rooms geometry, this diffusion model has been validated successfully by comparison with the classical statistical theory in a parametrical study of the coupling parameters [Billon *et al.*, J. Acoust. Soc. Am. **116**, 2553 (2004)]. In the present work, the diffusion model results are validated by means of a comparison with experimental results, both in terms of sound attenuation and reverberation time. A comparison is also provided with results given by the statistical theory and a ray tracing program. For this purpose, experiments have been conducted in two coupled classrooms with two different sound source locations. The results show a very good agreement between the diffusion model and the experiments. Conversely, the statistical model is not valid for modeling accurately the sound field distribution and decay in both coupled rooms. At last, the diffusion model runs much faster than the ray tracing program.

## Session 5aAB

# Animal Bioacoustics and ASA Committee on Standards: Behavioral Audiometric Methods in Animal Bioacoustics: The Search for Standards I

Edward J. Walsh, Cochair
*Boys Town National Research Hospital, Omaha, NB 68131*

Ann. E. Bowles, Cochair
*Hubbs-Sea World Research Inst., 2595 Ingraham St., San Diego, CA 92109*

**Chair's Introduction—8:50**

### *Invited Papers*

**9:00**

**5aAB1. Common hearing functions among mammals and birds.** Richard R. Fay (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626, rfay@luc.edu)

Among many mammals and birds, frequency-place cochlear maps have a similar form and differ primarily with respect to the length of the cochlea or basilar papilla, and with respect to the frequency range of hearing. This is remarkable, especially given the differences in anatomical organization and evolutionary history between mammals and birds. This is a striking example of convergent or parallel evolution. With few exceptions, the cochleae of mammals and birds are indeed scale models of one another. One of the primary functions of the frequency-place map is in frequency analysis. Whether this function is defined by frequency discrimination thresholds, critical masking ratios, critical bands or auditory filters in general, frequency analytic performance, behaviorally-defined, can be laid out on the basilar membrane and a critical basilar membrane distance can be associated with a given performance criterion. In general, these critical distances differ according to the function (e.g., frequency delta *F*s versus critical bandwidths), but are remarkably similar functions of frequency within and between species (they tend to be constant). It seems likely that evolution has selected the consequences of frequency analytic mechanisms for perception rather than the mechanisms themselves. [Work supported by NIH.]

**9:20**

**5aAB2. How do you know what an animal can hear?** Henry E. Heffner (Dept. of Psych., Univ. of Toledo, Toledo, OH 43606)

The comparative study of hearing is based on the ability to obtain valid behavioral measures of hearing in different species. This requires careful generation and measurement of sound, a behavioral task and reinforcer appropriate for the species, and a comparable definition of threshold. For audiograms, it is important to generate artifact-free pure tones and test with the animal's head fixed in a free-field sound field. For sound localization, a two-point discrimination with the animals head fixed in the sound field is commonly used. For all discriminations, it is important to obtain data for stimuli that are below threshold to ensure that an animal is not using extraneous cues. Descriptions of techniques should be sufficiently detailed to allow other researchers to replicate them; new techniques can be calibrated by replicating the results of others. Behavioral procedures that work well with one species may not elicit optimal performance from others. Although performance should be corrected for false positives, signal detection measures are difficult to interpret and cannot compensate for the failure to keep an animal under stimulus control. In general, valid results are obtained by following good scientific practice.

**9:40**

**5aAB3. The critical band and the critical ratio.** William A. Yost and William Shofner (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626)

The critical band is arguably one of the most important psychophysical descriptors of auditory function. The critical ratio is often used in animal psychophysical research to estimate critical bandwidth because it is easier to obtain than estimates of critical bandwidth. However, in most cases the underlying assumptions required to use the critical ratio as an estimate of critical bandwidth are rarely met. The critical bandwidths for chinchilla and guinea pigs are similar to those obtained for human subjects, while the critical ratios are considerably higher, an observation that violates the use of the critical ratio as an estimate of critical bandwidth. This paper will describe the assumptions necessary for the critical ratio to estimate critical bandwidth. We will describe data on critical bands and critical ratios for chinchilla and guinea pigs, and we will provide evidence that the chinchilla appears to be a wideband processor of the signal in critical-ratio experiments. The inability of the chinchilla to process the signal in its narrow critical band in the critical-ratio psychophysical procedure leads to the large critical ratios. We recommend that the critical ratio not be used to estimate critical bandwidth. [Work supported by NIDCD grants to Dr. Yost and to Dr. Shofner.]

10:00–10:10 Break

10:10

**5aAB4. Estimating loudness in animals.** Micheal L. Dent (Dept. of Psych., 361 Park Hall, Univ. at Buffalo-SUNY, Buffalo, NY 14260, mdent@buffalo.edu)

Although auditory acuity experiments in animals are valuable, this is not how animals typically encounter real world sounds. It is also important to know how animals respond to stimuli at more ''realistic'' intensities. The perception of above-threshold level sounds has been well documented in humans, but is much less studied in animals, where acoustic communication is important for survival. In humans, equal loudness contours are used to show that loudness does not increase with intensity equally at all frequencies. Loudness is a subjective experience, however, and much harder to measure in animals. Reaction time is one untrained response that can be used to gauge how different or similar two stimuli are: high intensities yield short and low intensities yield long reaction times. Comparing reaction times across frequencies (equal latency contours) in birds and mammals show results that are similar to equal loudness contours obtained in humans. At low SPLs equal-latency contours closely parallel threshold curves, while at high SPLs the contours flatten and all frequencies are perceived as being about the same loudness. These results in animals should be considered when thinking about aspects of acoustic communication such as sound transmission, vocal signals designs, and sound-attenuating properties of the environment.

10:30

**5aAB5. Animal behavioral psychoacoustics: Issues related to methodology and interpretation.** David Kastak, Ronald Schusterman, Brandon Southall, Marla Holt (UCSC Long Marine Lab., Santa Cruz, CA 95060), and Colleen Kastak (UCSC Long Marine Lab., Santa Cruz, CA 95060)

A brief survey of the literature in animal behavioral psychophysics shows that researchers use numerous methods to obtain information on sound detection, discrimination, and identification. Behavioral methodology in animal psychoacoustics includes both classical and operant conditioning, go/no-go or multiple alternative forced-choice tasks, and various methods of estimating detection and discrimination thresholds. Recent emphasis on comparing data across subjects, species, and media (e.g., hearing in air versus water), as well as investigations into the effects of age, noise, and ototoxins on auditory perception highlight the need for methodological standardization. In this paper we will discuss several important issues related to behavioral audiometrics, focusing primarily on marine mammals. These issues include variability among species, individual subjects, and laboratories; experimental concerns such as time constraints; adaptive versus non-adaptive psychophysical methodology and threshold reliability; signal detection theory versus threshold models of audition and the search for unbiased estimates of auditory performance; and measurement and interpretation of subject response bias. Standards for animal psychoacoustic methodology should be sensitive to each of these factors.

10:50

**5aAB6. Body size and assessment of auditory function: A comparative conundrum.** Edward J. Walsh, JoAnn McGee (Boys Town Natl. Res. Hospital, Omaha, NE 68131), John Rosowski, and William Peake (Harvard Med. School, Boston, MA 02115)

One goal of the bioacoustics community is to compare auditory function among species representing the entire animal kingdom, including terrestrial mammals. As an alternative to behavioral measures, it is frequently necessary, and/or desirable, to assess auditory function using electrophysiological approaches. Body size is an important factor that can effect the distribution and amplitude of evoked brain potentials (EP) measured from the surface of the head and the ranges of body mass and size within *Mammalia* are extensive. Consequently, the development of comparison protocols must include consideration of factors affected by size differences, e.g., the distance between EP generators and recording electrodes and the thickness of the skull. Ultimately, these factors, along with the acoustical character of the recording environment itself, affect acquired signal-to-noise ratios (SNR). In this context it is notable that the SNR associated with large animals are reduced relative to those observed in smaller animals, making the comparison of results from one species to another complex. This procedural challenge is further complicated by the requirement to acquire data efficiently and rapidly in recording environments that are non-optimal from an acoustic perspective. These issues will be addressed by considering auditory brainstem responses in tigers, bobcats, manuls, sandcats and rodents.

11:10

**5aAB7. Psychometric data and standards for the estimation of noise exposure for animals.** Ann E. Bowles (Hubbs-SeaWorld Res. Inst., 2595 Ingraham St., San Diego, CA 92109, annb1@san.rr.com)

ASA standards are used in the estimation of noise exposure for humans. The approaches pioneered for humans can be used as a model for animal standards, but a number of differences must be considered. First, animal standards will normally be applied across multiple taxa rather than to a single, monotypic species. Thus, it will be essential to find defensible methods for generalizing across taxa. Second, samples of subjects and measurement conditions are often inadequate in animal studies (e.g., measurements on a single animal, noisy testing conditions), but may be the only data available. Therefore, standards are needed for specifying the limitations of various data sources. Third, taxa may have very different psychoacoustic capabilities (e.g., best sensitivity, temporal integration time, and loudness perception). Therefore, comparative measures will be essential. For example, while weighting functions for humans are standardized to zero at 1 kHz, there are advantages to developing weighting functions for animals with an appropriate correction for taxon-specific sensitivity. Although the differences between standards for animals and humans represent a significant challenge, the development process and research in support of animal standards will yield valuable new perspectives that could be applied to humans in the future.

# Session 5aBBa

## Biomedical Ultrasound/Bioresponse to Vibration: Hemostatis and Bleeding Detection

Tyrone M. Porter, Chair
*Univ. of Cincinnati, Biomedical Engineering, 231 Albert Sabin Way, Cincinnati, OH 45267-0761*

### *Contributed Papers*

**8:00**

**5aBBa1. Vector-Doppler ultrasound for the detection of internal bleeding.** Bryan W. Cunitz, Peter J. Kaczkowski, and Andrew A. Brayman (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

A vector Doppler (VDop) ultrasound system uses a transmitter and a spatially separated pair of receivers to measure bistatic scattering from blood. VDop has two principal advantages over color-flow Doppler in identifying internal bleeding: (1) measures flow direction, and thus absolute magnitude of flow velocity (2) does not require special orientation to detect and measure flow, thus can measure flows perpendicular to the transmitter. Our hypothesis is that real-time flow direction and magnitude can be used to detect and characterize internal bleeding. A real-time vector Doppler system has been built and tested *in vitro*. The system is capable of measuring flow magnitude and direction up to 145 cm/s at a depth of 3.6 cm at a processing rate of 10 Hz. Accuracy was measured using a calibrated moving string phantom and the system performs well within a useful range. A blood flow phantom was developed to mimic arterial flow into an open cavity as well as into tissue and replicate both pulsatile flow as well as the energy storage due to vascular elasticity. Flow signature data is gathered under conditions of normal branching flow, and vessel breach. The talk will describe the VDop system and the flow phantom and summarize results.

**8:15**

**5aBBa2. Color-Doppler guided high intensity focused ultrasound for hemorrhage control.** Vesna Zderic, Brian Rabkin, Lawrence Crum, and Shahram Vaezy (Appl. Phys. Lab. and Bioengineering, Univ. of Washington, 1013 NE 40th St, Seattle, WA 98105, vesna@u.washington.edu)

To determine efficacy of high intensity focused ultrasound (HIFU) in occlusion of pelvic vessels a 3.2 MHz HIFU transducer was synchronized with color-Doppler ultrasound imaging for real-time visualization of flow within blood vessels during HIFU therapy. HIFU was applied to pig and rabbit pelvic vessels *in vivo*, both transcutaneously and with skin removed. The *in situ* focal intensity was 4000 W/cm$^2$ on average. Vessel occlusion was confirmed by color or audio Doppler, and gross and histological observations. In rabbits, five out of 10 femoral arteries (diameter of 2 mm) were occluded after 30–60 s of HIFU application. The average blood flow reduction of 40% was observed in the remaining arteries. In pigs, out of 7 treated superficial femoral arteries (2 mm in diameter), 4 were occluded, one had 80% blood flow reduction, and 2 were patent. In addition, 3 out of 4 superficial femoral arteries, punctured with 18 gauge needle, were occluded after 60–90 s of HIFU application. Larger vessels (diameter of 4 mm) were patent after HIFU treatment. Doppler-guided HIFU has potential application in occlusion of injured pelvic vessels similar to angiographic embolization.

**8:30**

**5aBBa3. A pulsatile flow phantom for image-guided high-intensity focused ultrasound studies on blood vessels.** Robyn Greaby and Shahram Vaezy (Ctr. for Medical and Industrial Ultrasound, APL, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, rgreaby@u.washington.edu)

A pulsatile flow phantom has been developed for controlled studies of acoustic hemostasis and high intensity focused ultrasound (HIFU) effects on blood vessels. The flow phantom consists of an excised carotid artery attached to a pulsatile pump and embedded in an optically and acoustically transparent gel to create an *ex vivo* model of a human artery. The artery was punctured with a needle to simulate percutaneous vascular injury. A HIFU transducer with a focal distance of 6 cm and a frequency of 3.33 MHz was used to treat the puncture. *B*-mode and power-Doppler ultrasound were used to locate the site of the puncture, target the HIFU focus, and monitor treatment. Also, the effects of vascular flow on HIFU lesions were studied by treating vessels in the phantom for different times at a variety of flows. In both studies, histology was done on the artery. In nine trials, HIFU was able to provide complete hemostasis in $55 \pm 31$ s. It is feasible to use the flow phantom to study acoustic hemostasis of blood vessels. Histology shows that the flow in the phantom appears to diminish the vascular damage from HIFU.

**8:45**

**5aBBa4. High throughput high intensity focused ultrasound (HIFU) treatment for tissue necrosis.** Vesna Zderic, Jessica Foley, Sean Burgess, and Shahram Vaezy (Appl. Phys. Lab. and Bioengineering, Univ. of Washington, 1013 NE 40th St., Seattle WA 98105, vesna@u.washington.edu)

To increase HIFU throughput, a HIFU transducer (diameter of 7 cm, focal length of 6 cm, operating frequency of 3.4 MHz), coupled to an imaging probe (ICT 7-4, Sonosite), was driven by 100 W driving unit with 500 W amplifier. A water pillow connected to a circulation pump with degassed water provided transducer coupling and cooling. Input electrical power was 400 W, corresponding to focal intensity of 51 300 W/cm$^2$ (in water). HIFU was applied transcutaneously in porcine thigh muscle *in vivo*, and intraoperatively in liver hilum, for 30–60 s. After 30 s of treatment, muscle lesions had diameter of $2.1 \pm 0.4$ cm and length of $2.0 \pm 0.4$ cm. After 60 s of treatment, muscle lesions had diameter of $3.1 \pm 0.9$ cm and length of $3.1 \pm 1.0$ cm. In few cases of severe skin burns, the lesions were formed immediately under the skin and were shallow ($\sim 1$ cm). Liver lesions had diameter of $2.1 \pm 0.3$ cm and length of $2.3 \pm 0.6$ cm. In comparison, after the 30 s treatment with our standard HIFU device (focal intensity of 27 000 W/cm$^2$, in water) the muscle lesions had diameter of $1.3 \pm 0.6$ cm and length of $1.1 \pm 0.6$ cm. High power HIFU device can quickly coagulate large tissue volumes.

**9:00**

**5aBBa5. Image-guided high intensity focused ultrasound annular array for hemostasis and tumor treatment.** Vesna Zderic, Robert Held, Thuc Nguyen, and Shahram Vaezy (Appl. Phys. Lab. and Bioengineering, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, vesna@u.washington.edu)

To develop and characterize an ultrasound-guided high intensity focused ultrasound (HIFU) array, an 11-element annular phased array (aperture of $3.5\times6.0$ cm, focal depth of 3.0–6.0 cm, frequency of 3 MHz) was coupled to an imaging probe (C9-5, Philips). LabView software was developed to control driving electronics and image guidance. Radiation force balance measurements, Schlieren imaging, and hydrophone field mapping were performed. Lesions were produced in gel phantoms, and *ex vivo* porcine liver and human cancerous uterus. The lesions were formed beginning at a focal depth of 6.0 cm and moving by 0.5 cm increments to 3.0 cm, and vice versa, with the overall treatment time of 2 min. The transducer had efficiency of 38%, with intensities of up to 6200 W/cm$^2$ at the natural focus of 5 cm, in water. The 6 dB focal area varied from 0.4 mm$^2$ (at 3 cm) to 1.5 mm$^2$ (at 6 cm). The 3 to 6 cm tissue lesions were $2.7\pm0.5$ cm in length, compared to $4.1\pm0.3$ cm for the 6 to 3 cm lesions. The average lesion width was 1 cm. Image-guided HIFU array may enable treatment of large tumor volumes and hemorrhage control at different depths inside the body.

**9:15**

**5aBBa6. Image guided acoustic hemostasis.** Sean Burgess (Dept. of Bioengineering, Univ. of Washington, Box 355640, 1013 NE 40th St., Seattle, WA 98105, sburgess@u.washington.edu), Vesna Zderic (Univ. of Washington), and Shahram Vaezy (Univ. of Washington)

Previous studies have shown that high intensity focused ultrasound (HIFU) can successfully control visible bleeding from solid organ injuries. This study investigates the ability of ultrasound image-guided HIFU to arrest occult hemorrhaging in the posterior liver parenchyma using a pig model. The image-guided HIFU device consisted of an intraoperative imaging probe and a spherically-curved HIFU transducer with focal length of 3.5 cm, frequency of 3.23 MHz, and focal acoustic intensity of 2350 W/cm$^2$. A total of 19 incisions (14 HIFU-treated and 5 control incisions) were made in five pig livers. The incisions were 30 mm long and 7 mm deep with HIFU application occurring within 20 s of making an incision. Hemostasis was achieved in all treated incisions after a mean $\pm$ SD of $65\pm15$ s of HIFU application. The mean blood loss rate of the control incisions initially and after seven minutes was 0.268 and 0.231 mL/s, respectively. Subsequent histological analysis showed coagulative necrosis of liver tissue around the incision which appeared to be responsible for hemostasis. Ultrasound image-guided HIFU offers a promising method for achieving hemostasis in surgical settings in which the hemorrhage site is not accessible.

**9:30**

**5aBBa7. Ultrasound-guided high frequency focused ultrasound neurolysis of peripheral nerves to treat spasticity and pain.** Jessica L. Foley, Frank L. StarrIII, Carie Frantz, Shahram Vaezy (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jlf2@u.washington.edu), Jessica L. Foley, Shahram Vaezy (Univ. of Washington, Seattle, WA 98195), and James W. Little (Univ. of Washington, Seattle, WA 98195)

Spasticity, a complication of central nervous system disorders, signified by uncontrollable muscle contractions, is difficult to treat effectively. The use of ultrasound image-guided high-intensity focused ultrasound (HIFU) to target and suppress the function of the sciatic nerve of rabbits *in vivo*, as a possible treatment of spasticity and pain, will be presented. The image-guided HIFU device included a 3.2-MHz spherically-curved transducer and an intraoperative imaging probe. A focal intensity of $1480-1850$ W/cm$^2$ was effective in achieving complete conduction block in 100% of 22 nerves with HIFU treatment times of $36\pm14$ s (mean $\pm SD$). Gross examination showed blanching of the nerve at the treatment site and lesion volumes of $2.8\pm1.4$ cm$^3$ encompassing the nerve. Histological examination indicated axonal demyelination and necrosis of Schwann cells as probable mechanisms of nerve block. Long-term studies showed that HIFU intensity of 1930 W/cm$^2$, applied to 12 nerves for an average time of $10.5\pm4.9$ s, enabled nerve blocks that remained for at least 7–14 days after HIFU treatment. Histological examination showed degeneration of axons distal to the HIFU treatment site. With accurate localization and targeting of peripheral nerves using ultrasound imaging, HIFU could become a promising tool for the suppression of spasticity and pain.

**9:45**

**5aBBa8. Percutaneous high intensity focused ultrasound on pigs.** Herman du Plessis (Surgery, 1 Military Hospital, Thaba Tshwane 0143, South Africa) and Shahram Vaezy (Univ. of Washington, Seattle, WA 98195)

Two types of HIFU Handpieces were tested on pigs under general anesthesia. The direct applicator (focal distance 4 cm) was adequate to control bleeding from a liver injury, although direct pressure was effective in a shorter time. The percutaneous application did not work at all, as the fixed focal distance (6 cm) was too deep for the size pigs available, and the diagnostic modality was difficult to integrate with the therapeutic window. We were able to create necrotic lesions in the liver substance, but not in the areas where the injury was situated. More development is needed to build a better, user-friendly application device before HIFU can be used in the clinical situation for control of hemorrhage.

# Session 5aBBb

## Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Audible-Frequency Medical Diagnostic Methods, Including Multimode Techniques I

Thomas J. Royston, Cochair
*Dept. of Mechanical Engineering, Univ. of Illinois at Chicago, 842 W. Taylor St., Chicago, IL 60607-7022*

Hans Pasterkamp, Cochair
*Pediatrics Dept., Univ. of Manitoba, 840 Sherbrooke St., Winnipeg, MB R3A 1R9, Canada*

**Chair's Introduction—8:00**

*Invited Papers*

**8:05**

**5aBBb1. Audible-frequency medical diagnostic methods: Past, present and future.** Armen Sarvazyan (Artann Labs., Lambertville, NJ 08530)

From time immemorial, the medical practitioners used audible-frequency diagnostic methods (percussion, pulse tone analysis, lung, heart and abdominal sound listening) for diagnosing and treatment of ailments without much help from other tests. During the last century, despite the fact that the stethoscope became the sign of the physician, auscultatory techniques are becoming outmoded and are often considered of little clinical value in comparison with many other modern diagnostic technologies. But obviously, there is rich information on the normal and diseased states of human organs in audible-frequency signals since characteristic times for many physiologic processes and resonance frequencies of anatomical structures are in that frequency range. In recent years, numerous emerging technologies started to bring new life into the ancient diagnostic approaches using low-frequency signals. Significant part of these new technologies are based on ultrasonic methods such as remote detection of tissue oscillation using the Doppler technique and especially on emerging ultrasonic technologies based on the use of acoustic radiation force. This talk presents an overview of current trends in the development of methods realizing the tremendous untapped medical diagnostic potential of sonic range acoustics.

**8:30**

**5aBBb2. *In vivo* breast vibro-acoustography.** Mostafa Fatemi, Azra Alizad, Dana H. Whaley, and James F. Greenleaf (Mayo Clinic College of Medicine, 200 First Ave. SW, Rochester, MN 55905)

Vibro-acoustography is based on audio-frequency harmonic vibrations induced in the object by the radiation force of focused ultrasound. The resulting object vibration produces a hydrophone detectable acoustic emission that is a function of regional dynamic properties of the object at the vibration frequency. The amplitude of the hydrophone signal is mapped into an image that represents the object's vibro-acoustic response to the radiation force by scanning the focused ultrasound across the object. Contrast in Vibro-acoustography images represent several properties of the object, including the dynamic characteristics at the audio frequency and scattering and absorption properties at the ultrasound frequency. A Vibro-acoustography system has been combined with a stereo-tactic x-ray mammography system for *in vivo* breast imaging and has been tested on a number of volunteers. Resulting images show soft tissue structures and microcalcifications within breast with high contrast, high resolution, and no speckles. The results have been verified using x-ray mammograms of the breast. The encouraging results from *in vitro* and *in vivo* experiments suggest further development of vibro-acoustography may lead to a new clinical breast imaging modality. [Work supported in part by a grant from the Susan G. Komen Breast Cancer Foundation. Vibro-acoustography is patented by MF and JFG.]

**8:55**

**5aBBb3. Supersonic shear imaging.** Mathias Fink, Mickael Tanter, and Jeremy Bercoff (Laboratoire Ondes et Acoustique, ESPCI, 10 rue Vauquelin, 75005, Paris, France)

There is a great interest for developing in vivo quantitative imaging of the shear tissue properties. This can be achieved by using an ultrafast ultrasonic imaging device able to follow in real time the propagation of low frequency transient shear wave in the body. We have developed such an ultrafast scanner that can give up to 5.000 images/s. From the spatio-temporal evolution of the elastic field a local inversion algorithm allows to recover the shear modulus and viscosity map without the limitations induced by diffraction. Transient shear sources are induced by ultrasonic radiation pressure generated by the ultrasonic probe array. The most interesting configuration that induces high amplitude shear waves used a sonic shear source that moves at supersonic velocities. *In vitro* and *in vivo* results will be presented that demonstrate the great interest of supersonic shear imaging.

**9:20**

**5aBBb4. Shear wave interferometry, an application of sonoelastography.** Clark Z. Wu and Kevin J. Parker (Hopeman 204 ECE Dept., Univ. of Rochester, Rochester, NY 14623, wuzhe@ece.rochester.edu)

Sonoelastography is an ultrasound imaging technique where low-amplitude, low-frequency (LF) vibration is detected and displayed via real-time Doppler techniques. When multiple coherent shear wave sources exist, shear wave interference patterns appear. Two shear wave sources at the same frequency create hyperbolic shaped interference patterns in homogeneous, isotropic elastic media. Shear wave speed can be estimated from the fringe separation and the source frequency. If the two sources are driven at slightly different sinusoidal frequencies, the interference patterns no longer remain stationary. It is proven that the apparent velocity of the fringes is approximately proportional to the local shear wave velocity. With this approach, local shear wave speed in elastic media can be estimated. In addition, with a single shear wave source at frequency $f$ and the ultrasound probe externally vibrated at frequency $f-\Delta f$, a novel type of moving interference between the shear waves and the frame of reference motion is created. The moving interference fringes represent the shape of shear wave wavefronts while traveling at a much slower speed. This approach provides a real-time visualization of shear wave propagation and local wave speed estimation from which local stiffness is inferred. [Work supported by NIH.]

**9:45**

**5aBBb5. Magnetic resonance elastography.** Richard Ehman and Armando Manduca (Depts. of Radiol. and Bioengineering, Mayo Clinic, Rochester, MN 55905)

The goal of our research is to develop MRI-based methods for assessing the mechanical properties of tissues *in vivo*. We have focused on a novel MRI technique for visualizing propagating acoustic shear waves [Science **269**, 1854–1857 (1995)]. Suitable dynamic shear stress for Magnetic Resonance Elastography (MRE) can be generated by surface drivers, inertial effects, acoustic radiation pressure, or endogenous physiologic mechanisms. The MRE acquisition sequence is capable of visualizing cyclic tissue motion of less than 1 micron in displacement amplitude, with imaging times ranging from 100 ms to several minutes. Inversion algorithms based on continuum mechanics are used to process the acquired data to generate maps of mechanical properties such as depict stiffness, viscosity, attenuation, and anisotropic behavior. We have applied MRE to assess specimens of a variety of tissues, ranging in stiffness from lung to cartilage. Human studies have demonstrated that it is feasible to apply MRE to quantitatively image the mechanical properties of skeletal muscles, gray and white matter in the brain, thyroid, kidney, liver, and skin. Our preliminary clinical studies have to date applied MRE to observe changes in tissue mechanical properties in patients with breast, brain, and thyroid tumors, liver fibrosis, and diffuse diseases of skeletal muscle.

**10:10–10:25 Break**

## *Contributed Papers*

**10:25**

**5aBBb6. Microscopic dynamic magnetic resonance elastography.** Shadi F. Othman, Thomas J. Royston, and Richard L. Magin (Univ. of Illinois at Chicago, 842 W. Taylor St. MC 251, Chicago, IL 60607)

Microscopic magnetic resonance elastography (uMRE) is a high resolution imaging technique for measuring the viscoelastic properties of small synthetic and biological samples. Mechanical shear waves, typically with amplitudes of less than 100 $\mu$m and frequencies of 500–600 Hz, are induced using a piezoelectric oscillator directly coupled to the region of interest. By using multiple phase offsets and motion encoding gradients we acquire data that allows the generation of images that depict shear wave motion and the calculation of local values of the tissue viscoelastic properties. Recent MRE investigations are increasingly being conducted at higher spatial resolution to establish histological correlations between elasticity maps and tissue structures; such microscopic MRE studies require stronger static fields, stronger magnetic field gradients, higher performance RF coils, and more compact, higher frequency mechanical actuators. Microscopic MRE experiments were conducted at 11.74 T in a 54 mm diameter vertical bore magnet with a 10 mm diameter$\times$25 mm length cylindrical space available for imaging. The field-of-view ranged from 4 to 14 mm. The study was conducted on agarose gel phantoms of different concentrations ranging from 025%–1% w. Different biological samples, including frog oocytes and tissue engineered constructs, were also tested.

**10:40**

**5aBBb7. Coupled vibration and sound radiation from a fluid-filled and submerged or embedded vascular tube with internal turbulent flow due to a constriction.** Yigit Yazicioglu, Thomas J. Royston, Todd Spohnholtz, Bryn Martin (Univ. of Illinois at Chicago, 842 W. Taylor St. MC 251, Chicago, IL 60607), Francis Loth (Univ. of Illinois at Chicago, Chicago, IL 60607), and Hisham Bassiouny (Univ. of Chicago, Chicago, IL 60637)

The vibration of a thin-walled cylindrical, compliant viscoelastic tube with internal flow and an axisymmetric constriction that results in turbulent fluid flow is studied theoretically and experimentally. Vibration of the tube is considered with internal fluid-coupling only and with coupling to internal flowing fluid and external stagnant fluid or external tissue-like viscoelastic material. The theoretical analysis includes the adaptation of a model for turbulence in the internal fluid and its vibratory excitation of and interaction with the tube wall and surrounding viscoelastic medium. Theoretical predictions are compared with experimental measurements conducted on a flow model system using laser Doppler vibrometry to measure tube vibration and the vibration of the surrounding viscoelastic medium, as well as miniature hydrophones to measure fluid pressure in the tube. Discrepancies between theory and experiment and the coupled nature of the fluid-structure interaction are highlighted. This study is relevant to and may provide further incite into vascular patency and mechanisms of

failure, as well as diagnostics via noninvasive acoustic measurements. [Work supported by NIH EB002511 and HL55296, and Whitaker Foundation BME RG 01-0198.]

**10:55**

**5aBBb8. A multimode sonic and ultrasonic diagnostic imaging system with application to peripheral vascular characterization.** Todd Spohnholtz, Thomas J. Royston, Yigit Yazicioglu, Bryn Martin (Univ. of Illinois at Chicago, 842 W. Taylor St. MC 251, Chicago, IL 60607), Francis Loth (Univ. of Illinois at Chicago, Chicago, IL 60607), and Hisham Bassiouny (Univ. of Chicago, Chicago, IL 60637)

Ultrasound (US) medical imaging technology is enhanced by integrating a simultaneous noninvasive audible frequency measurement of biological sounds that could be indicative of pathology. Measurement of naturally-occurring biological acoustic phenomena can augment conventional imaging technology by providing unique information about material structure and system function. Sonic phenomena of diagnostic value are associated with a wide range of biological functions, such as breath sounds, bowel sounds, and vascular bruits. The initial focus of this multimode technology was to provide an improved diagnostic tool for common peripheral vascular complications that result in transitional or turbulent blood flow, such as associated with arteriovenous (AV) grafts and stenoses in common carotid and other arteries due to plaque buildup. We review: (1) the development the multimode system by combining a commercial US system with a novel sonic sensor array and associated instrumentation, and (2) the evaluation of its capability via controlled phantom models of basic subsurface sound sources/structures, as well as simulations of constricted peripheral blood vessels. [Work supported by NIH EB002511 and HL55296, and Whitaker Foundation BME RG 01-0198.]

**11:10**

**5aBBb9. Characterization of a vibro-acoustography system designed to detect kidney stones during lithotripsy.** Neil R. Owen, Michael R. Bailey, and Lawrence A. Crum (Appl. Phys. Lab., University of Washington, 1013 NE 40th St., Seattle, WA 98105)

Acoustic properties of a vibro-acoustography system designed to detect kidney stones were measured. Our system was formed with two spherical transducers (10 cm diameter, 20 cm curvature) in degassed water that were confocal and separated by an angle of 30 degrees. They were driven at 1.1 MHz and 1.125 MHz to generate a difference frequency of 25 kHz. The acoustic field was characterized by scattering from a known target, the curved surface of a steel cylinder with 6.4 mm diameter. Waveforms of both the low and high frequency scattered signals were measured for different target locations, different hydrophone locations encircling the target, and different acoustic pressures. Focal dimensions of the $-6$ db pressure profile measured at 25 kHz and the fundamental were both $3 \times 10$ mm, in an elliptical shape, which is highly localized. Scatter amplitude was rather insensitive to hydrophone position when the target was in the focus, quite sensitive to hydrophone position when the target was out of the focus, and increased linearly with the sum of the sources. It is hoped that this characterization will help improve the understanding of the mechanisms of the targeting technique. [Work supported by NIH grants DK43881 and DK55674, and NSBRI grant SMS00203.]

**11:25**

**5aBBb10. 3D steady-state ultrasound-elastography.** Ralph Sinkus, Jeremy Bercoff, Mickael Tanter, and Mathias Fink (Laboratoire Ondes et Acoustique, 10 rue Vauquelin, 75005 Paris, France)

One of the current main limitations of Elastography is the lack of access to the full displacement field within a volume. Standard ultrasound techniques provide good motion estimation only along the ultrasound beam. These data are subsequently used for the reconstruction of elastic parameters assuming that the missing displacement components are negligible. Although feasible under certain circumstances, these assumptions typically do not hold for in-vivo applications. Thus, there is need for a technique to assess the entire displacement field in 3D. Recently, the 1D motion estimation has been extended to the measurement of the 2D displacement field [Tanter *et al.*, IEEE Trans. Ultrason. Ferroelectr. Freq. Control **49**, 1363–1374 (2002)]. This method is utilized for the assessment of the 3D displacement field by rotating and translating the US-transducer around the object of interest. Low-frequency mechanical vibrations (approx. 80 Hz) are coupled into the object from underneath (i.e., from the direction perpendicular to the plane of rotation). The measured displacement field is used to reconstruct within a volume shear elasticity and shear viscosity by local inversion of the wave equation. Contributions from the compressional wave are removed via application of the curl-operator. In-vitro results for phantoms and excised specimen are presented.

**11:40**

**5aBBb11. Feasibility of shear-mode transcranial ultrasound imaging.** Greg T. Clement, P. Jason White, and Kullervo Hynynen (Dept. of Radiol., Harvard Med. School, Brigham and Women's Hospital, Boston, MA 02115, gclement@hms.harvard.edu)

Despite ultrasound's potential to provide a low cost method for imaging blood flow and diagnosing certain brain disorders, distortion and low signal to noise ratios caused by the skull have severely limited the use of existing clinical devices, such as trancranial Doppler sonography. Presently we investigate the potential to propagate ultrasound through the skull with reduced distortion and higher signal amplitudes by using high incident angles. In such cases the ultrasound angle of entry is set beyond Snell's critical angle for the longitudinal pressure wave, so that propagation in the bone is purely due to a shear wave. This wave then converts back to a longitudinal acoustic wave in the brain. This conversion from a longitudinal wave (skin) to a shear wave (skull) and again to a longitudinal wave (brain) does not necessarily produce a highly distorted or small-amplitude wave. Basic images and measurements of shear speed-of-sound and attenuation values for *ex vivo* human skull bone will be presented for frequencies between 0.25 MHz and 2 MHz. Similar measurements with porcine samples will also be shown, indicating the large discrepancy between shear characteristics of the two species; the porcine samples showing no detectable shear mode.

# Session 5aMU

## Musical Acoustics: String Instrument Design and Construction

Thomas D. Rossing, Cochair
*Physics Dept., Northern Illinois Univ., De Kalb, IL 60115*

Christopher E. Waltham, Cochair
*Dept. of Physics and Astronomy, Univ. of British Columbia, 6224 Agricultural Rd., Vancouver, BC V6T 1Z1, Canada*

### *Invited Papers*

**8:15**

**5aMU1. Acoustical considerations in the design—and re-design—of the violin.** Joseph Curtin (Joseph Curtin Studios, 3493 W. Delhi, Ann Arbor, MI 48103, violins@josephcurtinstudios.com)

The violin is a highly evolved instrument which has long resisted significant changes to its design and construction. Still, acoustical research over the past several decades has shed sufficient light on the workings of the violin that makers can now consider non-traditional approaches to their craft in order to optimize the sound, playability, and consistency of their instruments. The work of researchers such as Duennwald, Haines, Hutchins, and Weinreich will be considered in terms of its usefulness as a guide to building better violins.

**8:40**

**5aMU2. The violin octet and bowed string instrument design.** George Bissinger (Phys. Dept., East Carolina Univ., Greenville, NC 27858)

Modal analyses were combined with room-averaged acoustic measurements of a complete octet to assess Schellengs fundamental scaling design assumptions: similarity of shape and flat plate scaling. The scaling employed only the two lowest strongly radiating violin resonances, the main air $A0$ and main wood comprised of the first corpus bending modes $B1^-$ and $B1^+$. $A0$ fell below the desired pitch placement (1.5× lowest string pitch), while the $B1$ complex generally straddled the desired placement at 2.25×. Difficulties in properly scaling $A0$ derived partly from an unreliable theory that failed to incorporate $A0$ coupling to $A1$ (first length-wise cavity mode), and partly from inability to incorporate cavity wall compliance. Wall compliance dropped $A1$ into main wood region even though larger instruments were designed successively shorter; the $A1/A0$ frequency ratio rose from ~1.5 to ~2.0 (smallest to largest). Modern models sensitive to cavity shape predict $A0$ and $A1$ within ~10% over the octet, ranging over ~4.5:1 in length, ~10:1 in $f$-hole area, ~3:1 in top plate thickness, and ~128:1 in volume. $A0$ radiates strongly over the octet, while surprisingly $A1$ is the dominant radiator in the main wood region for the large bass even though $A1$ was never included in the scaling.

**9:05**

**5aMU3. The acoustics of hammered dulcimers.** David R. Peterson (Dept. of Mathematics, Univ. of Central Arkansas, Conway, AR 72035, DavidP@uca.edu)

The hammered dulcimer, a stringed instrument played with two wooden hammers, probably originated in the Middle East, but it has become part of the musical culture of many countries. In the U. S., the folk revivial in the 1970's sparked renewed interest in the hammered dulcimer as a concert instrument. Today, despite some consolidation in the retail market, there are still hundreds of builders, mostly amateurs, who experiment with the basic design. The most important design parameters will be discussed from a practical and acoustical point of view: soundboard size, shape, and composition, internal bracing, bridge shape, string arrangement and composition, hardness of bridge caps, hammer weight and stiffness, instrument resonances due to the unique string splitting and stiffness of the body, and soundboard modes.

**9:30**

**5aMU4. Classical guitar construction: The acoustician's tale.** Bernard E. Richardson (School of Phys. and Astron., Cardiff Univ., 5 The Parade, Cardiff CF24 3YB, UK, RichardsonBE@cardiff.ac.uk)

The vast majority of guitars produced today are built according to general principles laid down in the nineteenth century. Nevertheless, the devil is in the detail, and innovative makers constantly reappraise the design and construction of instruments in their endeavors to control quality or to seek "improvement." The maker's approach, necessarily, tends to be pragmatic, but it is one which can be greatly informed by the application of relatively simple acoustical models. This paper will examine various important design aspects—for example choice of materials, body size, strutting, soundboard thickness—introducing the basis for making informed decisions.

**9:55**

**5aMU5. Normal modes of a single mandolin with different bracing patterns installed.** David J. Cohen (Cohen Musical Instruments, 9402 Belfort Rd., Richmond, VA 23229) and Thomas D. Rossing (Northern Illinois Univ., DeKalb, IL 60115)

The vibrational modes and sound spectra of a single archtop mandolin constructed with a removable back plate were studied with different bracing patterns installed. The bracing patterns were (i) symmetric and (ii) asymmetric parallel ''tone bars,'' (iii) X-bracing, and (iv) an asymmetric pattern previously used and studied [D. Cohen and T. D. Rossing, Can. Aeronautics Space, J. **4**(2), 48–54 (2000), D. Cohen and T. D. Rossing, Acoust. Sci. Tech. **24**, 1–6 (2003)]. The results have been compared with those obtained previously for vintage mandolins [D. Cohen and T. D. Rossing, ''Normal modes of different types of pre-1929 mandolins,'' presented at the 147th meeting of the Acoustical Society of America, New York, NY (2004)] and for mandolins constructed more recently. As was the case for the different types of vintage mandolins, changing the bracing pattern has little effect on the shapes of the normal modes, but does affect the frequencies of the normal modes. The X- or crossed bracing pattern imparts more cross-grain stiffness than the other bracing patterns, with the result that modes involving cross-grain bending occur at higher frequencies than is the case for the other bracing patterns.

**10:20–10:30 Break**

**10:30**

**5aMU6. Design aspects of the five string banjo.** James Rae (Dept. Physio. and Biomed. Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, rae.james@mayo.edu) and Thomas Rossing (Northern Illinois Univ., DeKalb, IL 60115)

The five string banjo is a stringed instrument that uses a drum head (membrane) to radiate sound. The strings couple to the head through a moveable bridge. Many have a removable resonator whose spacing from the main part of the banjo is adjustable to allow tuning of the internal air cavity. Holographic interferometry demonstrates vibrational modes of the head that have a strong dependence on head tension. Driving point accelerance measurements demonstrate that the ability of the bridge to be vibrated depends on the mass and stiffness of the materials used in its construction. Peak accelerances usually occur between 1800–2000 Hz. Power spectra measurements of the sound show that 99% of the sound is in a frequency range of 147–5200 Hz. Two substantial formants are seen in the power spectra. The first and largest occurs in about the 400–1800 Hz range, the same frequency range where the most substantial head modes are found. The second is from 2000–4000 Hz, a range where the series combination of bridge and head show increasing accelerances by driving point measurements. Measurements of the transient response following a pluck quantify the frequency content of the rising and falling phase.

**10:55**

**5aMU7. Tuning the lower resonances of carved Baltic psalteries by adjusting the areas of the sound holes.** Andres Peekna (Innovative Mech., Inc., 5908 N River Bay Rd., Waterford, WI 53185) and Thomas D. Rossing (Northern Illinois Univ., De Kalb, IL 60115)

Previous work has shown that with carved Baltic psalteries it is highly desirable to have a body-resonance with high radiating efficiency between the keynote and the tone above. In many cases, it is also desirable to have a body-resonance close to the low dominant. The frequencies of the two lowest body-resonances can be adjusted, to some extent, by adjusting the areas of the sound holes. We describe two trials. In one case, the instrument started out with no sound holes, and their areas were increased step by step, mostly according to listening trials, but the results were also verified by electronic TV holography. In the second case, the instrument started with excessive sound hole area. In this case, the listening tests first determined the extent of sound holes to be covered by sufficiently stiff material. The findings were subsequently confirmed by electronic TV holography. In both cases, a point was reached below which (smaller areas) the sound quality became less bright and more dull, and above which the sound quality was perceived as good. In each case, an optimum was reached.

**11:20**

**5aMU8. Harp design and construction.** Chris Waltham (Dept. of Phys. & Astron., Univ. of British Columbia, Vancouver, BC, Canada V6T 1Z1, waltham@physics.ubc.ca)

The harp is an instrument with a set of plucked strings connected directly to the sound board. Thus the requirement that the sound board be very light for efficient radiation is pitted against the requirement that the strings be tight enough to maintain harmonicity. These factors have determined the evolution of the harp since Sumerian times. As materials and construction methods have improved, strings have become tighter and sound boards lighter. Large harps have total tensions in the range of ten kilonewtons. The material of choice for the sound board, usually spruce, is pushed to the limits of its strength, and even professionally built concert harps tend not to last more than a few decades. The strings themselves are made from a variety of materials, wrapped or plain, to balance the physical requirements of tension and feel, with the acoustical demands of harmonicity and tone. Historical materials such as copper and silk have given way to steel and nylon, although gut still has a niche in the mid-range of a concert harp. These physics and engineering issues will be considered in the context of practical harp construction.

## *Contributed Paper*

11:45

**5aMU9. The motion of harp strings.** Gary Chan and Chris Waltham (Dept. of Phys. & Astron., Univ. of British Columbia, Vancouver, BC, Canada V6T 1Z1)

The harp is an instrument with a set of plucked strings that excite the sound board directly, without the medium of a bridge. The strings are positioned at an acute angle to the plane of the sound board. The quality of the sound produced depends on the motion of the string and its interaction with the resonances of the sound board. The string and sound board motions of small and large harps have been studied using small, fast position sensors. The results are compared to those of a simple non-linear model based on the measured elastic properties of the string materials, and those of the sound board. The implications for the sound production are discussed.

FRIDAY MORNING, 20 MAY 2005 REGENCY F, 8:30 TO 11:55 A.M.

# Session 5aNS

# Noise and Psychological and Physiological Acoustics: Workshop on Methods for Community Noise and Annoyance Evaluation I

Brigitte Schulte-Fortkamp, Cochair
*Technical Univ. Berlin, Inst. of Technical Acoustics, Secr TA 7, Einsteinufer 25, Berlin 10587 Germany*

Bennett M. Brooks, Cochair
*Brooks Acoustics Corp., 27 Hartford Turnpike, Vernon, CT 06066*

**Chair's Introduction—8:30**

## *Invited Papers*

8:35

**5aNS1. Assessing multi-source noise environments with an "equally annoying" exposure summation model.** Paul Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821)

Miedema [J. Acoust. Soc. Am. **116**, 949–957 (2004)] posits an "equally annoying" energy summation model for assessing annoyance in areas exposed to multiple differing noise sources. His "proof" is a theorem. Since his model fits the requirements of the theorem, he states that this "proves" that his model is correct. While it may be true that meeting the conditions of the theorem is a necessary condition, it is not a sufficient condition. From a psycho-acoustical standpoint, the Meidema model is very complex and not really believable. Moreover, it fails to address differences between types of aircraft, types of operation, types of motor vehicle, etc. An "equally annoying" exposure summation model, presented here, is psycho-acoustically much simpler and better satisfies the same theorem conditions because it better satisfies the independence requirement.

8:55

**5aNS2. Neural network analysis of soundscape in urban open spaces.** Lei Yu and Jian Kang (School of Architecture, Univ. of Sheffield, Western Bank, Sheffield S10 2TN, UK, j.kang@sheffield.ac.uk)

The physical and social environments of open spaces in a city influence people's behavior as well as their perception of the city. Soundscape is an important component of an open urban space, and key aspects for producing a good soundscape include its description, evaluation, and design. In this research, an artificial intelligent system is being developed for the analysis of soundscape evaluation in urban open spaces. The system considers acoustic factors including the level, type and spectrum of sound sources and the reflection effects of an urban space; users social and demographic factors; and other physical and environmental factors including urban morphology, microclimate conditions, and thermal and visual comfort. Since 2001 a large scale social survey has been carried out in England, Switzerland, Italy, Greece, Germany and China. Correspondingly, a database suitable for artificial neural networks (ANN) modeling has been established. Simulating biological brains, ANN are simplified models of the central nervous system, which have the ability to respond to input stimuli and to learn to adapt to the environment. The software Qnet is being used, and initial results suggest that there is a good convergence of using ANN to predict peoples perception of soundscape in urban open spaces.

9:15

**5aNS3. Soundscape assessment.** Osten Axelsson, Birgitta Berglund, and Mats E. Nilsson (Inst. Environ. Medicine, Karolinska Inst. and Dept. Psych., Stockholm Univ., SE-10691, Stockholm, Sweden)

In order to improve the quality of the soundscape it is necessary to know its descriptive and evaluative properties, and the relationships between these properties. This was explored in a listening experiment with 100 participants (48 women, 52 men; mean age 25.6 years). Each participant scaled 5 out of 50 soundscapes with regard to 116 single verbal attributes, using a visual analogue scale of agreeableness. In addition, acoustical properties of the soundscapes were assessed. A principal component analysis identified two major evaluative components, labeled Hedonic Tone and Eventfulness. Furthermore, it was found the mere presence of common sound sources, regardless of sound level, correlated significantly with these evaluative components. Technological sounds (e.g., traffic noise) were negatively associated with both Hedonic Tone and Eventfulness, while a positive association was found between Hedonic Tone and sounds of nature (e.g., bird song), and a positive association was found between Eventfulness and human sounds (e.g., human voices). These relationships lead to the hypothesis that introduction of nature and human sounds, in combination with the reduction of technological sounds may improve the quality of soundscapes considerably.

**9:35**

**5aNS4. Assessment of outdoor soundscapes in quiet areas.** Mats E. Nilsson and Birgitta Berglund (Inst. of Environ. Medicine, Karolinska Inst. and Dept. of Psych., Stockholm Univ., SE-106 91, Stockholm, Sweden)

Existing quiet outdoor areas should be preserved. Appropriate indicators and limit values are needed, which are grounded in knowledge on positive aspects of soundscapes, such as perceived pleasantness and psychological restoration. For this reason, a questionnaire study was conducted in four green areas close to a major city and in four city parks. Measured equivalent sound levels (LAeq, 15 min) ranged between 42 and 50 dBA in the green areas, and between 49 and 60 dBA in the city parks. Sounds from nature, such as bird song, completely dominated the soundscape in the green areas. The city-park soundscapes were more complex, containing sounds from nature, as well as technological sounds (e.g., traffic noise), and human sounds (e.g., human voices). In general, sounds from nature were perceived as pleasant, technical sounds as annoying, and human sounds as neutral. Between 84 and 100% of the visitors in the green areas assessed the soundscapes as good or very good. The corresponding percentages for the city parks were distinctly lower, between 52 and 65%. The results indicate that the equivalent sound level should be below 50 dBA in order to secure pleasant and restorative outdoor soundscapes in urban areas.

**9:55–10:15 Break**

**10:15**

**5aNS5. Techniques of analysis and their applicability in the context of community noise.** Andre Fiebig and Brigitte Schulte-Fortkamp (ITA-Tech. Univ. Berlin, Einsteinufer 25, D-10587 Berlin, Germany)

There is common consent to the necessity using and combining subjective and objective data for a sufficient understanding of human perception and evaluation of noise. Moreover, a wide discussion about methodology and techniques of analysis with respect to new methods in the context of soundscape research, community noise, and annoyance evaluation is urgently needed. The presentation at hand shows a methodological discussion of qualitative data and its systematic analysis based on explorative approaches applied in two different surveys. For example, by means of the analysis a transfer from verbal evaluations based on habitual language to acoustical descriptors is possible. Furthermore, the principals and steps of the analysis of qualitative data in the context of community noise are presented. Insofar, with regard to an effective and unitized handling, options and possibilities of methods in connection with specific techniques of analysis will be discussed.

**10:35**

**5aNS6. Correlation of airport noise complaint terminology with available noise metrics.** Nancy S. Timmerman (Consultant in Acoust. and Noise Control, 25 Upton St., Boston, MA 02118-1609)

In the discussion of airport noise, there is a disconnect between the language used by complainants and the acoustical terminology available to the technical community. It will be shown in what ways the complaint terminology can be described with the available noise metrics and in what ways it cannot. The complaint terminology analyzed comes from the author's personal (previous) experience as noise control officer at Logan International Airport. Examples of some of these are (aircraft noise is): too loud, too late, too early, unending, continuous, at the wrong time, so loud it sets off car alarms, so loud I cannot hear the TV/telephone/radio, so loud it woke me up, and annoying. Terms which are not related to noise are mentioned, but not correlated. The acoustical terminology comes from the international literature and includes the standard noise metrics as well as some more recently proposed. The airport noise metrics discussed include measures of single events (maximum level, SEL, duration, signal-to-noise) and measures of noise exposure (DNL, CNEL, Leq). It is shown that while cumulative noise metrics are useful for planning and comparison, they correlate with complaint terminology less well than single event measures.

**10:55**

**5aNS7. Explorative interviews as a tool for sound evaluation.** Stephan Paul (Lab. of Vib. and Acoust., Dept. of Mech. Eng., UFSC, CxP 476 Florianopolis, 88.040-900 SC, Brazil)

Sound and noise evaluation needs to combine physical-measurements of sound with research on the interaction of the sound and the human being. For this kind of research qualitative techniques and especially explorative interviews, with the addressed people are appropriate as they consider the human being acting within this environment. The evaluation process refers to the basic concepts of qualitative research. Different forms of interviews, how to prepare and conduct an interview and not to forget about the discipline under investigation have to be considered. This clearly shows the requirement of transdisciplinarity including knowledge from disciplines like engineering sciences, social sciences and psychology. To obtain the required information on the subject under investigation within an interview the verbal ability of the addressed people has to be considered. The possibilities of transdisciplinary work, check of verbal abilities as well as approaches for basic standardization in interviews and the need for cultural adaptation will be discussed.

**11:15**

**5aNS8. Digital identification of intrusive noise: Pilot study to digitally characterize soundscapes and intrusion.** Tim Lavallee (LPES, Inc., 14053 Lawnes Creek Rd., Smithfield, VA 23430), Robert Kull (Parsons, Norfolk, VA 23502), and Brigitte Schulte-Fortkamp (ITA/TU-Berlin, Einsteinufer 25, D-10587 Berlin, Germany)

One of the difficulties with soundscape investigation and the definition of intrusion is the need for an individual to manually identify and log specific acoustical events. Event logging can be labor intensive, costly, and difficult to standardize. If information about physical setting and in situ acoustical signatures can be used to define a given soundscape, event logging could possibly be performed digitally and intrusion could be defined based on the spectral fingerprint of a given soundscape. Two soundscapes with different settings and acoustical signatures were identified. Sound time histories and periodic third octave spectra were measured over a given period. An individual manually logged acoustical events. Independently, algorithms to identify both normal and acoustically intrusive events for the given soundscape were applied to the data. The digitally identified events were compared to the manually taken event log. An evaluation of the results will be presented.

**11:35**

**5aNS9. Evaluation of sound environment characteristics: Comparative study between objective and subjective criteria.** Francoise Chartier and Catherine Semidor (GRECO, EAPBx, Domaine de Raba F-33400 Talence, France, catherine.semidor@bordeaux.archi.fr)

The evaluation of urban sound environment quality depends on quantitative information as well as the opinion of city dwellers. In order to underline the relation between objective and subjective points of view, a comparative study was carried out. The subjective survey consisted in listening to several binaural sound recordings of very short extracts from the urban scene, called ‘‘sonoscene.’’ During these sessions, the listeners’ opinions were noted. The binaural sound recordings are performed in accordance with the ‘‘soundwalk’’ method explained in previous papers [C. Semidor, ASA 2004, ASA 2005]. The same binaural sound recordings were analyzed in form of objective criteria such as Equivalent Sound Level, Loudness, Roughness, etc. This paper deals with the comparison between some of these objective criteria and subjective judgments such as Lack or Presence of traffic noise, Lack or Presence of human activities, Spatial Dimension, Attractiveness or Annoyance, etc. These first results point out some significant correlations between Loudness and Attractiveness for example, according to the nature of the sound sources (traffic, human activity).

FRIDAY MORNING, 20 MAY 2005 REGENCY B, 8:00 TO 10:00 A.M.

## Session 5aPAa

## Physical Acoustics: Topics in Acoustic Mine Detection

Murray S. Korman, Chair
*Physics Dept., U.S. Naval Academy, Annapolis, MD 21402*

### *Contributed Papers*

**8:00**

**5aPAa1. Long-range excitation of vibrational response in landmines with Rayleigh and Love waves.** Thomas Muir (Natl. Ctr. for Physical Acoust., Univ. Mississippi, One Coliseum Dr., University, MS 38677)

An experiment was conducted in a clay soil at a range of 25 m from two types of electromagnetic shaker sources, operating in a cw pulse mode centered at 100 Hz. The sources were in co-located arrays, one vertically oriented to generate Rayleigh waves and another transversely oriented to generate Love waves. Two cylindrical landmine targets were used: an unloaded (empty) plastic shell, and a metal shell, loaded with paraffin to simulate an explosive. These target types represent extreme variations in what could conceivably be encountered. The targets were instrumented with three-axis seismometers and accelerometers, and were also viewed with a laser Doppler vibrometer (LDV). Data were acquired on both mine cases in various stages of deployment, including sitting upon the ground surface, snugged in to a tight hole in the ground, and completely buried some 5 cm beneath the surface. The responses of the mine cases to Rayleigh and Love wave excitation were measured and compared to data from a reference seismometer deployed nearby, on natural, undisturbed soil. The mine case responses to these seismic excitation fields, under a number of conditions are presented and discussed. [Work supported by the U.S. Marine Corps Systems Command.]

**8:15**

**5aPAa2. Influence of buried objects on the dynamic nonlinearity of inhomogeneous materials.** Keith Attenborough, Qin Qin (Dept. of Eng., Univ. of Hull, Hull HU6 7RX, UK), Jonathan Jefferis, and Gary Heald (DSTL, Fort Halstead, Kent, UK)

Donskoy *et al.* [J. Acoust. Soc. Am. **111**, 2705–2714 (2002)] and Korman *et al.* [J. Acoust. Soc. Am. **116**, 3354–3369 (2004)] have shown that the nearby presence of compliant buried objects results in a significant nonlinear response in the spectrum of surface vibration induced by acoustic excitation. The latter have suggested that there is a strong connection with the inherent Nonlinear Mesoscopic Elasticity (NME) of inhomogeneous materials. Laboratory experiments are reported that investigate the comparative NME of granular and fibrous materials and the variation observed between several states of a single material (sand) subject to high sound pressure levels in the frequency range 100 Hz to 2 kHz. Fiberglass and gravel are shown to have strong inherent nonlinearity whereas wet compacted sand has relatively little. In both two-tone and single tone experiments, it is shown that, the presence of buried objects may enhance or reduce the inherent NME of the embedding medium. In cases where the NME is enhanced, compliant buried objects have a greater effect than relatively stiff objects. [Work supported by DSTL, UK.]

**8:30**

**5aPAa3. Investigation of the applicability of microphones to nonlinear acoustic mine detection.** Douglas Fenneman and Brad Libbey (U.S. Army RDECOM CERDEC Night Vision and Electron. Sensors Directorate, 10221 Burbeck Rd., Fort Belvoir, VA 22060)

The feasibility of using microphones to measure intermodulation effects above a buried mine is under investigation at the U.S. Army RDECOM CERDEC Night Vision and Electronic Sensors Directorate. Acoustic mine detection techniques employ acoustic energy to excite soil and buried mines. The resultant linear and nonlinear response at the surface can then be measured using non-contacting vibrometers. Carrier signal scattering by rough surfaces, however, can limit the measurement accuracy of these vibrometers and, subsequently, successful detection of the landmine. A microphone is proposed as an alternative non-contact sensor specifically for nonlinear acoustic mine detection applications. In this scenario, distinct frequency content facilitates separation of the intermodulation effects at the surface from the acoustic excitation. Experimental results employing intermodulation effects for acoustic mine detection have been reported in the literature [Donskoy *et al.*, J. Acoust. Soc. Am. **111**, 2705–2714 (2002)]. Preliminary experimental results demonstrate the ability of microphones to sense pressure radiated from soil in the presence of realistic ground velocities. The applicability of these measurements to practical mine detection systems will also be addressed.

**8:45**

**5aPAa4. Application of time-reversal focusing in elastic wave propagation to buried object detection.** Pelham D. Norville and Waymond R. Scott, Jr. (School of Elec. and Comput. Eng., Georgia Inst. of Technol., 777 Atlantic Dr., Altanta, GA 30332-0250)

Time-reversal focusing is a technique for focusing energy both spatially and temporally to a single desired point. First studied in fluid media, additional studies have demonstrated the applicability of time-reversal focusing to solid media and elastic wave propagation. A significant difference between time-reversal and other focusing methods is time-reversal focusings immunity to the variation of wave speeds and the effects of scattering objects within the medium. In the detection of buried objects, this feature is paramount where wave speed variations and clutter are common in the region to be searched. The effectiveness of time-reversal focusing is investigated for a variety of configurations of scattering objects. Fields of uniform objects are examined as well as distributions of randomly shaped clutter, and changes in the propagation medium. Both experimental and three-dimensional numerical results are presented. In previous studies, a phenomenon of super-resolution caused by high-order scattering has been observed where focusing exceeds the diffraction limits of the excitation array. The extent of super-resolution is evaluated for the various configurations of scattering objects in an elastic medium. [Work supported by ARO.]

**9:00**

**5aPAa5. Nonlinear acoustic experiments involving landmine detection: Connections with mesoscopic elasticity and slow dynamics in geomaterials, Part II.** Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD 21402) and James M. Sabatier (Univ. of Mississippi, University, MS 38677)

In nonlinear acoustic detection, airborne sound at two primary tones, $f_1$, $f_2$ (chosen several Hz apart from resonance) insonifies the soil surface over a buried landmine, and due to soil wave interactions with the landmine, a scattered surface profile can be measured by an LDV. Profiles at $f_1$, $f_2$, $f_1-(f_2-f_1)$ and $f_2+(f_2-f_1)$ exhibit a single peak while profiles at $2f_1-(f_2-f_1)$, $f_1+f_2$ and $2f_2+(f_2-f_1)$ are attributed to higher order mode shapes. The lowest resonant frequency for a VS 1.6 plastic, inert, anti-tank landmine, buried at 3.6 cm deep is $\sim$125 Hz. The "on target" to "off target" contrast ratio, for some of the nonlinearly generated combination tones, is roughly 15–20 dB higher compared to either primary component. Near resonance, the bending (softening) of a family of increasing amplitude tuning curves, involving the surface vibration over the landmine, exhibits a linear relationship between the peak particle velocity and corresponding frequency. The tuning curves exhibit hysteresis effects. Landmine-soil vibrations exhibit similar characteristics to nonlinear mesoscopic/nanoscale effects that are observed in geomaterials like rocks or granular materials. Nonlinear mechanisms of soil and the soil interacting with the top-plate of the mine case are compared. [Work supported by U.S. Army RDECOM CERDEC, NVESD.]

**9:15**

**5aPAa6. Time-dependent finite-element model for optimizing source array element position and excitation for a seismic sonar for buried mine detection.** Anthony N. Johnson (CPT, U.S. Army, and Dept. of Mathematics, Naval Postgrad. School, Monterey, CA 93943), Clyde L. Scandrett (Naval Postgrad. School, Monterey, CA 93943), and Steven R. Baker (Naval Postgrad. School, Monterey, CA 93943)

A three-dimensional (3-D) continuum mechanics approach to the development of a time-dependent finite-element model for optimizing the position and excitation of source array elements for use in a seismic sonar to detect buried landmines is presented. Mathematical formulation of the problem consists of the coupling of a system of linear, second order, partial differential equations and related boundary conditions into one single wave equation, from which a composite elastic finite element is derived. The *hp*-adaptive finite-element kernel, ProPHLEX [Altair Engineering, Inc., McKinney, TX], is used to perform the numerical computations. The radiation characteristics of a discrete number of transient seismic sources are analyzed in a linear, isotropic, homogeneous half-space. Results for radial and vertical radiation fields, and for radiated Rayleigh wave strength will be presented for various source configurations. Particular attention will be paid to those configurations which maximize the radiation of unidirectional Rayleigh waves, while suppressing the radiation of unwanted body waves.

**9:30**

**5aPAa7. Deduction of ground impedance using level difference measurements.** Shahram Taherzadeh (Faculty of Technol., The Open Univ., Milton Keynes MK7 6AA, England) and Keith Attenborough (Univ. of Hull, Hull HU6 7RX, England)

Previously, a numerical method for deducing ground-surface impedance from measurements of complex excess attenuation spectra was reported [J. Acoust. Soc. Am. **105**, 2039–2042 (1999)]. Subsequent applications have predicted unrealistic values at low frequencies. Here, we report improved results using two-microphone, transfer-function measurements and discuss its merits as opposed to measuring excess attenuation in regards to a free field. Furthermore, effects of errors in measured parameters on the deduced impedance are discussed also.

**9:45**

**5aPAa8. Comparison of two mapping methods for computing sound propagation over irregular terrain.** Xiao Di (Appl. Res. Lab., Penn State Univ., University Park, PA 16804, xxd1@psu.edu) and Kenneth E. Gilbert (Univ. of Mississippi, University, MS 38677)

This paper compares a piecewise linear mapping and a cascaded conformal mapping for computing sound propagation over irregular terrain using the parabolic equation (PE) approximation. Although the piecewise linear mapping does not preserve the form of the underlying Helmholtz equation (i.e., it is not conformal), it does preserve the form of the narrow-angle PE. Further, it is shown that the correct impedance boundary condition at the ground surface can be closely approximated with the piecewise linear mapping. Compared to the cascaded conformal mapping, the piecewise mapping is extremely simple and easy to implement. It involves only a phase multiplication at the end of each range step. To test the accuracy of the piecewise linear mapping, it is compared with a cascaded conformal mapping for propagation over a single hill at various frequencies. Propagation predictions for more general terrain using the piecewise linear mapping are presented and discussed. [Work supported by the Federal Aviation Administration.]

# Session 5aPAb

## Physical Acoustics: Topics in Nonlinear Acoustics

David T. Blackstock, Chair
*Dept. of Mechanical Engineering, Univ. of Texas at Austin, 1 University Station, Austin, TX 78712-0292*

### *Contributed Papers*

**10:30**

**5aPAb1. Clayton H. Allen's discovery of nonlinear acoustic saturation.** David T. Blackstock (Appl. Res. Labs. and Dept. Mech. Eng., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

In 1950 Clayton H. Allen (1918–2004) discovered nonlinear acoustic saturation. His Penn State doctoral thesis, "Finite Amplitude Distortion in a Spherically Diverging Sound Wave in Air," is about experiments on intense sound produced by a 12.2-cm diameter baffled piston vibrating at 14.6 kHz. At $r=3.8$ cm (the next-to-last nearfield peak) the maximum SPL produced was 161 dB (re 20 $\mu$Pa). The farfield began at about 50 cm. Measurements were made of the fundamental and 2nd–6th harmonics out to $r=200$ cm. His received level versus source level curves at $r=200$ cm show the now familiar plateau that is reached as nonlinear effects limit the achievable SPL for a given distance at a given frequency. Clay's discovery is quite remarkable because (1) it was totally unexpected and (2) it came long before any theoretical predictions of the phenomenon were developed. Alas, however, the work was never published except in the Army Signal Corps report from Penn State, "Atmospheric Physics and Sound Propagation, Final Report for Period July 1, 1945 to May 20, 1950," and in two papers presented orally at the Penn State ASA Meeting in June 1950. As a result, Clay's remarkable work has gone unappreciated for many years. This presentation is dedicated to his memory.

**10:45**

**5aPAb2. Nonlinear enhancement and saturation phenomena in focused ultrasound beams of various geometry.** Vera A. Khokhlova, Marina S. Basova (Dept. of Acoust., Faculty of Phys., Moscow State Univ., Moscow, Russia), Michael R. Bailey, and Lawrence A. Crum (Univ. of Washington, Seattle, Washington)

The effects of nonlinear enhancement of focusing gain and saturation are studied and compared for high-intensity focused ultrasound sources with an initial Gaussian shading and uniform amplitude distribution. Simulations are performed using the Khokhlov Zabolotskaya (KZ) nonlinear parabolic equation for weakly dissipative medium in a wide range of source linear focusing gains and source pressure amplitudes, including the strongly nonlinear regime with shocks. An artificial absorption proportional to the fourth power of frequency or an asymptotic frequency-domain approach is employed in the algorithm in order to reduce the number of harmonics for accurate modeling of strongly distorted waveforms with shocks. The effect of focusing gain and amplitude shading of the source on nonlinear enhancement of acoustic energy concentration and saturation levels at the focus is discussed. It is shown that nonlinear enhancement of focusing gain is different for different values of linear gain, different spatial distributions of the source amplitude, and different parameters of acoustic field. The levels of nonlinear saturation at the focus are obtained for very high source amplitudes. The results of simulations give lower enhancement and higher saturation levels compared to the known approximate analytic predictions. [Work supported in part by NIH Fogarty and NSBRI.]

**11:00**

**5aPAb3. Numerical simulation of nonlinear phenomena using a dispersion-relation-preserving solution of the Navier-Stokes equations.** Mark S. Wochner and Anthony A. Atchley (Grad. Program in Acoust., The Penn State Univ., 217 Appl. Sci. Bldg, University Park, PA 16802)

A modified Navier-Stokes equation set that includes classical absorption and molecular relaxation is solved using a fourth-order Runge-Kutta scheme in time and a fourth-order dispersion-relation-preserving algorithm in space. The algorithm is applied to examine nonlinear phenomena such as waveform evolution of a high amplitude sine wave to old age, shock coalescence, and evolution of arbitrary time waveforms. The results are compared to analytical solutions whenever possible. The drawbacks of such an approach are discussed along with possible solutions to problems associated with the numerical method.

**11:15**

**5aPAb4. Parabolic approximation versus geometrical acoustics for describing nonlinear acoustic waves in inhomogeneous media.** Vera A. Khokhlova, Mikhail V. Averianov (Dept. of Acoust., Faculty of Phys., M. V. Lomonosov Moscow State Univ., Leninskie Gory, Moscow 119992, Russia), Robin O. Cleveland (Boston Univ., Boston, MA 02215), and Philippe Blanc-Benon (Ecole Centrale de Lyon, 69134 Ecully Cedex, France)

Propagation of intense periodic acoustic waves in inhomogeneous media is studied in the nonlinear geometrical acoustics (NGA) approximation and using nonlinear parabolic equation (NPE). Various types of 2-D inhomogeneities are considered, such as a phase screen, single Gaussian inhomogeneities, and random inhomogeneous media. Distributions of acoustic rays are obtained by numerical solution of the eikonal equation. Pressure field patterns are calculated numerically with account for nonlinear effects and diffraction using a frequency-domain algorithm for the NPE. The location of caustics and shadow zones in the ray patterns are compared with the results of the parabolic model for the areas of increased and decreased sound pressure. Both linear and nonlinear propagation is investigated in order to reveal the validity of NGA in predicting the acoustic field structure and to better understand how the combined effects of inhomogeneities, diffraction, and nonlinearity determine the overall peak and average parameters of the acoustic field. It is shown that NGA does not accurately represent all the locations of enhanced or reduced acoustic pressure even for single scattering inhomogeneities, and the discrepancies become larger for smaller size inhomogeneities and at longer distances in random inhomogeneous medium. [Work supported by CNRS, RFBR, and NIH Fogarty.]

11:30

**5aPAb5. Measurements of the rate of change of the power spectral density due to nonlinearity in one-dimensional propagation.** Lauren E. Falco, Kent L. Gee, and Anthony A. Atchley (Grad. Program in Acoust., Penn State Univ., 217 Appl. Sci. Bldg., University Park, PA 16802-5018)

The influence of nonlinear effects in the propagation of jet noise is typically characterized by examining the change in the power spectral density (PSD) of the noise as a function of propagation distance. The rate of change of the PSD is an indicator of the importance of nonlinearity. Morfey and Howell [AIAA J. **19**, 986–992 (1981)] introduced an analysis technique that has the potential to extract this information from a measurement at a single location. They develop an ensemble-averaged Burgers equation that relates the rate of change of the PSD with distance to the quantity $Q_{p^2p}$, which is the imaginary part of the cross-spectral density of the pressure and the square of the pressure. Despite its potential applicability to jet noise analysis, the physical significance and utility of $Q_{p^2p}$ have not been thoroughly studied. This work examines $Q_{p^2p}$ for the one-dimensional propagation of plane waves in a shock tube. The use of such a simple, controlled environment allows for a better understanding of the significance of $Q_{p^2p}$. [Work supported by the National Science Foundation, the Office of Naval Research, and the Strategic Environmental Research and Development Program.]

11:45

**5aPAb6. Nonlinear standing waves in shaped resonators driven by a piston.** Cheng Luo, Xiaoyang Huang, and Nam Trung Nguyen (School of Mech. and Production Eng., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore 639798)

The nonlinear standing waves in a shaped resonator driven by a piston are investigated analytically and experimentally. In the study, the resonator is an exponentially expanded horn with a piston oscillating at its large end. The air pressure waves inside the resonator are obtained by solving a 1-D nonlinear wave equation with the Galerkins method, taking into account the moving boundary due to the piston. The simulation results, in terms of the pressure waveforms, the pressure amplitudes, and the resonance frequency shift, show that the nonlinear standing waves generated by the piston have characteristics similar to those in the same resonator but under entirely shaking condition. The experiment is also conducted on an exponentially expanded resonator. The small end of the resonator is sealed and the big end is connected to a 5-inch 40 W loudspeaker, functioned as the driving piston. The resonance frequency shift is observed, and high amplitude pressures, up to $0.3\times10^5$ Pa, have been detected at the small end, which is 20 times higher than that at the big end. The experimental results agree well with the simulated values.

FRIDAY MORNING, 20 MAY 2005 REGENCY C, 8:00 A.M. TO 12:00 NOON

## Session 5aPP

## Psychological and Physiological Acoustics: Auditory Perception and Psychophysics II (Poster Session)

Neal F. Viemeister, Chair

*Dept. of Psychology, Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455*

### *Contributed Papers*

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

**5aPP1. Sensitivity to stimulus distribution characteristics in auditory categorization.** Sarah C. Sullivan (Dept. of Psych., Univ. of Texas, Austin, TX 78712-0187), Andrew J. Lotto (Boys Town Natl. Res. Hospital, Omaha, NE 68131), Elizabeth T. Newlin, and Randy L. Diehl (Univ. of Texas, Austin, TX 78712-0187)

Several experiments were performed to examine the ability of humans to categorize sounds as a function of training distribution characteristics. Participants were presented non-speech sounds randomly sampled from two overlapping distributions. The sounds consisted of 25 narrow-band noise bursts varying in center frequency from 1000–1360 Hz. Two conditions were created by varying the ratio of stimuli in each category (i.e., prior probabilities of each category), resulting in different ideal boundaries (maximizing accuracy). Participants were asked to categorize the sounds as either *A* or *B* and feedback was provided. In one experiment, prior probabilities were altered mid-session without alerting the subjects. Performance was tracked by plotting identification functions and noting boundary placement for each individual block. After only six blocks of training ($\sim$35 min), most subjects had established optimal or near-optimal boundaries. Identification function slopes were calculated and found to be steeper than training distribution slopes; suggesting that listeners established criterion-like boundaries as opposed to performing probability matching. The fact that listeners adjusted their responses in accordance to distribution changes within relatively short periods of time demonstrates the perceptual systems effectiveness at optimizing categorization based on distribution characteristics. [Work supported by NSF and NIDCD.]

**5aPP2. Hearing silent shapes: Identifying the shape of a sound occluding surface.** Ryan L. Robart and Lawrence D. Rosenblum (Univ. of California, Riverside, Riverside, CA, 92521, rosenblu@citrus.ucr.edu)

While most psychoacoustics is concerned with perception of sound sources, there is evidence that surfaces which reflect or occlude sound can also be detected and can guide behavior [e.g., M. S. Gordon and L. D. Rosenblum, J. Acoust. Soc. Am. **107**, 2851 (2000)]. While there is also evidence that listeners can hear the shape of sound reflecting surfaces [e.g., C. E. Rice, Science **155**, 655–664 (1967)], it is not known whether the shape of sound occluding surfaces can be heard. In a series of experiments, blindfolded listeners were asked to judge the shape of surfaces (of equal area) which occluded a set of loudspeakers emitting white noise. Overall, listeners were successful at this task, with some listeners showing near perfect performance. Follow-up experiments examined the acoustic information supportive of this skill. The findings suggest a type of auditory sensitivity not often considered in the psychoacoustics literature.

**5aPP3. Comparing linear regression models applied to psychophysical data.** Zhongzhou Tang, Virginia M. Richards (Dept. of Psych., Univ. of Pennsylvania, Philadelphia, PA 19104, richards@psych.upenn.edu), and Andrew Shih (Univ. of Pennsylvania, Philadelphia, PA 19104)

Relative weights for a profile analysis task were obtained using four regression/classification models; correlation coefficients, linear regression, logistic regression and probit regression. The aim of the study was to examine the impact of the choice of model on the accuracy and the efficiency with which the relative weights were determined. A yes/no task was used with observers indicating whether or not there was an increment in level to the central component of an 11-component standard. On each presentation the amplitudes of the individual components of the complex were randomly perturbed using draws from a normal distribution. When a large number of trials (1250) were used to estimate the relative weights, the four methods generated nearly identical weight estimates. When smaller numbers of trials were used (112), the different methods generated patterns of relative weights that were largely similar, and the patterns deviated only modestly from the large-number solution. In terms of efficiency, the error boundaries of the different methods were nearly indistinguishable. All in all, the number of trials needed to obtain statistically significant weights is sufficiently large that there is no apparent advantage of using one method over the others. [Work supported by NIH/NIDCD.]

**5aPP4. The difference in auditory memory between young and aged listeners with normal hearing.** Bruce Schneider, James Qi (Dept. of Psych., Univ. of Toronto at Mississauga, Mississauga, ON, Canada L5L 1C6, bschneid@credit.erin.utoronto.ca), Juan Huang, Xihong Wu, and Liang Li (Peking Univ., Beijing 100871, China)

The perceptual fusion of a noise with its delayed copies implies that there is a memory for maintaining a detailed representation of an arbitrary waveform. This auditory memory would be important for perceptually grouping correlated sounds and segregating uncorrelated sounds in noisy, reverberant environments, in which older listeners find it much more difficult than younger listeners to process signals. To determine the temporal extent of the auditory memory and whether it is affected by aging, this study investigated the detection of a break in correlation between two correlated broadband noises in younger and older listeners with normal hearing. The results show that younger listeners could detect a 100-ms break in correlation up to interaural delays ranging from 6.3 to 23.0 ms, suggesting that higher-order central mechanisms beyond the brainstem delay lines are likely to be involved in maintaining a memory trace of the fine details of the acoustic waveform. Aged listeners, however, could detect the break up to interaural delays ranging only from 6.7 to 9.7 ms, indicating the age-related decline in auditory memory. This decline may be one of the main causes leading to perceptual difficulties experienced by older listeners in noisy, reverberant environments. [Work supported by NSERCC and MSTC.]

**5aPP5. About the neglected auditory abilities within psychological tests of intelligence.** Jenny Papenbrock, Kristin Seidel, Susanne Weis, and Heinz-Martin Suess (Dept. of Psych., Univ. of Magdeburg, Pfaelzer Platz, Geb. 24, Postfach 4120, 39016 Magdeburg, Germany, si-projekt@gse-w.uni-magdeburg.de)

Auditory abilities play an important role in human intellectual abilities. In his model of human cognitive abilities Carroll (1993) identified a separate domain of auditory ability. Despite their theoretical and practical relevance, these abilities were mainly excluded in psychological tests. In order to compensate for this lack we developed a test which is intended to complement existing intelligence tests. To measure general auditory abilities already existing materials (Stankov and Horn, 1980) were used along with newly developed auditory tasks. Our test includes nonverbal tasks containing only tones as well as auditory textual tasks using spoken words. We expect that the auditory tasks measure a separate ability and therefore can make an important contribution to complement already existing intelligence tests. To demonstrate that the test is able to measure different aspects of auditory abilites pure tone tasks should be distinguishable from auditory textual tasks. 120 subjects worked on the auditory test as well as on a well-established test of intelligence (Test for the Berlin Intelligence Structure Model, BIS-4; Jger, S & Beauducel, 1997) which operates with written, numerical and figural material. Main results of our study are reported and implications for future research concerning psychological ability tests are discussed.

**5aPP6. Contributions of individual components to the overall loudness of a multi-tone complex.** Lori Leibold, Samar Khaddam, and Walt Jesteadt (555 North 30th St., Omaha, NE 68131, leiboldl@boystown.org)

The contributions of individual components to the overall loudness of a multi-tone complex were examined in a two-interval, loudness-matching task. Stimuli were five-tone complexes centered on 1000 Hz, with six different logarithmic frequency spacings, ranging from a frequency ratio of 1.012 to 1.586. Stimuli were presented for 300 ms (10 ms rise/fall). The overall level of the standard complex was fixed at 60 dB SPL (53 dB/component). Levels of the individual components of the comparison complex were selected at random from a rectangular distribution with a range of 20 dB. Ten 100-trial blocks were run for each frequency-spacing condition. Perceptual weights were computed by calculating the point-biserial correlation between the difference in the level of each component across the two intervals and the subjects response. When all components fell within the same critical band, weights varied little across components. In contrast, the range of weights increased with increasing frequency separation, with increasing weight given to the lowest and highest frequency components. The audibility of individual components was determined by measuring detection thresholds for each component in the presence of the remaining four components. A strong relation between component weight and audibility was observed for all but the narrowest spacing conditions.

**5aPP7. Differences in loudness of tone complexes with positive and negative Schroeder phase.** Manfred Mauermann and Volker Hohmann (Medizinische Physik, Institut fr Physik, Fakultt V, C.v.O Universität Oldenburg, 26129 Oldenburg, Germany)

Tone complexes with positive ($m+$) and negative Schroeder phase ($m-$) have an identical long-term spectrum, the same temporal envelope but are inverted in time. They show large differences in masking efficiency most probably being related to a different cochlear representation. The current study investigates to which extent loudness perception is affected similarly by the different phase characteristic of $m+/m-$ stimuli. Therefore, the loudness of $m+/m-$ stimuli (1.6 octave bandwidth) was matched in seven normal hearing and three hearing impaired subjects. In the first experiment the fundamental frequency $f0$ of the tone complexes was varied from 2–1536 Hz for different center frequencies at a soft level of the reference signal ($m-$). In normal hearing subjects the $m+$ stimuli need a 6 dB higher level to be perceived as equally loud as the respective $m-$ stimuli (for $f0$ in the range of 24–96 Hz). In the second experiment the difference in loudness of $m+/m-$ was investigated as function of the of the reference-stimulus level (5–80 dB SL) at a center frequency of 2660 Hz and $f0$ at 48 Hz. The largest differences in loudness were found for levels between 40–60 dB, clearly reduced for higher and lower levels of the reference stimulus.

**5aPP8. Perception of combined intensity and frequency contours by normal-hearing and hearing-impaired listeners.** Marjorie R. Leek, Michelle R. Molis (Army Audiol. & Speech Ctr., Walter Reed Army Medical Ctr., 6900 Georgia Ave, N.W., Washington, DC 20307-5001, Marjorie.Leek@na.amedd.army.mil), and Jennifer J. Lentz (Indiana Univ., Bloomington, IN 47408)

When a speaker utters a consonant-vowel or vowel-consonant syllable, the frequency transitions are critical to identification of the sound. However, an intensity contour is also superimposed on the frequency glides, as the mouth opens or closes. In this study, the interaction of frequency and intensity contours was explored using a frequency glide discrimination task. Offset frequency difference limens (offset DLs) were measured for upward and downward frequency glides in two frequency regions, and with three intensity contours, increasing from silence, decreasing to silence, and steady-state. For both normal-hearing (NH) and hearing-impaired (HI) subjects, larger offset DLs were observed for the high frequency stimuli than for the lower frequencies, and for upward-gliding stimuli than for downward frequency glides. Amplitude contour had little effect on the NH data, but did influence offset DLs for HI subjects. These findings indicate that for these stimuli, the interaction of frequency and amplitude contours plays only a small role in the perception of transition-like glides for NH listeners, but may affect the perception of frequency transitions by HI listeners. [Work supported by NIDCD.]

**5aPP9. Processing two stimuli simultaneously: Switching or sharing?** Frederick J. Gallun, Christine R. Mason, and Gerald Kidd, Jr. (Hearing Res. Ctr. and Comm. Disord., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215)

This study explored whether listeners can perform two auditory tasks simultaneously or whether they must switch rapidly between tasks. The stimuli were two simultaneous speech streams (sentences) processed into sets of mutually-exclusive narrow frequency bands. One was presented to each ear and listeners reported either the keywords from the sentences (Task 1, recognition) or simply whether or not a sentence had been presented (Task 2, detection). For both sentences and tasks, overlapping bands of noise were presented at a level that reduced performance to below 90% correct in a single-stream reference condition. Listeners were informed of which stream to report either before or after stimulus presentation. The two tasks were either the same (Task 1 at both ears) or different (Task 1 on the right, Task 2 on the left). The effect of having the opportunity to rapidly switch between sentences was examined by presenting full sentences or only the keywords. Interference occurred primarily when the tasks were the same in both ears and was greatest when the target ear was not specified in advance. Presenting only the keywords hurt single-stream performance but did not increase interference in the different-task case, arguing against a rapid-switching explanation. [Work supported by NIH/NIDCD.]

**5aPP10. Explaining two-tone suppression and forward masking data using a compressive gammachirp auditory filterbank.** Toshio Irino (Faculty of Systems Eng., Wakayama Univ., 930 Sakaedani, Wakayama 640-8510, Japan) and Roy Patterson (Univ. of Cambridge, Cambridge, CB2 3EG, UK)

The gammatone filter was imported from auditory physiology to provide a time-domain version of the roex auditory filter and enable the development of a realistic auditory filterbank for models of auditory perception. The gammachirp auditory filter was developed to extend the domain of the gammatone auditory filter and simulate the changes in filter shape that occur with changes in stimulus level. Recently, the gammachirp was extended to explain the level-independent frequency glide of the impulse response and a large volume of simultaneous masking data quantitatively. Although this could be implemented with a static filter, we used a time-varying filter whose active component is an IIR asymmetric compensation filter. In this case, it is necessary to estimate the signal level that controls the level dependency, and explain how the level measurement produces the time course of threshold observed in forward masking experiments. In this talk, we propose a new implementation of the time-varying, compressive gammachirp auditory filterbank to explain two-tone suppression and forward masking data qualitatively. We will also show that the system can resynthesize compressed speech sounds and so provide the basis for a gammachirp analysis/synthesis filterbank. [Work supported by GASR(B)(2) No. 15300061, JSPS.]

**5aPP11. Behavioral estimates of level-dependent shifts in the vibration pattern of the apical region of the basilar membrane.** Luis F. Barrios, Enrique A. Lopez-Poveda, and Ana Alves-Pinto (Instituto de Neurociencias de Castilla y Leon, Univ. of Salamanca, Avda. Alfonso X El Sabio s/n, 37007 Salamanca, Spain)

The best frequency of any given site on the BM shifts to a different value as the stimulus level increases. For basal sites, the shift occurs towards lower frequencies. For apical sites, however, the direction of the shift is unclear. Auditory nerve data [e.g., Rose *et al.*, J. Neurophysiol. **34**, 685–699 (1971)] suggest that shifts may not occur or may occur toward higher frequencies. The present work investigates which is the case in human. To do it, psychophysical tuning curves were measured for three normal-hearing listeners using forward masking. The level of a masker tone required to just mask a fixed, low-level probe tone was measured for different masker-probe time intervals. The duration of the interval was adjusted as necessary to obtain PTCs for the widest possible range of masker levels. Probe frequencies of 125, 250, and 500 Hz were considered. Masker frequencies ranged from 0.5 to 2.0 times the probe frequency. A clear shift toward lower frequencies occurred for all probe frequencies for one listener only. For the other two listeners, shifts were not observed for any probe frequency. No clear shifts toward higher frequencies were observed. [Work supported by Spanish FIS PI020343 and G03/203.]

**5aPP12. Modeling the influence of the cochlear nonlinearity on estimates of psychophysical tuning.** Xuedong Zhang and Andrew J. Oxenham (Res. Lab. of Electron., MIT, Cambridge, MA 02139)

Most behavioral measures of human frequency selectivity have been made with simultaneous masking and the notched-noise technique. The resulting filter shapes may be influenced by the effects of cochlear nonlinearity, such as suppression. Forward masking may provide a measure that is more comparable to neural tuning curves, because it does not involve stimuli that interact with each other along the basilar membrane. This study investigated the extent to which cochlear nonlinearities can account for differences in results between forward and simultaneous masking. The model was constructed using a nonlinear auditory filter, a sliding temporal integrator, a logarithmic transform and a template mechanism. The effects of compression and suppression on psychophysical performance were simulated by varying the relevant parameters of the model auditory filter. The psychophysical results were simulated for both forward and simultaneous masking, using the same parameters and tracking procedure as in the behavioral studies. The results provide a detailed evaluation of the role of compression and suppression in the models predictions of psychophysical tuning and assist in the development of the refined nonlinear cochlear models for human. [Work supported by the ASA Hunt Fellowship and NIH R01DC03909.]

**5aPP13. Estimating cochlear-filter shapes, temporal-window width and compression from tone-sweep detection in spectral and temporal noise gaps.** Gaston Hilkhuysen, Tammo Houtgast, and Johannes Lyzenga (Dept. of Otolaryngol., VU Univ. Medical Ctr., de Boelelaan 1117, 1081 HV Amsterdam, The Netherlands, G.Hilkhuysen@vumc.nl)

A test, designed for naive listeners, measured tone-sweep detection in noise with either spectral or temporal gaps. For normal-hearing (NH) listeners, detections in spectral gaps depended on level, which can be ex-

plained from Outer-Hair-Cell (OHC) activity. At low levels, OHC activity increased frequency-selectivity by amplifying the signal in the spectral gap, improving the signal-to-noise ratio (SNR). Relative to the broad passive cochlear filter, OHC activity decreased with rising signal levels. In consequence, SNRs decreased and detection deteriorated. Similarly, decreasing OHC activity may explain the observed level dependence of detection thresholds in temporal gaps. At low and high intensities, signal and noise were equally amplified. Detection was best at intermediate levels when the low-level signal in the temporal gap was amplified more than the high-level noise. All effects are modeled using a one-parameter time window with decaying-exponential shape preceded by a simplified dual-resonance non-linear (DRNL) filter. The filter contains two parallel, one-parameter, Rounded Exponential filters: a broad filter representing passive cochlear filtering and a narrow one, combined with a level-dependent amplifier, representing OHC activity. From estimated filter and time-window widths and OHC amplification, compression curves are derived. Additionally, results for hearing-impaired listeners will be presented.

**5aPP14. Speech understanding in noise: Contributions of compressive nonlinearities in the basilar-membrane response.** Amy R. Horwitz, Jayne B. Ahlstrom, and Judy R. Dubno (Dept. of Otolaryngol.-HNS, Medical Univ. of South Carolina, 135 Rutledge Ave., P.O. Box 250550, Charleston, SC 29425, horwitar@musc.edu)

The contribution of compressive nonlinearities in the basilar-membrane response was assessed by measuring speech recognition in noise as a function of noise level and growth of masking for tones. Consonant recognition was measured in interrupted noise at overall masker levels of 47–77 dB SPL. Additionally, thresholds for a 10-ms, 2.0-kHz tone were measured in a masker ranging in level from 40 to 85 dB SPL and centered at 1 kHz. Subjects were younger and older adults with normal hearing. With speech at a constant level and the masker level varying, it was hypothesized that the response of the basilar membrane to the masker would be linear at lower levels and compressed at medium to higher levels, resulting in less effective masking at higher masker levels. Consistent with the hypothesis, differences between observed and predicted consonant-recognition scores, determined using AI calculations, varied as a function of masker level, especially for younger subjects. At lower masker levels, observed scores declined as predicted as masker level increased; with further increases in masker level, scores declined less than predicted. For tonal growth-of-masking functions, breakpoints were correlated with quiet thresholds: both were significantly higher for older than younger subjects. [Work supported by NIH/NIDCD.]

**5aPP15. Psychometric-function slopes for forward-masked tones in listeners with cochlear hearing loss.** Kim S. Schairer, Jessica Messersmith, and Walt Jesteadt (Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131)

Slopes of psychometric functions (PF) for forward-masked tones in normal-hearing (NH) listeners decrease as a function of signal threshold in both on- and off-frequency masker conditions. This result is attributed to nonlinear cochlear response growth, and has been used to demonstrate comparable compression at 4000 and 250 Hz. The current study further tested the hypothesis that the slope effect is due to nonlinearity. In hearing-impaired (HI) listeners who presumably lack cochlear nonlinearity, PF slopes should remain steep across signal threshold levels. Four NH and six HI listeners with a range of hearing thresholds were tested in on- and off-frequency masker conditions, with signals of 4000 and 250 Hz. Otoacoustic emissions were measured to verify absent or decreased cochlear nonlinearity in the HI listeners. In general, PF slopes were steeper for the HI than NH listeners. One subject had normal hearing at 250 Hz, and hearing loss at 4000 Hz. Her PF slopes were comparable to the mean slopes across NH subjects in the 250-Hz signal conditions, but were elevated in comparison to the NH subjects in the 4000-Hz conditions. These results are consistent with predictions from a model of forward masking that incorporates cochlear nonlinearity as one of its stages.

**5aPP16. Masker variability effects in backward-masked frequency discrimination.** Blas Espinoza-Varas and Hyunsook Jang (Commun. Sci. and Disord., Univ. Oklahoma Health Sci. Ctr., Oklahoma City, OK 73190)

Masker variability effects have been studied mostly in detection tasks with simultaneous maskers; this paper reports variability effects in a frequency discrimination threshold (FDT) paradigm with backward maskers. FDTs for sinusoidal targets were measured unmasked (or in isolation) and in two backward-masked conditions: one included within-trial masker frequency variability, the other did not. The 1500-Hz, 40- or 80-ms targets, and 500-, 1500-, or 2500-Hz, 80-ms maskers were presented at 70-dB SPL with a 20-ms ISI. A standard and two comparisons were displayed, respectively, in the 1st, 2nd, and 3rd observation interval of a 3I/2AFC task. One comparison, chosen randomly with 0.5 probability, contained an increment in target frequency as specified by an adaptive rule that estimated 71 percent-correct FDTs. Identical increments in masker frequency were added to both comparisons in within-trial variability conditions, but not in conditions without variability. In all conditions, the task was to determine which comparison contained the target frequency increment. Masked no-variability FDTs were higher than unmasked FDTs, but training nullified this effect. Masker variability induced greater FDT elevations with 40- than with 80-ms targets, and in conditions having the same rather than different target-masker frequencies. Prolonged training nullified all masker variability effects.

**5aPP17. Hearing relative phases for two harmonic components.** D. Timothy Ives, Roy D. Patterson (C.N.B.H., Physio. Dept., Univ. of Cambridge, Downing St., Cambridge, CB2 3EG, UK, tim.ives@mrc-cbu.cam.ac.uk), and H. Martin Reimann (Univ. of Berne, 3012 Berne, Switzerland)

There is a new wavelet model of basilar membrane motion which predicts that existing roex and gammatone filterbanks underestimate the interaction of harmonically related components in complex tones. This interaction appears as a modulation of the basilar membrane motion associated with a higher harmonic by the presence of a lower harmonic; the period of the modulation is that of the difference frequency, or the fundamental of the implied harmonic series. The modulation depth is largest for stimuli whose spectra have a similar energy distribution as natural sounds: a high frequency roll-off of about 12-dB per octave. The strength of the modulation frequency is dependent on the relative phase of the components, which predicts that listeners will be able to hear a relative phase change. A series of experiments were undertaken to determine the threshold for a change in the relative phase of the components. The results of the experiments show that a change in the relative phase is detectable for components that are generally believed to be resolved on the basilar membrane and therefore should not interact. [Work supported by the U.K. Medical Research Council (G9901257).]

**5aPP18. The fading of auditory memory.** Liang Li, Juan Huang, Lingzhi Kong, Ying Huang (Dept. of Psych., Speech and Hearing Res. Ctr., Peking Univ., Beijing 100871, China liangli@pku.edu.cn), Xihong Wu, Jing Chen, Qiang Huang, Yuan Yao (Peking Univ., Beijing 100871, China), James Qi, and Bruce Schneider (Univ. of Toronto at Mississauga, Mississauga, ON, Canada)

Due to auditory memory, the auditory system is capable of maintaining a detailed representation of arbitrary waveforms for a period of time, so that a broadband noise and its delayed copies can be perceptually fused. This auditory memory would be critical for perceptually grouping correlated sounds and segregating uncorrelated sounds in noisy, reverberant environments. Its fading process over time was investigated in the present study at the behavioral level, using a break in correlation (BIC, a drop of inter-sound correlation from 1.00 to 0 and then return to 1.00) between two correlated broadband noises. The results show that with the rise of inter-sound delay from 2 to 10 ms under either headphone-stimulation or loudspeaker-stimulation conditions, the shortest BIC duration necessary

for listeners to correctly detect the occurrence of the BIC increased rapidly. This elevation in the duration threshold was faster under the headphone-stimulation condition than the loudspeaker-stimulation condition. Also, the listeners reaction time in response to the BIC but not that to a comparable silent gap elongated quickly with the increase in the intersound delay from 1 to 8 ms. Thus the auditory memory of fine structures fades rapidly after the sound waves are received. [Work supported by MSTC and NSERCC.]

**5aPP19. Evidence for a cancellation mechanism in perceptual segregation by differences in fundamental frequency.** John F. Culling, Gareth M. Linsmith, and Tracy L. Caller (School of Psych., Cardiff Univ., Tower Bldg., Park Pl. Cardiff, CF10 3AT, UK)

Identification of target speech material is better when it differs in fundamental frequency ($F0$) from interfering speech material. Two experiments tested the idea that this effect is mediated by a cancellation process that perceptually removes sound on one $F0$. Expt. 1 used monotonized target speech at 112.5 Hz $F0$ against two monotonized interferers. The $F0$s of the interferers were either both 112.5 Hz, both 126 Hz, or 100 and 126 Hz. Speech reception thresholds (SRTs) were about 3.5 dB lower when the two interferers shared a 126 Hz $F0$ than in the other two conditions, which differed by only 0.3 dB. These results are consistent with a cancellation mechanism that is able to remove sounds at only a single $F0$. Expt. 2 replicated and extended Expt. 1, testing whether the results were affected by the allocation of the target and interferers to particular $F0$s. When all sentences shared a common $F0$, SRTs were now 1–1.5 dB higher than when they were all different. However, all 4 SRTs for interferers sharing the same $F0$ (differing from that of the target by 2 or 4 semitones) were lower than all three of the SRTs for interferers with independent $F0$s.

**5aPP20. Effects of high-frequency amplification and training on impaired-listeners' ability to discriminate redundant cues.** Blas Espinoza-Varas, Shelagh Bowman-Edmundson, and Hyunsook Jang (Commun. Sci. and Disord., Univ. Oklahoma Health Sci. Ctr., Oklahoma City, OK 73190)

This paper examines effects of high-frequency amplification and training on the ability to discriminate redundant duration increments added simultaneously to the low ($L$) and high ($H$) frequency component of two-component complexes. The frequency of $L$ (500 or 1000 Hz, 80 ms) and $H$ (2000 or 3127 Hz, 60 ms) was chosen so as to stimulate simultaneously normal- and impaired-sensitivity regions of high-frequency sensorineural patients. The 3I/2AFC task displayed a standard followed by two comparisons. Duration increments in $L$, $H$, or both ($tL$, $tH$, or $tLH$), occurred randomly in comparison 1 or 2, and listeners had to decide which had an increment. The training stages were: (1) no $H$ amplification, with $L=35$, $H=-5$ dB SL and $tL$, $tH$, and $tLH$; (2) $H$ amplification, same as before but with $H=35$ dB SL; (3) $H$ amplification and $tH$ discrimination training, same as stage 2, but with $tH$ only and $H$ in isolation or together with $L$; and (4) retest of stage 2, post $tH$ discrimination training. Only in stage 4 did redundancy yield a significant improvement in overall discrimination accuracy; amplification alone yielded little improvement.

**5aPP21. Wideband reflectance in normal and otosclerotic ears.** Navid Shahnaz and Karin Bork (Univ. of British Columbia, School of Audiol. & Speech Sci., 5804 Fairview Ave., Vancouver, BC, Canada V6T 1Z3, nshahnaz@audiospeech.ubc.ca)

For years immittance has been used in order to help diagnose middle ear pathologies. Specifically, multi-frequency tympanometry (MFT) is able to relate information regarding the mechano-acoustical characteristics of the middle ear system. In the past two decades a new method of middle ear measurement, wideband reflectance (WBR), has been introduced. WBR is the ratio of energy reflected from the surfaces of the ear canal and middle ear on its way to the cochlea in relation to the energy that reaches the surface, or incident energy. This ratio is known as energy reflectance. This paper adds to the limited normative data available, as well as explores whether these normative data have a clinical utility in the diagnosis of otosclerosis. Descriptive statistics were gathered from 62 (115 ears) Caucasian normal hearing adults as well as in seven patients (seven ears) with otosclerosis. All of the otosclerotic patients in this study deviated from the normative values on at least one of the four WBR parameters of power absorption, admittance, susceptance, or conductance even when their MFT results were within normal limits. Although only seven patients were tested, these results provided evidence in favor of the utility of WBR for diagnosis of otosclerosis.

**5aPP22. Sensitivity to combined frequency and amplitude speech cues by cochlear implant users.** Eric W. Healy, Cheryl F. Rogers, and Allen A. Montgomery (Dept. of Commun. Sci. and Disord., Arnold School of Public Health, Univ. of South Carolina, Columbia, SC 29208, ewh@sc.edu)

Although a considerable amount of work has been directed toward examining the sensitivity of the cochlear implant (CI) user to various acoustic cues, less has examined the ability to combine cues. The sensitivity to concurrent cues was examined using an adaptive 3I-2AFC paradigm. Stimuli were synthesized three-syllable words having middle syllables that possessed an increase in fundamental frequency, amplitude or both. Sensitivity to increments in the individual cues was first measured for five subjects with a CI and five age-matched controls. These DLs were then used to equate sensitivity to the two cues and create stimuli having concurrent increments. It was found that the presence of the two cues reduced the mean DL to half that of either cue alone, for both groups of subjects. Thus, this combination of cues is similar across groups and simply additive. Individual differences in the ability of CI users to combine cues were more strongly predictive of performance on speech recognition tests than was sensitivity to either cue alone. These results have implications for the perception of the multiplicity of speech cues. [Work supported by NIDCD.]

**5aPP23. Recognition of vowels that have been spectrally warped according to the frequency map for the spiral ganglion.** Peter S. Popolo and Christoper W. Turner (Dept. of Speech Pathol. and Audiol., 119 WJSHC, The Univ. of Iowa, Iowa City, IA 52242, peter-popolo@uiowa.edu)

The purpose of this study was to explore the effect of a specific spectral warping on vowel recognition in cochlear implant users. The warping was designed to simulate the misalignment of frequency-to-place stimulation which occurs if the spiral ganglion cells are electrically stimulated instead of the eighth-nerve dendrites in the organ of Corti. Simulated CIS-processed vowel stimuli were generated in which the analysis and carrier band center frequencies were shifted according to the projection of the spiral ganglion cells onto the organ of Corti, derived from published data [A. Kawano, H. L. Seldon, and G. C. Clark, Ann. Otol. Rhinol. Laryngol. **105**, 701–709 (1996)]. Similar spectral warping was applied to unprocessed vowel stimuli. Listening experiments were conducted in which normal hearing subjects were presented with four types of stimuli: unprocessed-unwarped, processed-unwarped, unprocessed-warped, and processed-warped. For each condition, performance was allowed to stabilize over repeated trials with feedback, to allow for learning. Vowel recognition scores averaged across listeners were drastically reduced for the warped stimuli, regardless of whether the stimuli were CIS-processed or not. The combination of spectral warping and reduced spectral resolution resulted in the poorest intelligibility scores. These results may have implications for the design of frequency-to-place maps in cochlear implants.

**5aPP24. Utilizing different channels for multiple inputs in cochlear implant processing.** Bom Jun Kwon (Cochlear Americas, 400 Inverness Pkwy, Ste. 400, Englewood, CO 80112, bjkwon@gmail.com)

While cochlear implants successfully provide auditory sensation for deaf people, speech understanding through the device is compromised when there is a background noise or competing sounds, partly due to implant users' reduced ability in auditory grouping. The present study investigates whether providing multiple streams of input on different channels would facilitate auditory grouping, thereby assisting speech understanding in competing sounds. In acoustic hearing, presenting two streams of input (such as speech and noise) in spectrally separate channels generally facilitates grouping; however, in electric hearing it is difficult to predict and separating them could lead to a negative result, because channel interactions inferred from the excitation patterns are severe and the actual amount of electric current for the noise delivered to the cochlea would be much higher for a given SNR, therefore contaminating the target more effectively. Results from consonant identification measured in a variety of speech/noise conditions (same/different channels) indicate that speech understanding generally improves with separate channels, implying that implant users appear to extract speech information on the basis of spatial (channel) separation, easily overcoming the distracter from the adjacent channels with higher intensity. This also proposes a new measure of channel interactions based on auditory grouping.

FRIDAY MORNING, 20 MAY 2005 PLAZA B, 9:00 A.M. TO 12:00 NOON

# Session 5aSA

## Structural Acoustics and Vibration: General Vibration: Modeling, Propagation, Dissipation and Control

Linda P. Franzoni, Chair
*Dept. of Mechanical Engineering and Material Science, Duke Univ., Durham, NC 27708-0300*

***Contributed Papers***

**9:00**

**5aSA1. Time reversal in heterogeneous flexural beams.** Dany Francoeur and Alain Berry (Dept. Gen. Mecanique, Universite de Sherbrooke, 2500 boul. de l'Universite, Sherbrooke, QC, Canada J1K 2R1, Alain.Berry@USherbrooke.ca)

Time reversal of acoustic and structure-borne waves has been explored in recent years mostly for ultrasound and for nondispersive propagation, that is under frequency-independent wave velocity. In contrast, the case of time reversal in flexural beams presented here involves dispersive propagation, and is carried for frequencies below 5 kHz. The study has been started with analytical time reversal simulations in infinite homogeneous or heterogeneous beams (comprising point-mass scatterers). Experiments have also been realized on a 5 m beam with anechoic terminations and under transverse impact excitation. The time-reversal mirror was made of several thin piezoceramic elements bonded on the beam to sense the impulse response of the structure and re-emit its time-reversed image. The experimental results are in good agreement with the analytical results, and show that time spreading due to dispersive propagation of bending waves is successfully compensated by the time reversal operation. The presentation will illustrate the main results of the simulations and a comparison with the experiments.

**9:15**

**5aSA2. Medium frequency vibration modeling of cracked plates using hierarchical trigonometric functions.** Jérôme Pinsonnault, Patrice Masson, Philippe Micheau (GAUS, Mech. Eng. Dept., Université de Sherbrooke, Sherbrooke, QC, Canada J1K 2R1, Patrice.Masson@USherbrooke.ca), and Nezih Mrad (Defence R&D Canada, Dept. of Natl. Defence, Ottawa, ON, Canada K1A 0K2)

A modeling tool is proposed to describe the vibration behavior of pristine and damaged plates in the medium frequency range (below 10 kHz). This tool is intended to provide a platform for the development and assessment of damage detection algorithms for aircraft structural health monitoring applications. The proposed analytical model employs a Hierarchical Trigonometric Function Set (HTFS) to characterize homogeneous plates with through cracks. This approach takes advantage of the very high order of stability of the HTFS [O. Beslin and J. Nicolas, J. Sound Vib. **202**, 633–655 (1997)] to approximate the effects of a small crack in a plate for all combinations of classical boundary conditions (e.g., CFSC, CCFF, FSFS). The model is first presented and then assessed for healthy and cracked CCCC plates with eigenvalues and eigenmodes presented in the literature. For a healthy square plate, numerical results provide good agreement up to the 1000th mode while, for a cracked rectangular plate, good agreement is obtained up to the 3rd mode, corresponding to the highest mode order available in the literature. Wave propagation simulation obtained from HTFS shows the scattering around the cracks in the plates. Experimental validation of the model is conducted both in frequency and time domains for healthy and cracked plates. [Work supported by the Consortium for Research and Innovation in Aerospace in Quebec (CRIAQ) and Defence R&D Canada.]

**9:30**

**5aSA3. An exact analytical expression of the shear coefficient in the Mindlin plate equation.** Andrew Hull (Code 8212, Naval Undersea Warfare Ctr., Newport, RI 02841, hullaj@npt.nuwc.navy.mil)

This paper derives an exact analytical expression of the shear coefficient in the Mindlin plate equation for a plate of infinite extent. The equation of motion derived from the Mindlin plate equation is set equal to the equation of motion derived from the thick plate equation, and the result is a closed-form expression of the shear coefficient at all wave numbers and frequencies. A numerical example is included to illustrate the variation of the shear coefficient. It is shown that the shear coefficient is extremely dependent on wave number and only slightly dependent on frequency. Shear coefficients derived in other work are compared favorably to the values calculated by this new method at the plate flexural wave response. [Work funded by the Office of Naval Research.]

**9:45**

**5aSA4. Structural acoustic control of plates with variable boundary conditions.** Robert L. Clark, Joseph D. Sprofera (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Box 90300, Durham, NC 27708-0300), Gary P. Gibbs, and Ran H. Cabell (NASA Langley Res. Ctr., Structural Acoust. Branch)

A method for optimizing a structural acoustic control system with respect to potential variations in plate boundary conditions is provided. The assumed modes method is used to build a plate model with varying levels of rotational boundary stiffness to span a range of possible boundary conditions which can be used to capture uncertainty in the model. A transducer placement scoring process, involving Hankel singular values (HSVs), is combined with a genetic optimization routine to find spatial locations robust to boundary condition variation. Predicted frequency response characteristics are examined, and optimized results are discussed in relation to boundary condition variations. Results indicate it is possible to minimize the impact of uncertain boundary conditions in active structural acoustic control by optimizing the placement of transducers with respect to uncertainties. Both analytical and experimental results will be discussed.

**10:00–10:15 Break**

**10:15**

**5aSA5. Optimal energy dissipation in a semi-active friction device.** Paulin Buaka, Philippe Micheau, and Patrice Masson (GAUS, Mech. Eng. Dept., Université de Sherbrooke, Sherbrooke, QC, Canada J1K 2R1, Patrice.Masson@USherbrooke.ca)

A semi-active device is presented for vibration control using energy dissipation by dry friction at contact surfaces. Semi-active behavior is provided by two piezoelectric stack actuators driven in real time to apply a normal force on a mobile component through two friction pads. Theoretical and experimental results show that there is an optimal constant normal force to maximize the energy dissipated for the case of a harmonic disturbance. In order to improve the energy dissipation by real time control of the normal force, two nonlinear controllers are proposed: (1) the Lyapunov method leading to a nonlinear bang-bang controller law and (2) the feedback linearization approach leading to equivalent viscous friction. The implementation of both strategies is presented and both are experimentally assessed using a clamped-free beam with the semi-active device attached to the beam. It is shown that a proper choice for the parameters of the controllers leads to an increased energy dissipation with respect to the case where the normal force is constant. This dissipation is further increased by adjusting a phase shift in the nonlinear feedback loop in order to avoid a stick-slip motion of the mobile component.

**10:30**

**5aSA6. Energy sinks: Vibration absorption by an optimal set of undamped oscillators.** Ilker Koç (Mech. Eng. Dept., Carnegie Mellon Univ., Pittsburgh, PA 15213), Antonio Carcaterra (Universita Degli Studi di Roma "La Sapienza," 00184 Roma, Italy), Zhaoshun Xu, and Adnan Akay (Carnegie Mellon Univ., Pittsburgh, PA 15213)

This presentation offers the concept of energy sinks as an alternative to conventional methods of vibration absorption and damping. A prototypical energy sink envisioned here consists of a set of oscillators attached to, or an integral part of, a vibrating structure. The oscillators that make up an energy sink absorb vibratory energy from a structure and retain it in their phase-space. In principle, energy sinks do not dissipate vibratory energy as heat in the classical sense. The absorbed energy remains in an energy sink permanently so that the flow of energy from the primary structure appears to it as damping. This paper demonstrates that a set of linear oscillators can collectively absorb and retain vibratory energy with near irreversibility when they have a particular distribution of natural frequencies. The approach to obtain such a frequency response is based on an optimization that minimizes the energy retained by the structure as a function of frequency distribution of the oscillators in the set.

**10:45**

**5aSA7. Experiments on vibration absorption using energy sinks.** Adnan Akay, Zhaoshun Xu (Mech. Eng. Dept., Carnegie Mellon Univ., Pittsburgh, PA 15213), Antonio Carcaterra (Universita Degli Studi di Roma "La Sapienza," 00184 Roma, Italy), and Ilker Koç (Carnegie Mellon Univ., Pittsburgh, PA 15213)

This presentation describes experiments that demonstrate the concept of energy sinks where a set of multiple undamped linear oscillators attached to a vibrating structure can absorb most of its energy. In principle, energy sinks do not require presence of damping in the classical sense. A set of undamped oscillators that make up an energy sink collectively absorb the vibratory energy and retain it in their phase space. Earlier optimization studies by the authors have shown the feasibility of vibration absorption and retention by energy sinks if the set of oscillators have a particular frequency distribution. Experimental results support the concept of energy sinks. Different physical realizations of energy sinks demonstrate the significance of frequency distributions and the ability of energy sinks to reduce vibration amplitude of a primary structure to which they are attached.

**11:00**

**5aSA8. Experimental study on passive/active hybrid isolation.** Zi Jun Zhang, Woo Suk Chang, Koon Meng Nyang, and Yew Wing Chan (DSO Natl. Labs., Singapore. No 20, Sci. Park Dr., Singapore 118230)

This paper discusses the control of a high stroke low stiffness nonlinear actuator which formed the key element of a smart engine mount system that provides strong support for the engine while at the same time absorbing the engine vibration energy. The actuator is made of a stacked PZT embedded in an elliptical shaped metal frame, in the horizontal direction. Due to the geometry of the frame, the displacement generated by the PZT stacks is amplified up to about 5 times in the vertical direction. However, due to the geometry of the actuator, the vertical and horizontal motions are non-linear and excite motions at multiples of the driving frequency. Feedback controllers cannot be used since the high frequency motion causes spillover problem and control becomes unstable. The filtered-X adaptive controller with sufficient high sampling rate is tested and found to be a useful and reliable controller for the actuator for suppressing the vibration to almost ambient noise level. This actuator requires relatively lower driving voltage level and delivers larger stroke range. This would be more suitable for vibration isolation application where the actuator is serially connected with passive dampers.

**11:15**

**5aSA9. Comparison between control strategies for active constrained layer damping treatment to control the sound radiation from a vibrating structure.** Hui Zheng, Chun Lu, and Heow Pueh Lee (Inst. of High Performance Computing, 1 Sci. Park Rd., #01-01 The Capricorn, Sci. Park II, Singapore 117528)

A comparative study is presented of three control strategies for active constrained layer damping (ACLD) treatments to control the sound radiation from a vibrating planar structure. The first control strategy is one commonly used in most existing studies of ACLD for structural vibration control; i.e., the error signal to be minimized for the controller is the vibration response sensed by point transducers, and a proportional derivative controller is employed where the sensor signal and the voltage output is related by real-valued feedback gains. The second control strategy is similar to the first, except that the real-valued control gains are substituted by complex-valued ones. In the third control strategy, the discrete structural acoustic sensing approach is introduced for observing the sound radiation from the structure, and the estimated sound power constitutes the controller input. The control gains aiming to reduce the sound radiation from a simply-supported beam are optimized respectively for the three control strategies, and the control performances are compared. Numerical results show that using the complex-valued control gain in the controller design is always better than using real-valued gain. Furthermore, the ACLD treatments adopting the third control strategy require lowest control efforts.

11:30

**5aSA10. Vibration control of optomechanical components.** Vyacheslav M. Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA, 92606, vyacheslav.ryaboy@newport.com)

The purpose of optomechanical components is to anchor optical elements (such as light sources, lenses, mirrors, etc.) in space so that the optical paths would be unperturbed by environmental impacts, first of all by vibration. Dynamic alignment of components is a main design optimization criterion for the optomechanical systems. An optical table or breadboard is often used as a common base for the whole optomechanical assembly. It can be sufficiently isolated from the floor vibration, but its own flexural resonance vibrations may still cause misalignments. The paper estimates various ways of vibration damping of optical tables, including new methods of passive damping combined with motion transformation, as well as active damping. Optical rods and posts offset optical elements from the table. These components have their own resonance properties. The paper presents analytical estimates and optimization methods for introducing damping materials in the optical posts. Experimental results comparing dynamic properties of damped and non-damped structures will be presented.

11:45

**5aSA11. Whole body vibration measurements on forklift trucks.** Alberto Behar (Noise Control, 45 Meadowcliffe Dr, Scarborough, ON, Canada, M1M 2X8 behar@sympatico.ca) and Steven Libich (WESA)

Data on acceleration values on forklift trucks related to Whole Body Vibration are notably missing in the literature. Although there are several standards that provide guidance on how measurements have to be performed, they are somehow conflicting and difficult to implement since they require simultaneous measurement and processing of data in the three axes, $x$, $y$ and $z$. Standards also provide limit values for safety of the personnel exposed to the vibrations. The development of new instruments has made these kind of measurements much easier to perform and to interpret the results, since they include filters following the standards and allow for the use of tri-axial accelerometers. This paper describes measurements done on 11 forklift trucks in a real-life situation, while performing tasks normal for their use. The trucks are of the standing-operator type. The accelerometers were mounted on the body of the truck, so that vibrations were representative for what the operators were exposed to. The three-axes orientation of the accelerometer were taken into account. The paper will present results of the measurements and their assessment following the existing ISO and BS Standards.

FRIDAY MORNING, 20 MAY 2005 REGENCY D, 8:00 A.M. TO 12:00 NOON

## Session 5aSC

## Speech Communication: Intelligibility and Studies of Populations with Speech and Hearing Disorders (Poster Session)

Sigfrid D. Soli, Chair
*House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057*

### *Contributed Papers*

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

**5aSC1. Evaluation of effect of presbycusis on speech intelligibility by several kinds of speech test in rooms.** Hiroshi Sato (Inst. for Human Sci. & Biomed. Eng., Natl. Inst. of Adv. Industrial Sci. and Technol., 1-1-1 Higashi, Tsukuba, Japan) and Hayato Sato (Kobe Univ., Rokko, Nad, Kobe 657-8501, Japan)

Word recognition tests with logatom and word familiarity controlled word lists and sentence intelligibility test in simulated sound fields with noise and/or reverberation were carried out to assess the effect of hearing loss due to aging on speech communication in rooms. The result demonstrates that (1) speech recognition scores of elderly listeners are 25% lower than those of young adults for any kinds of speech test. This difference is equal to the 5 dB increase of ambient noise for elderly listeners. (2) Detailed speech recognition process in noise and/or reverberation is described with scores of various kinds of speech test. (3) Peripheral auditory functions are mainly affected by aging. On the other hand, central auditory processing functions of the aged examined with word familiarity and meanings of words shows same performance as the young. These results were expected to lead the discussion for speech communication in aged society and the standardization for sound environment.

**5aSC2. Effect of training using lexically easy and hard words on speech understanding in noise.** Matthew H. Burk and Larry E. Humes (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405, maburk@indiana.edu)

Older hearing-impaired adults often have difficulty understanding speech in noise, even with proper amplification. One reason for this difficulty may be a lack of practice or inability to make use of new auditory information, which has been absent due to a progressive, peripheral hearing loss over many years. An often overlooked aspect of the rehabilitation process, which could help to improve this deficit, is listener training. The goal of this study was to create a word-based training protocol which could improve speech understanding in noise when listeners are presented with new, novel stimuli outside the clinic. Previous work with word-based training using one talker showed a large training effect that generalized to novel talkers; however, sufficient generalization to novel words and sentences was lacking. The current study attempts to increase generalization by training listeners with multiple talkers and lexically hard words. Generalization to novel words, both lexically easy and lexically hard, novel talkers, and sentences, with the latter also constructed from lexically easy and hard words, will be described. [Work supported, in part, by a research grant from NIA, R01-AG08293, awarded to the second author, and an NIH training grant.]

**5aSC3. Comparison of speech intelligibility measures.** Jacqueline S. Laures and Gary G. Weismer (Georgia State Univ., Atlanta, GA 30302) and (Univ. of Wisconsin-Madison, Madison, WI)

The speech intelligibility of dysarthric speakers is perceptually measured by one of the following four techniques: direct magnitude estimation with a modulus, free modulus magnitude estimation, interval scaling, and transcription. Weismer and Laures (2002) suggest that magnitude estimates may provide a more complete representation of speech intelligibility than other methods of measurement because it may be more sensitive to non-segmental aspects of speech, such as prosody. However, the empirical data supporting such a statement is quite limited. The purpose of the current study is to explore the relationship of the four different measurement techniques to determine if one approach may provide a more accurate determination of the speech intelligibility of dysarthric speakers. Twelve listeners measured the speech of six dysarthric speakers and two healthy control speakers using the four different measurement techniques. Each speaker produced three sentences twice. The sentences were presented via a loudspeaker in a sound attenuated booth. Listeners rated the sentences using the four techniques. The order of techniques used was counterbalanced. A correlation analysis revealed that the four techniques were highly related. Implications of this finding are discussed.

**5aSC4. Effects of speech-rate and pause duration on sentence intelligibility in younger and older normal-hearing listeners.** Akihiro Tanaka, Shuichi Sakamoto, and Yô-iti Suzuki (R.I.E.C., Tohoku Univ., Katahira 2-1-1, Aoba-ku, Sendai 980-8577, Japan)

Speech-rate conversion techniques aid speech comprehension by allowing more time for perceptual and cognitive processes. However, if only the speech-rate of a telecast is converted, auditory and visual information become asynchronous. One possible method to resolve the problem is to reduce the pause durations between phrases; unfortunately, this can evoke a marked negative effect. For that reason, the present study examines the effects of the speech-rate and pause duration on sentence intelligibility. We manipulated the lengths of phrases relative to the original length (0, +100, +200, +300, and +400 ms), and the pause durations between phrases in a sentence (0, 100, 200, 300, and 400 ms). Listeners were asked to write down sentences they discerned from the noise. The intelligibility score increased in younger and older listeners when the speech signal was expanded. Regarding the pause duration, intelligibility was best when the pause duration was 200 ms in younger listeners; in older listeners, the intelligibility score was highest when the pause durations were 200 ms and 400 ms. These results provide evidence that might benefit speech-rate conversion through better use of pause duration.

**5aSC5. Simulation of temporal aspects of auditory aging.** Ewen MacDonald (Inst. of Biomaterials and Biomed. Eng., Rm 407 Rosebrugh, Univ. of Toronto, Toronto, ON, Canada M5S 3G9, macdone@ecf.utoronto.ca), Kathy Pichora-Fuller, and Bruce Schneider (Univ. of Toronto at Mississauga (UTM), Mississauga, ON, Canada L5L 1C6)

A jittering technique to disrupt the periodicity of the signal was used to simulate the effect of the loss of temporal synchrony coding believed to characterize auditory aging. In one experiment jittering was used to distort the frequency components below 1.2 kHz and in a second experiment the components above 1.2 kHz were distorted. To control for spectral distortion introduced by jittering, comparison conditions were created using a smearing technique (Baer and Moore, 1993). In both experiments, 16 normal hearing young adult subjects were presented with SPIN sentences in three conditions (intact, jittered, and smeared) at 0 and 8 dB SNR. When the low frequencies were distorted, speech intelligibility in the jittered conditions was significantly worse than in the intact and smeared conditions, but the smeared and intact conditions were equivalent. When the high frequencies were distorted, speech intelligibility was reduced similarly by jittering and smearing. On low-context jittered sentences, results for young adults mimicked results found previously for older listeners with good audiograms (Pichora-Fuller *et al.*, 1995). It is argued that the jittering technique could be used to simulate the loss of neural synchrony associated with age-related changes in temporal auditory processing.

**5aSC6. Comparison of hearing loss compensation algorithms using speech intelligibility measures.** Meena Ramani (Dept. of Elec. and Comput. Eng., Univ. of Florida, P.O. Box 116130, Bldg. 33, Ctr. Dr. Rm. NEB 444, Gainesville, FL 32611), John G. Harris (Univ. of Florida, Gainesville, FL 32611), Alice E. Holmes (Univ. of Florida, Gainesville, FL 32611), Mark Skowronski (Univ. of Florida, Gainesville, FL 32611), and Sharon E. Powell (Univ. of Florida, Gainesville, FL 32611)

Sensorineural hearing loss includes loss of high-frequency sensitivity which results in decreased speech intelligibility. The loss cannot be compensated by inverting the audiogram because of the non-linear effects of sensorineural hearing loss (frequency smearing, decreased dynamic range, decreased time-frequency resolution). Several non-linear compensation schemes exist (Half-gain, POGO, NAL-R, Fig. 6, DSL and LGOB) and this paper provides a comparison of those using the objective Perceptual Evaluation of Subjective Quality (PESQ) score and the subjective Hearing In Noise Test (HINT). The listening tests were run on 15 unaided hearing impaired listeners as well as 15 normal hearing listeners using a simulated hearing loss algorithm. These results show marked improvement in intelligibility for the compensated speech over the normal speech for both normal and hearing impaired adults.

**5aSC7. A comparative study of perceived, predicted, and measured speech intelligibility.** Michael E. Hermes, Melinda J. Carney, and Dominique J. Cheenne (Dept. of Audio Arts & Acoust., Columbia College Chicago, Chicago, IL 60605)

Intelligibility metrics were obtained using a variety of methods in a gymnasium that serves as a place of worship. A word list trial, a computer model, and a computer-based %Alcons test provided the data. The results were compared in order to gauge their relative accuracy. The data from the %Alcons testing were found to be unreliable, but a direct relationship was established between the mean word list test scores and the results gathered from the computer model. This relationship allowed for a translation of the scores to %Alcons.

**5aSC8. A statistical model for prediction of functional hearing abilities in real-world noise environments.** Sigfrid Soli (House Ear Inst., 2100 W. 3rd St., Los Angeles, CA 90057), Chantal Laroche, Christian Giguère, and Véronique Vaillancourt (Univ. of Ottawa, Ottawa, ON, Canada)

Many tasks require functional hearing abilities such as speech communication, sound localization, and sound detection, and are performed in challenging noisy environments. Individuals who must perform these tasks and whose functional hearing abilities are impaired by hearing loss may constitute safety risks to themselves and others. We have developed and validated in two languages (American English and Canadian French) statistical techniques based on Plomps (1986) speech reception threshold model of speech communication handicap. These techniques predict functional hearing ability using the statistical characteristics of the real-world noise environments where the tasks are performed together with the communication task parameters. The techniques will be used by the Department of Fisheries and Oceans Canada to screen individuals who are required to perform hearing-critical public safety tasks. This presentation will summarize the three years of field and laboratory work culminating in the implementation of the model. Emphases will be placed on the methods

for statistical characterization of noise environments, since these methods may allow generalization of the model to a wider range of real-world noise environments. [Work sponsored by Department of Fisheries and Oceans Canada.]

**5aSC9. Predicting speech intelligibility in real-world noise environments from functional measures of hearing.** Christian Giguère, Chantal Laroche, Véronique Vaillancourt (Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON, K1H 8M5, Canada), and Sigfrid Soli (House Ear Inst., Los Angeles, CA 90057)

In everyday life situations and in many occupational settings, speech communication is often performed in noisy environments. These environments can sometimes be very challenging, particularly for individuals impaired by hearing loss. Diagnostic measures of hearing, such as the audiogram, are not adequate to make accurate predictions of speech intelligibility in real-world noise environments. Instead, a direct functional measure of hearing, the Hearing In Noise Test (HINT), has been identified and validated for use in predicting speech intelligibility in a wide range of face-to-face speech communication situations in real-world noise environments. The prediction approach takes into account the voice level of the talker in noise due to the Lombard effect, the communication distance between the talker and the listener, a statistical model of speech perception in specific noises, and the functional hearing abilities of the listener. The latter is taken as the elevation of the individual's speech reception threshold in noise above the normative value for the HINT test. This test is available in several languages, so that language-specific needs can be addressed. The detailed approach will be presented with an emphasis placed on application examples in clinical and/or occupational settings.

**5aSC10. Perceptual, acoustic, and tongue shape measures during /r/ production pre- and post-treatment using visual feedback from ultrasound: case studies of two adolescents.** Marcy Adler-Bock, Barbara Bernhardt, Penelope Bacsfalvi (Dept. of Audiol. and Speech Sci., Univ. of British Columbia, 5804 Fairview Ave., Vancouver, BC, V6T 1Z3), and Bryan Gick (Univ. of British Columbia, Vancouver, BC, Canada V6T 1Z1)

This study examined the effectiveness of using visual feedback from ultrasound in remediation of persistent /r/ errors. Ultrasound provided the participants and the clinician with a dynamic sagittal or coronal image of the tongue during speech production. The participants in this study were two adolescent boys ages 12 and 14 who were not able to produce an on-target North American /r/. Both participants had received at least one year of traditional /r/ therapy without improvement. Treatment was provided over 13 one-hour sessions using visual feedback from ultrasound. Initially, /r/ was broken down and rehearsed as individual motor targets (tongue tip, body, root); these components were then practiced in combination to produce /r/ in isolation, then in syllables, words, and phrases. Post-treatment changes in /r/ production were captured through transcription, acoustic analysis, and tongue shape measurement. Both participants /r/ productions were rated as having more tokens of on-target /r/ post-treatment. Acoustic results supported these findings with the third formant lowering post-treatment. Tongue shape measures indicated that the participants tongue shapes were more similar to the modeled /r/ tongue shape post-treatment. These case studies suggest that visual feedback as provided by ultrasound may be a useful adjunct to speech (re)habilitation.

**5aSC11. The influence of severity of speech involvement on acoustic measures in dysarthria.** Yunjung Kim, Gary Weismer, and Ray D. Kent (Dept. of Communicative Disord. and Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI 53705)

Several different acoustic measures have described the articulatory deficit and predicted the overall speech intelligibility deficit in speakers with dysarthria. The articulatory basis of acoustic variables that predict speech intelligibility variations across speakers have been thought to be the ones that should be manipulated clinically for maximum therapeutic effect. For example, $F2$ extent and slope are known to have strong correlations with speech intelligibility measures, across dysarthric speakers. This may suggest that within-speaker manipulation of magnitude and/or speed of articulatory movements will, if successful, result in improved speech intelligibility. Some of our previous work [Weismer *et al.*, 2001, Folia Phoniat. Logopaed.] indicates, however, that these measures may not be predictive of within-speaker fluctuations in intelligibility, but rather are indices only of across-speaker variation in severity. The large data base of dysarthric speakers at UW-Madison permits us to begin to determine how much variability exists for measures like $F2$ slope in a relatively large group of speakers with homogeneous speech severity. This paper will report results of $F2$ extent and slope measures for speakers with relatively mild speech involvement. If the measures are primarily tied to severity, they should not vary much within a homogeneous group of speakers. [Work supported by DC00319.]

**5aSC12. Variability of jaw movement in contrastive stress production of children with and without speech delay of unknown origin.** Jennell C. Vick, Lakshmi Venkatesh, and Christopher A. Moore (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA 98101)

This study was designed to evaluate the control of jaw movement in contrastive stress productions of children with and without speech delay. The spatiotemporal index (STI) was used to calculate variability in jaw movement trajectories in 12 children producing three different metrical forms of CVCV syllables (trochaic, iambic, and even stress; papa, mama, and baba). The children (mean age: 3;2 years) were categorized, four in each of three groups, as having Normal Speech Acquisition (NSA), Speech Delay (SD), or both (NSA/SD) using the Speech Disorder Classification System (Shriberg *et al.*, 1997). Results replicated findings in typically developing children of a similar age reported by Goffman and Malin (1999) where iambic forms were produced with greater stability than trochaic forms as measured by the STI. Analysis using a repeated measures ANOVA revealed significant effects for contrastive stress type, speech disorder classification, and syllable identity. Results are reported in the context of vowel acoustic measures of contrastive stress. [Work supported by NIDCD R01 DC 0000822-10.]

**5aSC13. Articulatory movements during vowels produced by speakers with dysarthria and normal controls.** Yana Yunusova, John Westbury, and Gary Weismer (Dept. of Commun. Disord. and Waisman Ctr., Univ. of Wisconson–Madison, 1500 Highland Ave., Madison, WI 53705)

The central goal of the study was to provide a quantitative description of contributions of the jaw, lower lip, tongue blade and tongue dorsum to vowel productions, and to determine patterns of interarticulatory interactions between movements. Kinematic and acoustic signals were collected using the x-ray microbeam. Thirty-four speakers, 12 with dysarthria due to Parkinson disease, 7 with amyotrophic lateral sclerosis, and 15 normal controls, were recorded reading sentences at a comfortable speech rate. Ten CVC words, each containing one of the English vowels /i, I, u, a, ae, o/ carrying primary stress, were selected for analysis. Each fleshpoint trajectory was characterized by marker positions at vowel onset and offset, and the moment when speed was lowest. Measures of distance traveled, time to and from the moment of minimum speed, and peak and average movement speed were employed. Movement characteristics, and associations between movements, were compared for different vowels, contexts, speakers and groups. Results are reported for vowels and vowel groups (e.g., lax versus tense), averaged separately by contexts for speaker groups. The data speak to previous claims that speakers with dysarthria exhibit evidence of discoordination in speech movements relative to normal performance. [Work supported by NIDCD Award R01 DC003723.]

**5aSC14. Spectral contributions to intelligibility of sentences with flattened fundamental frequency.** Peter J. Watson and Robert S. Schlauch (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. S.E., Shevlin 115, Minneapolis, MN 55455, pjwatson@umn.edu)

Recently, the contribution of fundamental frequency ($F0$) variation to speech intelligibility in background noise has been studied [J. Laures and G. Weismer, JSLHR **42**, 1148–1156 (1999)]. The process used for this analysis is to flatten the frequency contour at the average $F0$. Results show that sentences with flattened $F0$ are less intelligible than those with natural $F0$ variation. However, this technique may reduce the prominence of formant peaks because it excludes $F0$s that are below the average flattened frequency. As noted by Laures and Weismer (1999), eliminating lower $F0$s from the analysis results in wider spacing between the harmonics and the available energy to excite the vocal tract resonances is diminished. This study examined the effect of flattening at the average and lowest $F0$ of each sentence to intelligibility. Twenty subjects listened to sentences in a continuous noise background. Sentences were equated for root-mean square energy. Results showed that the low-flattened condition was significantly more intelligible than the average-flattened condition. However, a condition with natural $F0$ variation was more intelligible than the two flattened conditions. These findings show that that the frequency of flattening does contribute to intelligibility but that natural F0 variation appears to contribute to a greater degree.

**5aSC15. Comparison of cues in auditory selective and divided attention.** Jae hee Lee and Larry E. Humes (Indiana Univ. at Bloomington, Bloomington, IN 47405)

This study investigated auditory selective and divided attention in ten normal-hearing listeners. Subjects listened to two simultaneous sentence-like utterances from the Coordinate Response Measure (CRM) speech corpus at 90 dB SPL in dichotic or monotic listening conditions. This corpus has the following format, Ready (call sign), go to (color) (number), now, with all possible combinations of call signs (8), colors (4) and numbers (8). In all conditions, the listener identified the color-number coordinates spoken by the target talker. The target talker was identified by a cue provided either prior to (selective attention) or following (divided attention) the presentation of the two simultaneous utterances. In the first experiment, either the ear or the call sign served as the cue for the target talker in monotic or dichotic listening conditions. In the second experiment, talker gender served as the cue in monotic listening conditions. Across both experiments, performance for selective attention was superior to that for divided attention. Performance for ear or gender cueing was better than for call sign. Analysis showed that 80% of errors were due to intrusions from the competing utterance, supporting an informational (attention), rather than energetic, form of masking. [Work supported, in part, by NIH R01 AG08293.]

**5aSC16. Effects of high intensity on recognition of low- and high-frequency speech in noise.** Van Summers and Mary Cord (Army Audiol. & Speech Ctr., Walter Reed Army Medical Ctr., Washington, DC 20307-5001)

For listeners with normal hearing (NH), speech recognition scores often decrease when intensities exceed moderate levels (rollover is observed). It is currently unclear whether the factors leading to rollover in NH listeners also limit performance for hearing-impaired (HI) listeners at high sound levels. This study aimed at clarifying the stimulus conditions most clearly associated with rollover and whether rollover effects are similar for NH listeners and listeners with mild hearing impairment. In Stage 1, NH and HI listeners heard digitally-filtered sentences and adaptive procedures were used to determine high- and low-frequency bandwidths yielding 70%-correct word recognition in quiet at moderate levels. In Stage 2, broadband and band-limited stimuli (based on the high-, and low-frequency passbands measured in quiet in Stage 1) were tested at moderate levels in background noise. Noise levels were varied adaptively to determine signal-to-noise levels supporting 30%-correct recognition. Stimulus conditions leading to criterion performance at moderate presentation levels in Stage 2 were then retested at higher levels in Stage 3. NH listeners showed larger and more consistent rollover effects for high-frequency than for low-frequency or broadband stimuli. The results for HI listeners showed greater variability but also indicated clear rollover effects for high-frequency stimuli at high levels.

**5aSC17. The effect of auditory feedback alterations on the speech quality of hearing aid and cochlear implant users.** Dragana Barac-Cikoja and Leslie Klein (Gallaudet Univ., 800 Florida Ave. NE, Washington, DC 20002)

The speech of seven hearing aid (HA) users with severe-profound hearing loss and six cochlear implant (CI) users was recorded as they read the Rainbow Passage while their speech feedback was either spectrally altered in real-time, completely masked by multi-talker babble noise, or unaltered. Spectral alterations were implemented by filtering the speech signal into either one or four frequency bands, extracting their respective amplitude envelope(s), and amplitude-modulating the corresponding noise band(s). While the single-band condition provided only coarse information about the speech rhythmic structure, the four-band noise signal remained intelligible. Auditory feedback was presented via insert earphones to the HA users, and via the auxiliary jack (with the headpiece microphone silenced) to the CI users, at the participants' most comfortable listening level. The quality of the recorded speech (separated into individual sentences) was assessed using a 2IFC procedure. For each combination of the experimental conditions, six judges selected the more natural-sounding utterance in a pair. Preference scores were calculated for each of the four feedback conditions and statistically tested. HA and CI group differed in how feedback intelligibility affected the speech quality. Possible acoustic correlates of the perceived differences will be discussed.

**5aSC18. Articulatory and acoustic measurements of vowels in hearing impaired speakers following treatment.** Penelope Bacsfalvi and Barbara Bernhardt (School of Audiol. and Speech Sci. (SASS), UBC, Vancouver, BC, Canada V6T 1Z3, Penelope@audiospeech.ubc.ca)

The purpose of this study was to examine the relationships between ultrasound tongue shapes, electropalatography (EPG) contacts, formant data, and the perceptual data in the vowels of three hard of hearing adolescents. This pilot study examines the pre- and post-therapy speech of the participants in a 6 week group therapy programme with ultrasound and EPG technologies. Before treatment, participants showed particular difficulty with high vowels and the tense-lax distinction. Recordings were made of 2 males and 1 female with ultrasound and EPG. Three types of measurements were made; formant measurements, EPG tongue palate contacts, and perceptual judgements by experienced speech-language pathology researchers. Initial analysis values showed a change in the direction of EPG contact patterns, and perceptual categories. $F1$ values decreased and $F2$ values increased, moving in the direction of typical formant frequencies found in hearing people. Preliminary results of this study support a lingual-articulatory approach to treatment.

**5aSC19. The role of fundamental frequency ($F0$) contours in the perception of speech against interfering speech.** Christine Binns and John F Culling (School of Psych., Cardiff Univ., Tower Bldg., Park Pl. Cardiff, CF10 3AT, UK)

A set of experiments investigated the effect of the $F0$ contour on speech intelligibility against interfering sounds. Speech Reception Thresholds (SRTs) were measured in dB for sentences with different manipulations of their $F0$ contours. These manipulations involved either a scaled reduction in $F0$ variation, or the complete inversion of the $F0$ contour. Against speech-shaped noise, a flattened $F0$ contour did not have a significant impact on the intelligibility of speech compared to a normal $F0$ contour: the SRT for the flattened $F0$ contour being only 0.7 dB higher. The SRT for the inverted contour, however, was 1.6 dB higher than for the normal $F0$ contour. When the sentences were played against a single-talker interferer, the overall effect was greater, with a 2.1 dB difference between the normal and flattened conditions, and 3.3 dB between the normal and inverted $F0$ contours. There was found to be no effect of altering the $F0$ contour of the interferer, indicating that any abnormality of the $F0$ contour serves to mask the intelligibility of the target speech, but does not alter the effect of the interfering speech. Future research will investigate the impact on speech intelligibility of similar manipulations of a low-pass-filtered $F0$ contour.

**5aSC20. Perceptual strategies for identifying vowels produced by speakers with dysarthria.** Kate Bunton (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, P.O. Box 210071, Tucson, AZ 85721)

It has been previously reported that changes to the fundamental frequency contour of vowels influences their identification [e.g., H. Traunmuller, J. Acoust. Soc. Am. **69**, 1465–1475 (1981)]. Similar changes in vowel identification have been demonstrated for some listeners when the $F0$ contour of a dysarthric speaker has been modified (e.g., flattened or enhanced) [Bunton (2004)]. These listeners appear to rely on $F0-F1$ space for identification of vowel height; this strategy is consistent with a perceptual model of vowel recognition described by A. Syrdal and H. Gopal [J. Acoust. Soc. Am. **79**, 1086–1100 (1986)]. Other listeners, however, appear to utilize different acoustic cues to identify a given vowel. This study focused on defining those perceptual strategies. Multidimensional plots of acoustic cues likely to influence vowel perception are used to illustrate perceptual strategies that may have been used by these listeners for identification of vowels in the speech of individuals with dysarthria associated with Parkinson disease. [Work supported by NIH R03 DC005902.]

**5aSC21. Perception of coarticulation in gated words by dyslexic and non-dyslexic children.** Patricia Keating (UCLA, Los Angeles, CA 90095-1543), Frank Manis, Jennifer Bruno, and Jonathan Nakamoto (USC, Los Angeles, CA 90089-1061)

In an auditory word gating task, listeners are presented with portions of words, and try to identify these acoustic fragments as lexical items. It has been shown that children need more acoustic information than adults to succeed, and that dyslexic children can require more information than other children. It is also known that adults can use early acoustic cues to identify features of upcoming segments. For example, in English anticipatory vowel nasalization implies that a nasal consonant will follow, and similarly anticipatory lateralization. Our study asked whether dyslexic children are impaired in their perception or use of such anticipatory coarticulatory information. Successive gates from test words that ended in a nasal (8 words) or /l/ (4 words), or control words with final oral stops, were blocked by length and presented to 26 dyslexic and 26 non-dyslexic children. Responses were audiorecorded and later transcribed; responses were scored re both the full word and the nasality/laterality of the final consonant. Dyslexics as a group required more gates to identify nasality, but not whole words. Language-impaired dyslexics required more gates to identify whole words as well as nasality. Performance on the gating task significantly predicted reading ability when age and IQ were controlled.

**5aSC22. Phonetic labeling along a formant transition duration continuum in children with specific language impairment.** Harvey M. Sussman, Elizabeth Burlingame, Ronald B. Gillam (Dept. of Commun. Sci. & Disord., Univ. of Texas, Austin, TX 78712), and Jessica F. Hay (Univ. of Texas, Austin, TX 78712)

Fifteen children diagnosed with specific language impairment (SLI) and fifteen typically developing (TD) children were tested for identification performance on two synthetic speech continuua varying in formant transition durations (FTDs). One continuum varied from [ba]-to-[wa] and the other varied from [da]-to-[ja]. Several dependent measures were derived based on d'from signal detection theory. These measures were used to assess category boundaries and to indirectly assess sensitivity to phonetic changes in labeling category tokens along each continuum. The SLI group showed less consistent identification performance along the [ba]-[wa] series relative to the TD group, as well as reduced sensitivity to phonetic changes along the continuum. On the [da]-[ja] continuum, the SLI group revealed less consistent identification performance on the short FTD end, but similar identification levels to the TD group at the long FTD end. The overall results support the view that children with SLI reveal a deficiency in the processing of speech sounds at the level of segmental identity.

**5aSC23. Classification of dysarthric and non-impaired speech based on prosodic features.** Greg Kochanski (Oxford Univ., 41 Wellington Square, Oxford OX1 2JF, UK, gpk@kochanski.org) and Rupal Patel (Northeastern Univ., Boston, MA 02115)

Prosodic differences between dysarthric and healthy speakers were studied. Six acoustic properties that are plausibly more influenced by suprasegmental aspects of speech (e.g., emphasis) than the segmental details of the words were measured. The time course of these properties were analyzed over each utterance by fitting Legendre Polynomials. The resultant Legendre coefficients were then fed to linear- and quadratic-discriminant classifiers. All of the six properties were individually capable of distinguishing dysarthric speech from healthy speech. Based on one acoustic property measured over a single short sentence, we could correctly classify a speaker as healthy or dysarthric 55–75% of the time, depending on the acoustic property used. More complex classifiers that used all the acoustic properties correctly classified the speaker 97% of the time based on nine utterances. The strongest difference between normal and dysarthric speech was in loudness. Dysarthric speakers did not reliably produce the loudness patterns associated with stressed syllables. They also had a wider range in amplitude, greater voicing variability, smaller excursions of fundamental frequency, and less final lengthening compared to healthy speakers. The classification we demonstrated may be extended to a become a graduated measurement of severity, thereby contributing to diagnostics and intervention in dysarthria.

**5aSC24. Effects of hearing impairment on the perception and neural representation of time-varying spectral cues.** Ashley W. Harkrider, Patrick N. Plyler, and Mark S. Hedrick (Dept. of Audiol. and Speech Pathol., Univ. of Tennessee, 457 South Stadium Hall, Knoxville, TN 37996)

Differences in phonetic boundaries versus normal controls suggest that listeners with hearing impairment (HI) have difficulty categorizing stop consonant place of articulation based solely on the dynamic spectral information present in the second formant transition ($F2$), even when the stimuli are amplified. This may be due to a degraded ability of the central auditory nervous system to process time-varying spectral cues despite ensuring overall audibility. However, increasing the overall level of the stimuli may not result in improved audibility of $F2$. To determine if spectral shaping of $F2$ improves performance of listeners with HI, psychometric functions and $N1$-$P2$ cortical responses were compared in 10 older listeners with normal hearing versus 10 older listeners with HI. Stimuli were synthetic consonant-vowels along a /ba/-/da/-/ga/ place-of-articulation continuum in an unshaped and shaped condition. Generally, behavioral and $N1$-$P2$ results indicate that, with shaping, categorization of /d/ and /g/ improves. These findings suggest that enhanced audibility of $F2$ through spectral shaping does improve perception of stop consonant stimuli. However, categorical boundaries for the individuals with HI are shifted lower in frequency with shaping for all phonemes versus normal controls, indicating that enhancing audibility improves but does not completely normalize categorization performance.

**5aSC25. Effect of two-band dichotic listening for hearing impaired listeners.** Shuichi Sakamoto, Atsunobu Murase, Yôiti Suzuki (Res. Inst. of Elect. Comm./ Grad. School of Information Sci., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi, Japan, saka@ais.riec.tohoku.ac.jp), Tetsuaki Kawase, and Toshimitsu Kobayashi (Tohoku Univ., Aoba-ku, Sendai, Miyagi, Japan)

The increase of the upward spread of masking is a phenomenon that is typically observed among sensorineural hearing-impaired listeners. To resolve this problem, dichotic listening, by which an input speech spectrum is split into two complementary parts and is presented dichotically, seems effective to reduce masking between contiguous frequency bands. This study examines effects of simple two-band dichotic listening with a cut-off frequency around and between the typical first and second formant frequencies of the preceding vowel. We measured speech intelligibilities in both quiet and noisy environments (S/N 4 and 0 dB). Three types of vowel-consonant-vowel nonsense monosyllables, of which preceding vowels were /a/, /i/, and /u/, were used as speech stimuli. Results showed that this dichotic processing was effective, especially in relatively high S/N conditions. Moreover, the best dividing frequency was dependent on the preceding vowel. When /a/-consonant-vowel was used, the best dividing frequency was 1.0 kHz (around $F1$ of Japanese vowel /a/), whereas the best dividing frequency was 0.8 kHz (between $F1$ and $F2$ of Japanese vowel /u/) when the /u/-consonant-vowel was used.

**5aSC26. Signal to noise ratio loss and consonant confusions.** Yang-soo Yoon and Jont B. Allen (Univ. of Illinois, Speech and Hearing, 901 s sixth, Champaign, IL 61820, yyoon5@uiuc.edu)

Previous SNR loss (also called speech loss) studies showed that (1) SNR loss cannot be predicted from audiometric measures, (2) 40% of hearing aids wearers have 5 dB SNR loss or greater, and (3) SNR loss influences speech intelligibility significantly. These showed SNR loss to be important in speech recognition, but they do little, or no to illuminate the nature of consonant confusion, resulting from SNR loss. Thus, the goal of the current study was to investigate the effect of SNR loss on 16 consonants recognition in hearing impairment as a function of SNR. Confusion matrix data were collect and analyzed, and Fletcher's AI was calculated from the SNR. These two measures were utilized (1) to determine how SNR loss was related to the event loss, (2) to test whether clustering of syllables in terms of consonant confusions was complied with SNR loss, and (3) to compare PI functions obtained from subjects and AI model. The results show that the degree of consonant confusion varies, but members of consonants confused with target sound above chance level are similar, as a function of SNR loss and SNR. It suggests that SNR loss limits recognition for specific consonants, even in noise.

**5aSC27. Driving performance and auditory distractions.** Elzbieta B. Slawinski, Jane F. MacNeil, Mona Motamedi, Benjamin R. Zendel (Psych. Dept., Univ. of Calgary, 2500 Univ. Dr., Calgary, AB, Canada T2N 1N4), Kirsten Dugdale, and Michelle Johnson (Univ. of Calgary., Calgary, AB, Canada T2N 1N4)

Driving performance depends on the ability to divide attention during different tasks. In spite of the fact that driving abilities are associated with visual stimulation, driving performance depends on attention to stimulation and/or auditory distraction. Research shows that listening to the radio is a principal auditory distracter during the time of driving (Brodsky, 2002). In the laboratory a few experiments were conducted on the auditory distraction (e.g., music, stories) and signal processing by young and older drivers. Results show that older subjects involved in listening to the stream of information (independent of the hearing status) require higher intensity of the auditory stimulation than younger drivers. It was shown that cognition plays a role while listening to auditory stimuli. Moreover, it was demonstrated that driving performance was influenced by the type of performed music. A portion of these experiments and their results were presented at the Annual Meetings of CAA in 2002 and 2003 as well as being published in the Journal of Psychomusicology **18**, 203–209. Complete results of the experiments will be discussed.

**5aSC28. Speech intelligibility index calculations in light aircraft cabin during flight.** Tino Bucak and Ernest Bazijanac (Dept. of Aeronautics, Faculty of Transport and Traffic Eng., Univ. of Zagreb, Croatia)

High levels of cabin noise in small general aviation aircraft significantly deteriorate the quality of speech communications and potentially endanger the safety of flight. Several ground and inflight cabin noise measurements on new generation Cessna 172R were made during various phases of flight. The results are analyzed and used for Speech Intelligibility Index (SII) calculations, in order to quantify the influence of cabin noise on speech communications between crew members.

**5aSC29. A detailed study on the effects of noise on speech reception.** Tammo Houtgast and Finn Dubbelboer (VU Univ. Medical Ctr., Amsterdam, The Netherlands)

The effect of adding continuous noise to a speech signal was studied by comparing, for a series of quarter octave bands, the band output for the original speech and for the speech-plus-noise. Three separate effects were identified. (a) Average envelope-modulation reduction: the original intensity-envelope is, on average, raised by the mean noise intensity, resulting in a reduction of the original modulation index. (b) Random instantaneous envelope fluctuations: on an instantaneous basis, the speech-plus-noise envelope shows random variations, caused by the stochastic nature of the noise, and by the instantaneous changes in the phase relation between the speech and the noise. (c) Perturbations of the carrier phase: in the band output carrier signal the addition of the noise causes random phase changes. By applying signal processing techniques, we were able to either include or exclude each of these three effects separately. The results of intelligibility measurements indicated the following order of importance of the three different effects: (1) the average envelope-modulation reduction, (2) the perturbation of the carrier phase, and (3) the random envelope fluctuations. The results will be discussed in the light of modeling and enhancing (noise suppression schemes) speech reception in noise.

**5aSC30. Speech rate characteristics in dysarthria.** Kris Tjaden, Geoff Greenman, Taslim Juma, and Roselinda Pruitt (Dept. of Communicative Disord., Univ. at Buffalo, 122 Cary Hall, 3435 Main St., Buffalo, NY 14214, tjaden@acsu.buffalo.edu)

Speech rate disturbances are pervasive in dysarthria, with some reports suggesting that up to 80% of speakers with dysarthria exhibit speech rates that differ from neurologically normal talkers. The contribution of articulation time and pause time to the overall impairment in speech rate is not well understood. Studies investigating speech rate characteristics in dysarthria also tend to focus on reading materials, yet there is reason to suspect that the higher cognitive load of conversational speech may impact speech rate characteristics differently for individuals with impaired speech motor control, and neurologically normal talkers. The current study will report speech rate characteristics for both a reading passage and conversational speech produced by individuals with dysarthria secondary to Multiple Sclerosis (MS), individuals with dysarthria secondary to Parkinson Disease (PD) and healthy controls. The manner in which speech rate, articulation rate, and pause characteristics differ for speakers with dysarthria and healthy controls will be examined. The contribution of articulation time and pause time to overall speech rate also will be studied and compared for the reading passage and conversational speech. [Work supported by NIDCD R01DC04689.]

# Session 5aSP

## Signal Processing in Acoustics: Smart Acoustic Sensing for Land-Based Surveillance

Brian Ferguson, Chair
*Defense Science and Technology Organization Maritime Systems Div., Pyrmont 2009, Australia*

**Chair's Introduction—8:30**

### *Invited Papers*

**8:35**

**5aSP1. Acoustic methods for tactical surveillance.** Brian G. Ferguson and Kam W. Lo (Defence Sci. and Technol. Organisation, P.O. Box 44, Pyrmont, NSW 2009, Australia)

Smart acoustic sensor systems can be deployed for the automatic detection, localization, classification and tracking of military activities, which are inherently noisy. Acoustic sensors are appealing because they are passive, affordable, robust, and compact. Also, the propagation of sound energy is not limited by obstacles that block or obscure the clear line of sight that is required for the effective operation of electromagnetic systems. Methods, with examples, for extracting tactical information from acoustic signals emitted by moving sources (air and ground vehicles) are provided for both single sensor and multiple sensor configurations. The methods are based on processing either the narrowband or broadband spectral components of the sources' acoustic signature. Weapon firings generate acoustic impulses and supersonic projectiles generate shock waves enabling source localization and classification by processing the signals received by spatially-distributed sensors. The methods developed for land-based acoustic surveillance using microphone data are also applied to hydrophone data for passive acoustic surveillance of the underwater environment.

**9:05**

**5aSP2. Autonomous acoustic sensing on mobile ground and aerial platforms.** Tien Pham and Nassy Srour (US Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197)

Acoustic sensor systems on the ground and/or in the air can be used effectively for autonomous and remote intelligence surveillance and reconnaissance (ISR) applications. Acoustic sensors can be used as primary sensors and/or secondary sensors to cue other higher-resolution sensors for detection, tracking and classification of continuous and transient battlefield acoustic events such as ground vehicles, airborne aircraft, personnel, indirect fire, and direct fire. Current collaborative research activities at ARL in acoustic sensing from mobile ground platforms such as HMMWVs and small robotic vehicles [P. Martin and S. Young, Proc. of SPIE Defense & Security Symposium, 2004] and from aerial platforms such as UAVs and balloons [Reiff, Pham *et al*, Proc. of the 24th Army Science Conference, 2004] demonstrate practical performance enhancements over fixed ground-based platforms for a number of ISR applications. For both mobile ground and aerial platforms, self-generated noise (flow noise and platform noise) is problematic but they can be suppressed with specialized windscreens, sensor placement, and noise cancellation technology. Typical acoustic detection and processing results for mobile platforms are compared and contrasted against fixed ground-based platforms.

**9:35**

**5aSP3. Ferret and its applications.** Jacques Bedard (Defence R&D Canada–Valcartier, 2459 Pie XI North, Val-Belair, QC, Canada G3J 1X5)

Ferret is an acoustic system that detects, recognizes and localizes the source and direction of small arms fire. The system comprises a small array of microphones and pressure sensors connected to a standard PC-104 computer that analyzes, displays, reports and logs the parameters of a recognized shot. The system operates by detecting and recognizing the ballistic shock wave created by the supersonic bullet, combined with the muzzle blast wave propagating from the weapon. The Canadian Land Force Test and Evaluation Unit evaluated a vehicle-mounted version of the system and recommended deployment of the system during peacekeeping missions. The system is the result of a collaborative effort between Defence R&D Canada and MacDonald Dettwiler and Associates. This presentation describes the hardware and software components of the system along with the current and future applications of the system.

**10:05–10:20 Break**

## *Contributed Papers*

**10:20**

**5aSP4. Acoustic self-localization of a wireless sensor network.** Peter L. Schmidt, Stephen M. Williams, and Kenneth D. Frampton (Vanderbilt Univ., 2400 Highland Ave., Nashville, TN 37212)

One of the biggest challenges to the field of wireless sensor networks is self-localization: that is the determination of the relative and absolute coordinates of each sensor node in the network. Previous work has been done to locate hydrophone arrays. However sensor networks have some unique constraints that make this more challenging. A typical application would involve the distribution of hundreds or thousands of sensor nodes over an area either by hand, airdrop or other means. One of the primary constraints on such a system is that centralized processing of self-localization data may be prohibitively complex. Furthermore, the data may be incomplete, contain reflected path events, and may be subject to other mission specific constraints. Therefore, a distributed computational scheme has been developed to solve acoustic time-of-arrival equations. *A priori* information about some sensor locations and user triggered source localization events are used along with a regularized inversion solution. Results of this will be presented based on both small scale experiments and larger systems simulations. Limits of the types of a priori information required for accurate results are detailed, along with studies of the accuracies obtained using various distributed calculations parameters.

**10:35**

**5aSP5. Distributed source localization in a wireless sensor network.** Stephen M. Williams, Peter L. Schmidt, and Kenneth D. Frampton (Vanderbilt Univ., 2400 Highland Ave., Nashville, TN 37212)

This work concerns experimental implementation of distributed acoustic source localization with a wireless sensor network. The envisioned application is the distribution of hundreds or thousands of small, semi-disposable, wireless sensor nodes (possibly by airdrop). These nodes would form an ad-hoc communications network and monitor and track acoustic events within the array. Centralized processing of such data would be prohibited by the excessive communications and complexity of calculations. Furthermore, localization with a network of randomly placed sensors is not suited to use of traditional array theories due to irregular placement of sensors and the unknown sensitivity and phase relationships between sensors. Therefore, a fully distributed localization algorithm was developed in which nodes organize themselves into groups which collaborate to locate sources. Acoustic event time-of-arrival information is shared among group members and the source locations were determined using Tikhonov regularized inversion. Small scale localization experiments (with 10 sensor nodes) were conducted to validate the performance of the method. Results are discussed, along with limitations of the method discovered during conduct of the experiments. In addition, a comparison is made between this method and traditional matrix solution methods.

**10:50**

**5aSP6. Particle filters for tracking a vehicle through a field of acoustic directional frequency and recording (DIFAR) sensors.** Anton J. Haug (The MITRE Corp., 7515 Colshire Dr., McLean, VA 22102, ahaug@mitre.org)

For nonlinear/non-Gaussian processes, Bayesian approaches to tracking require integration over probability density functions, which cannot be accomplished in a closed form. Recently, numerical Monte Carlo "particle filter" integration techniques have been developed and applied to tracking problems. However, multidimensional application examples of these techniques have been limited because they require an analytic expression for the likelihood function, which is usually not easy to obtain when the noise is non-Gaussian and not additive. Tracking a vehicle through a field of acoustic DIFAR sensors is such an application. Although the vehicle dynamic model can be linear with additive Gaussian noise, the observation model is inherently nonlinear with observations that contain embedded non-Gaussian noise. For the DIFAR vehicle tracking problem, we will consider large time-bandwidth signals, develop both the dynamic and observation models, and show how the Gaussian additive noise on each DIFAR channel results in non-Gaussian noise embedded in the bearing observations. The likelihood function will then be presented and the performance of several particle filter trackers will be compared with that of an Extended Kalman Filter.

**11:05**

**5aSP7. Source localization in complex scattering environments using random amplitude and bearing information.** D. Keith Wilson, Mark L. Moran, and Roy Greenfield (USACE ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@erdc.usace.army.mil)

By scattering sound waves, atmospheric turbulence and other outdoor objects (such as building and trees) induce random fading and angle-of-arrival variations in signals received by small-baseline microphone arrays. This study addresses the relative utility of signal amplitude and bearing estimates when such variations are present. Source-localization simulations are performed for four idealized statistical scattering models: no scattering, weak turbulent scattering, strong turbulent scattering, and diffuse (multiple) scattering. Each of these cases is considered with low, moderate, and high SNR. It is found that bearing information provides highly accurate source localization when SNR is high and scattering is negligible. However, when the SNR is low and/or there is significant scattering, the bearing information loses its utility while the amplitude information remains relatively robust. Algorithms are considered that attempt to assess the relative reliability of the bearing and amplitude information, and subsequently weight signal features to provide the most satisfactory results. The simulations also confirm previous analyses suggesting that, for unfavorable propagation conditions and SNR, Cramer-Rao lower bounds predict substantially better performance than is obtainable in practice.

**11:20**

**5aSP8. A method for locating nearby sources.** John V. Olson, Kenneth Arnoult, and Curt. A. S. Szuberla (Geophysical Inst., Univ. of Alaska, Fairbanks, AK 99775)

A fast method for locating nearby sources based solely on the time-of-flight information is described. The time delays for a signal passing between all sensor pairs is determined using cross-correlations and a vector of time delays is constructed. In the noise free case, where the speed of sound is constant, each point in the plane is associated with a unique value of the time delay vector. The method we present uses a fast, simplex-based search of the plane for a source location by minimizing the difference between the vector associated with the candidate location on the plane and the value estimated from cross-correlations. The search takes place over the three dimensional space that includes two coordinates in the plane and the propagation speed. The starting point for the search is constructed from an analytic fit of circular wave fronts to groups of sensors within the array. This method has been useful in identifying near-field sources in the 153US and 155US CTBT/IMS infrasound arrays. Examples of the locations we have determined along with a description of the statistical confidence limits associated with the method will be presented.

# Session 5aUW

## Underwater Acoustics: Scattering and Reverberation

Purnima Ratilal, Chair
*Northeastern Univ., Electrical and Computer Engineering, 409 Dana Research Ctr., Boston, MA 02115-5000*

### *Contributed Papers*

**8:30**

**5aUW1. Identification of strong scatterer locations on the Malta plateau.** Mark K. Prior (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19138, La Spezia, Italy, prior@saclantc.nato.int)

Reverberation data gathered in the Malta Plateau region of the Mediterranean Sea using a Low-Frequency Active Sonar (LFAS) and a cardioid receiver array are presented. A processing strategy involving normalization, geo-referencing and averaging over pings is described and is shown to highlight the locations of strong scatterers and to provide a first-order estimate of their strength. A second process of calculating the ratio of local standard deviation and mean values is shown to reveal scatterer locations and to provide an indication of whether the probability density function of the reverberation is well-described by the Rayleigh distribution. The characteristics of the observed scatterers are revealed by comparison with independent sidescan, echo sounder and boomer data. Areas of high reverberation are identified as being associated with wrecks, oil production platforms, rock outcrops and variations in local sediment properties.

**8:45**

**5aUW2. Range-dependent 3D scattering and reverberation in the continental shelf environment from biology, geology and oceanography.** Purnima Ratilal (Northeastern Univ., 409 Dana Res. Ctr., Boston, MA 02115), Sunwoong Lee, Yisan Lai, Tianrun Chen, Deanelle Symonds (MIT, Cambridge, MA 02139), Ninos Donabed (Northeastern Univ., Boston, MA 02115), and Nicholas Makris (MIT, Cambridge, MA 02139)

Several unified scattering and reverberation models were developed in support of the ONR Acoustic Clutter Program. They include a range-dependent model based on the parabolic equation that can be used to efficiently model scattering from a random spatial distribution of random targets that obey the sonar equation in the waveguide and a similar but range-independent waveguide model based on normal modes for scattering from extended objects. Both these models are bistatic and fully 3D, and the latter model also accounts for modal coupling between propagation and scattering caused by extended objects in the waveguide. These models are applied to examine both coherent and diffuse scattering measured after beamforming and match-filtering on an array from schools of fish, plankton, volume inhomogeneities in the sea bottom, roughness on the seafloor, extended seafloor and sub-bottom features such as river channels and reflective strata, and internal and surface waves. We provide a review of the dominant sources of clutter as well as background reverberation for long range active sonar based on comparison of model predictions with measured data from the Acoustic Clutter Experiments of 2001 and 2003.

**9:00**

**5aUW3. Dominant source of background reverberation during the Acoustic Clutter 2003 experiment.** Ninos Donabed, Purnima Ratilal (Northeastern Univ., 409 Dana Res. Ctr., Boston, MA 02115), and Nicholas Makris (MIT, Cambridge, MA 02139)

Reverberation data from the Acoustic Clutter 2003 experiment on the New Jersey Continental shelf are analyzed to provide estimates of the background scattering strength over wide areas of the environment as a function of time. We corrected the acoustic data in the 390–440 Hz range for two-way transmission loss modeled using the range-dependent parabolic equation, the spatially varying beampattern of the receiving array, and source level to obtain estimates of the scattering strength per unit area of the environment. Data from regions of space devoid of strong clutter were statistically analyzed to provide a temporal and spatial characterization of the background scattering strength. They were then compared to scattering strength levels deduced from modeling from a variety of scatterers including schools of fish, the sea bottom, internal waves and surface waves. Analysis indicates that the dominant sources of background reverberation can be either low density fish schools or the sea bottom.

**9:15**

**5aUW4. Higher moment estimation for shallow water reverberation.** Kevin LePage (Naval Res. Lab., Code 7144, 4555 Overlook Ave. SW, Washington, DC 20375)

Shallow water reverberation is characterized by clutter which causes false alarms in sonar systems. In previous work we have estimated the non-Rayleighness of shallow water reverberation time series predictions obtained by the R-SNAP model. Here we extend the R-SNAP model to the closed form estimation of the second moment of reverberation intensity, allowing the estimation of pdf fits for the two parameter $K$ distribution as a function of system and channel characteristics and time after shot. This work is intended to theoretically guide the selection of clutter-robust system characteristics. [Work supported by ONR.]

**9:30**

**5aUW5. Statistical characterization of sonar-like clutter observed on the STRATAFORM during the 2003 Acoustic Clutter Experiment in the 400–1500 Hz region.** John Preston and Douglas Abraham (Appl. Res. Lab., The Penn State Univ., State College, PA 16804)

In 2003 ONR sponsored the Acoustic Clutter Experiment to study shallow water scattering and clutter in the STRATAFORM area off New Jersey. Sources were bistatic coherent pulses from a vertical array. The receiver was the Five Octave Research Array (used horizontally). The STRATAFORM is known to have benign surface morphology but contains many buried river channels and other sub-surface horizons. MIT researchers have shown fish to be a primary source of the observed clutter and reverberation. The $K$-distribution's shape and scale parameters have been shown to be useful in describing non-Rayleigh behavior. Statistical characterization is presented as a function of location. The ''bandwidth'' effect is shown where the shape parameter first decreases inversely proportional to bandwidth but then increases in a trend back toward the Rayleigh distribution at higher bandwidths. The shape parameter estimates are well fit by an elongated patch model of Abraham and Lyons. Differences between the 2003 and 2001 data taken in the same area are discussed. It is believed that the sound speed profiles in 2003 lead to more bottom interaction than in 2001 producing a larger time spread in the multipaths leading to observed differences. [Work supported by ONR Code 32, Grant N00014-05-1-0156.]

**9:45**

**5aUW6. Comparison of measured buried target backscatter levels with high-order predictions: Rippled interface effects.** Raymond Lim (Naval Surface Warfare Ctr.-Panama City, Code R21, 110 Vernon Ave., Panama City, FL 32407), Joseph L. Lopes, and Gary S. Sammelmann

Controlled sonar measurements with buried spheres and cylinders have demonstrated diffraction by bottom ripple can significantly improve target detection performance with sonar operated at grazing angles shallower than the critical grazing angle. Furthermore, buried target scattering models based on low-order perturbation theory to handle sound transmission through bottom roughness confirm that a significant enhancement in detection performance can be expected. But comparisons between measurements and model predictions have not always produced good agreement. As the bottom ripple height increases, discrepancies have been noted in comparisons with the predicted backscatter levels from a buried sphere, suggesting a failure of low-order perturbation theory to properly account for high-amplitude roughness effects. Recently, a recursive algorithm to generate arbitrary-order perturbation corrections for transmission and reflection through an idealized sinusoidal ripple was formulated and incorporated into our existing buried target scattering codes. The resulting higher-order predictions indicate previous discrepancies were not necessarily due to the neglect of high-order corrections. Calculations show model inputs such as bottom attenuation can influence backscatter levels significantly and measurement of these parameters may need to be performed more carefully. We demonstrate this through a presentation of data and model comparisons. [Work supported by ONR and SERDP.]

**10:00–10:15 Break**

**10:15**

**5aUW7. Calculation of scattering from underwater targets using the equivalent source technique.** Ahmad T. Abawi and Michael B. Porter (Heat, Light, and Sound Res., Inc., San Diego, CA 92130, Ahmad.Abawi@HLSResearch.com)

The equivalent source technique is a method of computing scattering and radiation from a target by replacing the target by a distribution of discrete sources, whose complex amplitudes are determined by applying the boundary condition on the total field. The advantage of using this method, particularly in underwater acoustics, is that it essentially transforms the scattering and propagation problems into just a propagation problem, where the target can be treated as a distribution of sources rather than an impedance discontinuity. In this paper the equivalent source technique and different propagation models such as the normal mode, the parabolic equation and the fast field method are used to compute scattering from various target shapes/properties placed in the ocean waveguide. The results are compared with those obtained from other numerical codes. In addition, the relationship between the equivalent source technique with boundary element method (BEM) and the method of moments (MoM) is investigated.

**10:30**

**5aUW8. Detecting small targets in strong bottom reverberation.** Jinyun Ren and John S. Bird (School of Eng. Sci., Simon Fraser Univ., Burnaby, BC, Canada V5A 1S6, jren@cs.sfu.ca)

The detection of small targets moving or appearing in heavy bottom reverberation is a challenging problem for surveillance sonars. Conventional detection techniques, such as variable thresholding, are inefficient because they require that the target return be large enough to compete with the total reverberation return. Based on a new reverberation model, this paper proposes a target detection scheme that provides target subclutter visibility in the presence of reverberation. First it is shown through experimental evidence that bottom reverberation as seen by a stationary sonar is coherent or at least partially coherent from ping to ping. Therefore reverberation for a particular range cell can be modeled as a complex signal composed of a stationary or slowing varying mean plus a rapidly varying diffuse component. This mean is easily estimated using an adaptive mean estimator and then removed so that the target need only compete with the diffuse component. Experimental results show that the detection gain over variable thresholding as measured by the coherent-to-diffuse ratio can be as much as 40 dB.

**10:45**

**5aUW9. Channel effects on target characteristics in very shallow water.** Kevin LePage (Naval Res. Lab., Code 7144, 4555 Overlook Ave. SW, Washington, DC 20375)

Very shallow water waveguides introduce channel effects which complicate the scattered response from objects. Here we model the effects of channel time spread and dispersion on the expected value of the time series backscattered by a sphere with and without uncertainty introduced by imperfect knowledge of the channel. We also model the predictable characteristics of the bottom reverberation against which detections are made. Results are intended to guide the design of dispersion and uncertainty-robust sonar systems in very shallow water. [Work supported by ONR.]

**11:00**

**5aUW10. Mode extraction from acoustic backscatter in a waveguide.** Shane Walker, Philippe Roux, and William A. Kuperman (Scripps Inst. of Oceanogr., Marine Physical Lab., 9500 Gilman Dr., La Jolla, CA 92093-0238, shane@physics.ucsd.edu)

Recently we introduced a technique for extracting the acoustic modes in a range-independent waveguide from measured data alone without *a priori* knowledge of the environment. This method requires a Vertical Line Array (VLA) to collect data from a moving broadband point source at a single depth in order to reconstruct the frequency-wavenumber ($f$-$k$) structure of the waveguide. Our goal is to apply this technique to the more complicated case of bottom reverberation using the VLA as both a source and receiver. As a broadband signal propagates away from the VLA, inhomogeneities along the water/bottom interface scatter some of the field back toward the VLA, mimicking a moving source. However, ambiguities inherent in assigning ranges to the received signal lead to ambiguities in the $f$-$k$ structure. As a preliminary step, we remove these ambiguities by applying the technique to individual point scatterers at various known ranges along the bottom. These results lay a foundation from which to approach the problem of modeling the bottom as a distribution of point scatterers at unknown locations. Theory, simulation, and laboratory measurements are presented.

**11:15**

**5aUW11. Nonlinear scattering of acoustical waves on sphere after passing the obstacle.** Nickolay Zagrai, Andrey Zagrai, and Irene Starchenko (TSURE, 347928, Taganrog, GSP-17a, Nekrasovskiy, 44, Russia, znp@tsure.ru)

The problem of nonlinear interaction of acoustical waves in the system medium-obstacle-medium-sphere was considered. The expressions for secondary field of parametric array were obtained. The contributions of components were evaluated on separated processes of non-linear interaction before and after the obstacle as for the primary signals field as for the secondary fields of combinational frequencies waves. The possible practical applications were also discussed.

**11:30**

**5aUW12. Separability of modal back-scattering matrix in shallow-water reverberation models.** Jinrong Wu, Tianfu Gao, Erchang Shang, and Li Ma (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China; No. 21, West Rd., Beisihuan, Beijing, China)

Normal mode model of the reverberation in shallow-water waveguides has been presented based on Born approximation. The key component of this model is the modal back-scattering matrix. Separability is one of the most important characteristics of the modal back-scattering matrix. The separability or quasi-separability of this matrix could make it easy for modal back-scattering matrix inversion. In this paper, the modal back-scattering matrix due to the roughness of the bottom interface in shallow-water was obtained and its separability error was analyzed. Numerical simulation shows that the separability error increases with modal grazing angle and density ratio of the sediment to the water, while decreases with sound speed ratio of the sediment to the water. It also shows that the effect of the separability error on shallow water reverberation level prediction is negligible. [Work supported by the National Science Foundation of China under Grant No 10474111.]

FRIDAY AFTERNOON, 20 MAY 2005 PLAZA C, 1:00 TO 2:50 P.M.

## Session 5pAB

# Animal Bioacoustics and ASA Committee on Standards: Behavioral Audiometric Methods in Animal Bioacoustics: The Search for Standards II

Edward J. Walsh, Cochair
*Boys Town National Research Hospital, 555 N. 30th St., Omaha NE 68131*

Ann E. Bowles, Cochair
*Hubbs-Sea World Research Inst., 2595 Ingraham St., San Diego, CA 92109*

**Chair's Introduction—1:00**

### *Contributed Papers*

**1:05**

**5pAB1. Dependence of detection threshold estimates in rabbit on method of constant stimulus parameters.** Laurel H. Carney, Yan Gai (Inst. for Sensory Res., Dept. of Biomed. and Chemical Eng., Syracuse Univ., Syracuse, NY 13244), Kristina S. Abrams (Syracuse Univ., Syracuse, NY 13244), Fabio Idrobo (Boston Univ., Boston, MA 02215), and John Michael Harrison (Boston Univ., Boston, MA 02215)

The goal of this study was to estimate behavioral detection thresholds in quiet and noise for subsequent physiological studies of responses to near-threshold stimuli. The difficulty in measuring detection thresholds is maintaining stimulus control of behavior over schedule control. Tracking paradigms are efficient, but the large proportion of trials below threshold weakens stimulus control. The Method of Constant Stimuli controls the proportion of trials below, near, and above the SPL for a given percent correct. Here, percentages of trials at different SPLs and zero intensity catch trials were varied to determine proportions that yielded satisfactory stimulus control by a 500-Hz tone in quiet. Trials were initiated by nose-poke observing responses; a nose-poke reporting response within a 3-s window after tone onset resulted in food reinforcement. Reporting responses in the absence of tones resulted in timeouts. A metric for stimulus control was the ratio of the number of correct detections to the number of responses in the absence of the tone. This ratio was highest when approximately 20% of trials were near the 50%-correct SPL and 80% of trials were 20 dB higher. The ratio was lowest when significant proportions of catch and/or below-threshold trials were included. [Work supported by NIDCD DC-001641.]

**1:20**

**5pAB2. Manatees masked thresholds for complex sounds.** Edmund Gerstein, Laura Gerstein, Joseph Blue (Leviathan Legacy Inc., 1318 SW 14th St., Boca Raton, FL 33486), and Steve Forsythe (Naval Undersea Warfare Ctr., Newport, RI)

Acoustical masked thresholds of complex real world sounds were measured with West Indian manatees. Motor boat and species-specific calls were recorded, filtered and played back in a forced-choice two-alternative test. A method of constant stimuli was used to estimate the 50% detection thresholds of peak spectral components. Both white noise and recorded wild ambient spectra were used as continuous background noise for two sets of threshold estimates. Manatee calls were detected at and within both white noise and wild ambient levels. Thresholds for broadband boat noise ranged from 9 to 29 dB above ambient conditions. Thresholds for idling boat noise were the highest. The levels were consistent with the subjects' auditory limits for the lower frequencies with the spectra. The manatee is well adapted to hear the species specific vocalizations under noisy conditions. The integration of multiple critical bands may result in loudness summation benefits which manatees can exploit at higher frequencies, however, at lower frequencies, near the manatees' limits of sensitivity, benefits of loudness summation were not apparent.

**1:35**

**5pAB3. Testing marine mammal hearing with a vocal response paradigm and the method of free response.** James J. Finneran, Donald A. Carder (U.S. Navy Marine Mammal Program, SPAWARSYSCEN San Diego, Code 2351, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Carolyn E. Schlundt (EDO Professional Services, San Diego, CA 92110), and Sam H. Ridgway (Univ. of California, La Jolla, CA 92093-0612)

Behavioral hearing tests with marine mammals often use single interval experiments with a go/no-go paradigm, where the subject pushes a response paddle or similar device to indicate the presence of a hearing test tone. This approach is widely accepted, but the time required for the subject to physically move and/or interact with the response device and the requirement of one trial per reinforcement interval limit the speed at which data can be collected (typically 20–30 minutes for a threshold). An alternative to the single interval/paddle press approach is the use of a vocal response technique and the method of free response, where multiple trials are presented within a single reinforcement interval. An acoustic or vocal response eliminates the need for the subjects to physically move, allowing for faster responses and shorter inter-trial intervals. The method of free response allows multiple tone presentations between reinforcement periods, dramatically increasing the rate at which data may be collected, especially from diving marine mammals. This talk describes the application of these techniques to hearing tests of bottlenose dolphins, white whales, and California sea lions and the advantages and disadvantages compared to the single interval experiment. [Work supported by ONR.]

**1:50**

**5pAB4. Noise localization ability in the Mongolian gerbil (*Meriones unguiculatus*).** Srijata Sarkar, Kristina S. Abrams, and Laurel H. Carney (Inst. for Sensory Res. and Dept. of Biomed. and Chemical Eng., Syracuse Univ., Syracuse, NY 13244)

Understanding localization ability is essential for comparison to physiological studies of binaural neurons. Heffner and Heffner [Beh. Neuro. **102**, 422 (1998)] reported relatively poor ability of the gerbil to localize a 100 ms noise burst (approximately 75% correct at 27° separation.) Their experimental setup required the gerbil to enter an observing response compartment, initiating a noise burst from one of two speakers separated by the test angle. The animal was required to respond by entering a reporting compartment, positioned 90° to the right or left, regardless of speaker location. This required mapping from speaker location to response location. In this study, response mapping was avoided in an effort to improve performance. After an observing response (jumping on a platform in the center of a circular cage), the animal responded by moving directly towards the active speaker. The results of this study were consistent with those of Heffner and Heffner at small angles of separation. [For 180° separation, this task resulted in poorer performance than the previous study, presumably due to lack of control of head orientation during stimulation, leading to front-back confusions.] In conclusion, the poor localization ability of gerbils is apparently not explained by response mapping. [Work supported by NIDCD R01-DC001641.]

**2:05**

**5pAB5. Strain differences in auditory sensitivity in canaries bred for song and plumage.** Amanda Lauer, Elizabeth Brittan-Powell, Bernard Lohr, and Robert Dooling (Dept. of Psych., Univ. of Maryland, College Park, MD 20742)

Canaries have been domesticated for well over 100 years. They have been bred for specific characteristics such as song, plumage, or body shape. Here we measured audiograms in several canary strains using both behavioral (operant conditioning) and physiological (auditory brainstem responses) methods. Overall, there was a good correspondence between behavioral and ABR audiograms, but there were differences in audiogram shape and sensitivity between strains. Belgian Waterslager canaries bred for low-pitched song had elevated high frequency thresholds, while Spanish Timbrado canaries bred for high-pitched song had slightly better high frequency thresholds than canaries that were not bred for song. Other strains showed intermediate sensitivity. These results suggest there can be more variability in hearing within a species than previously thought and that these differences are correlated with other behavioral or morphological traits. [Work supported by DC00198, DC001372, DC04664-01A2, and DC005450.]

**2:20–2:50**
**Panel Discussion**
**Panelists: Robert Dooling, Richard Fay, Henry Heffner, Paul Nachtigall, Constantine Trahiotis, William Yost.**

FRIDAY AFTERNOON, 20 MAY 2005 PLAZA A, 2:00 TO 4:15 P.M.

## Session 5pBB

## Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Audible-Frequency Medical Diagnostic Methods Including Multimode Techniques II

Hans Pasterkamp, Cochair
*Pediatrics Dept., Univ. of Manitoba, 840 Sherbrooke St., Winnipeg, MB R3A 1R9, Canada*

Thomas J. Royston, Cochair
*Dept. of Mechanical Engineering, Univ. of Illinois at Chicago, 842 W. Taylor St., Chicago, IL 60607-7022*

**Chair's Introduction—2:00**

### *Invited Papers*

**2:05**

**5pBB1. Monitoring changes of human airways by acoustical means.** Hans Pasterkamp (Dept. of Pediatrics, Univ. of Manitoba, CS5120-840 Sherbrook St., Winnipeg, MB, Canada R3A 1S1, pasterkamp@umanitoba.ca)

The state of human airways is typically assessed by spirometry, i.e., by measuring airflow during maximum forced expiration. The required effort excludes subjects who cannot or will not perform such maneuvers. Turbulent airflow during normal breathing generates sound that can be recorded at the chest surface. Changes in airway caliber, e.g., by constriction or dilation, and concurrent changes in the tension of the airway walls affect the relation of airflow and breath sounds. This relation is complex and incompletely understood. However, the possibility to assess changes in the state of human airways by the relation of airflow and breath sounds promises an opportunity to develop auxiliary methods to spirometry. Two regions of the human airways have attracted the interest of researchers in respiratory acoustics. Breath sounds over the trachea (windpipe) increase in their airflow-specific intensity on narrowing of the upper airways, e.g., due to abnormal anatomy, infectious causes etc. Furthermore, spectral characteristics of tracheal sounds may indicate the region of abnormality. On the other hand, breath sounds over the chest change in their airflow-specific intensity with constriction of airways in the lung, e.g., due to asthma. It may therefore be possible to monitor the effectiveness of treatment by acoustical means.

2:30

**5pBB2. A system to evaluate and compare transducers used for lung sound research.** Steve S. Kraman, George R. Wodicka, Gary A. Pressler, and Hans Pasterkamp (Kentucky Clinic J515, Univ. of Kentucky, Lexington, KY 40536) (Purdue Univ., West Lafayette, IN 47907) (Purdue Univ., West Lafayette, IN 47907) (Univ. of Manitoba, Winnipeg, MB, Canada)

Since lung sounds were first recorded and analyzed in the 1960s, a variety of custom made or adapted transducers have been used for detecting these sounds. There is no standard lung sound transducer nor have those in use been adequately compared. To address this problem, a test platform was constructed that provides a stable and reproducible sound source coupled to a viscoelastic surface that is mechanically and acoustically similar to human skin and subcutaneous tissue as judged from comparisons with similar thicknesses of fresh meat and fat. The device was designed to be equally suitable for air-coupled microphones and for accelerometers. When driven by a broadband noise source, the acoustic amplitude at the surface of the platform was found to be relatively uniform from 100 to 1200 Hz. The system was used to evaluate a variety of lung sound transducers. They were found to vary significantly in frequency response with no device appearing to be adequate as an all-purpose lung sound sensor. It is concluded that lung sound characteristics are likely to be significantly affected by the type of sensor used to acquire the sound. This makes comparisons between differently equipped laboratories difficult.

2:55

**5pBB3. Vascular sounds as an indicator of hemodialysis access patency.** Hansen Mansy, Silas Hoxie, Nilesh Patel, and Richard Sandler (Rush Medical College, Chicago, IL 60607)

Background: Vascular access for renal dialysis is a lifeline for about 200 000 individuals in North America. Stethoscope auscultation of vascular sounds has some utility in the assessment of access patency, yet may be highly skill dependent. The objective of this study is to identify acoustic parameters that are related to changes in vascular access patency. Methods: Fifteen patients participated in the study. Their vascular sounds were recorded before and after angiography, which was accompanied by angioplasty in most patients. The sounds were acquired using two electronic stethoscopes and then digitized and analyzed on a personal computer. Results: Vessel stenosis changes were found to be associated with changes in acoustic amplitude and/or spectral energy distribution. Certain acoustic parameters correlated well (correlation coefficient=0.98, $p$ less than 0.0001) with the change in the degree of stenosis, suggesting that stenosis severity may be predictable from these parameters. Parameters also appeared to be sensitive to modest diameter changes (greater than 20%), ($p$ less than 0.005, Wilcoxon rank sum test). Conclusions: Computerized analysis of vascular sounds may have utility in vessel patency surveillance. Further testing using longitudinal studies may be warranted. [Work supported by NIH/NIDDK DK 59685.]

3:20–3:30 Break

## *Contributed Papers*

3:30

**5pBB4. Computational and experimental models for sound transmission in the pulmonary system and chest to aid in medical diagnosis.** Serhan Acikgoz, Thomas J. Royston, M. Bulent Ozer (Univ. of Illinois at Chicago, 842 W. Taylor St. MC 251, Chicago, IL 60607), Hansen A. Mansy, and Richard H. Sandler (Rush Univ. Medical Ctr., Chicago, IL 60612)

Acoustic wave propagation in the pulmonary system and torso is simulated by coupling a numerical acoustic boundary element model that predicts sound propagation throughout the solid tissues with a proven comprehensive analytical model for sound wave propagation in the airways of the bronchial tree that is valid up to at least 2 kHz. This approach enables modeling various pathologies that result in structural changes in the lung and/or changes in breath sound source and strength. The model may be also used to predict the resulting acoustic changes measured by acoustic sensors, e.g., stethoscopes, accelerometers, or other skin-contact sensors. Experimental studies in a novel lung phantom model are used to partially validate the computational model. This study focuses on low audible frequencies, i.e., less than 2 kHz. This range encompasses naturally generated respiratory sounds that have been shown to have diagnostic value, as well as externally-introduced vibro-acoustic stimuli used for diagnosis. [Work supported by NIH EB 003286-01.]

3:45

**5pBB5. Recent developments in the characterization of anisotropic, biological media utilizing magnetic resonance elastography.** Anthony J. Romano, Nicolas P. Valdivia, Philip B. Abraham, Joseph A. Bucaro (Naval Res. Lab., 4555 Overlook Ave, Washington, DC 20375-5320, romano@pa.nrl.navy.mil), Philip J. Rossman, and Richard L. Ehman (Mayo Clinic and Foundation, Rochester, MN 55905)

In this paper, we present recent developments in the characterization of anisotropic, biological media utilizing displacements measured with dynamic MRE. In one approach, a waveguide constrained analysis is applied to physical structures (such as muscle), which evaluates the velocities of wave propagation along arbitrarily oriented fibers or fiber bundles in a local fashion, utilizing a spectral filter and a sliding window, spatial Fourier transform. In a second approach, the anisotropic equations of elasticity, in variational form, are inverted for a determination of the complex moduli comprising the medium. Issues concerning ill-conditioning, the effects of boundary conditions, and multiaspect excitation requirements will be addressed, as well as a discussion concerning the models which are appropriate for various physical situations. These two complementary methods of analysis will be demonstrated utilizing both phantom as well as *in vivo* dynamic displacement measurements.

**4:00**

**5pBB6. Local inversion of transient shear-wave propagation for elasticity and viscosity mapping in soft tissues.** Jeremy Bercoff, Mickael Tanter, and Mathias Fink (Laboratoire Ondes et Acoustique, ESPCI, 10 rue Vauquelin, 75005 Paris, France)

Observation of transient shear-wave propagation in soft tissue is of great interest for the study of tissue viscoelastic properties. In previous work, we introduced a technique, called Supersonic Shear Imaging (SSI), able to generate transient shear waves using the acoustic radiation force and image their propagation in real time in soft tissues. In this work, a local inverse problem of the shear wave propagation permitting the recovery of shear elasticity and viscosity is presented. Compressional and shear waves are decoupled by applying the curl operator to the experimental 2-D displacement field. The role of viscosity on the accuracy of the elasticity estimation is studied. The influence of out of plane shear propagation on the inversion algorithm is discussed. Finally, in media presenting shear viscoelasticity heterogeneities, finite difference simulations are used to study the spatial resolution of the algorithm and its sensitivity to the signal-to-noise ratio. Experiments on calibrated tissue-mimicking phantoms presenting different viscoelastic properties are presented validating the simulation results. First *in vivo* results on female breasts are presented.

FRIDAY AFTERNOON, 20 MAY 2005 REGENCY F, 1:30 TO 4:45 P.M.

## Session 5pNS

# Noise and Psychological and Physiological Acoustics: Workshop on Methods for Community Noise and Annoyance Evaluation II

Brigitte Schulte-Fortkamp, Chair

*Technical Univ. Berlin, Inst. of Technical Acoustics, Secr TA 7, Einsteinufer 25, Berlin 10587, Germany*

### *Invited Papers*

**1:30**

**5pNS1. Auditorium Mundi soundscape in the narration of the world. Artistic and scientific aspects in the presentation of soundscapes.** Harald Brandt (Philippsbergstrasse 40, D-65195 Wiesbaden, Germany, haraldbrandt@yahoo.fr)

Auditorium Mundi is a project for the exploration and presentation of sounds and soundscapes of the whole world. The aim is an installation of a museum of audible perception in interrelation with other senses. The concept of the museum is an acoustic planetarium in which the locations of the world are put in place of the stars. Also located here are world archives for sound in which the acoustic and musical multitude of this planet will be preserved. The archives of sound are available world-wide through the internet. The museum is, at the same time, a laboratory for the development of new formats in scientific definitions and the multimedia presentation of acoustic phenomenons. It is a show-place for artistic programs and an auditorium where questions from the area of acoustic as well as new developments of industry will be accessible to a wider public. A place of innovation, where concepts for other museums, promoters and scientific institutions can be developed. The leitmotif of Auditorium Mundi is the question how man influences the soundscenes in which he lives and how strongly he himself is influenced by the noise, the resonance and the dissonance between various ways of life.

**1:50**

**5pNS2. An $L_{Aeq}$ is not an $L_{Aeq}$.** Dick Botteldooren, Tom De Muer, Bert De Coensel (Acoust. Group, Dept. of Information Technol., Ghent Univ., St. Pietersnieuwstraat 41, B-9000 Ghent, Belgium), Birgitta Berglund (Stockholm Univ., SE-106 91 Stockholm, Sweden), and Peter Lercher (Univ. of Innsbruck, A-6020 Innsbruck, Austria)

Classical dose response relationships for environmental noise annoyance have been based on $L_{dn}$ or $L_{den}$. These exposure measures are essentially based on an energy averaging measure, $L_{Aeq}$. Differences between groups of sources (e.g., continuous or event based) are accounted for by using separate dose-effect relationships. In society today, one often sees that event loudness is traded for number of events which is perfectly acceptable within the $L_{Aeq}$ based annoyance concept. Clearly a more unified theory for noise annoyance is needed to fully account for the effect of such trade-offs. In this paper a model implementing such a theory is presented. The perceptual model starts from the premises that a sound event has to be noticed for it to contribute to overall annoyance. The model accounts for the fact that noticing a noise event not only depends on the level of the event itself but also on background noise, sound insulation and acoustic characteristics of the dwelling, level of attention, etc., the severity of the effect of a noticed sound on overall annoyance is assumed to primarily depend on the signal to noise ratio. The model allows to account for modifiers such as previous exposure, noise sensitivity, and coping. The model results are compared to the findings of a recent field experiment. Conclusions based on calculated and experimental trends will be presented.

**2:10–2:15 Break**

**2:15–4:15**
**Panel Discussion**

FRIDAY AFTERNOON, 20 MAY 2005 REGENCY D, 1:00 TO 5:00 P.M.

## Session 5pSC

## Speech Communication: Speech Production and Perception II (Poster Session)


Bryan W. Gick, Chair
*Dept. of Linguistics, Univ. of British Columbia, 1866 Main Mall E 270, Vancouver, BC V6T 1Z1, Canada*


### *Contributed Papers*

All posters will be on display from 1:00 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 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

**5pSC1. Perceptual development of phonotactic features in Japanese infants.** Ryoko Mugitani (NTT Commun. Sci. Labs., NTT Corp., 2-4 Hikari-dai, Seika-cho, Soraku-gun, Kyoto, 619-0237, Japan), Laurel Fais (Univ. of British Columbia, Vancouver, BC, Canada V6T 1Z4), Sachiyo Kajikawa, Shigeaki Amano (NTT Corp., Seika-cho, Soraku-gun, Kyoto, 619-0237, Japan), and Janet Werker (Univ. of British Columbia, Vancouver, BC, Canada V6T 1Z4)


Acceptable phonotactics differ among languages. Japanese does not allow consonant clusters except in special contexts and this phonotactic constraint has a strong effect on adults speech perception system. When two consonants follow one another in nonsense words, adult Japanese listeners hear illusory epenthetic vowels between the consonants. The current study is aimed at investigating the influence of language-specific phonotactic rules on infants' speech perception development. Six-, 12-, and 18-month-old infants were tested on their sensitivity to phonotactic changes in words using a habituation-switch paradigm. The stimuli were three nonsense words: "keet (/ki:t/)," "keets (/ki:ts/)," and "keetsu (/ki:tsu/)." "Keetsu" perfectly follows Japanese phonotactic rules. "Keets" is also possible in devoicing contexts in fluent speech, but the acceptability of "keets" for adult native Japanese speakers is much less than "keetsu." "Keet" is phonotactically impossible as a Japanese word. The results indicate the existence of a developmental change. Twelve months and older infants detected a change from the acceptable Japanese word "keetsu" to the possible but less acceptable word "keets." However, discrimination between "keets" and the non-Japanese "keet" is seen only in 18-month infants. Implications for infants' speech perception with relation to language specific phonotactical regularities will be discussed.


**5pSC2. Temporal evidence against the production of word-internal syllable structure.** Melissa A. Redford (Dept. of Linguist., 1290 Univ. of Oregon, Eugene, OR 97403, redford@darkwing.uoregon.edu)


A production study examined whether the temporal pattern that cues onset cluster syllabification in English should be attributed to syllable structure per se or to coarticulatory processes that depend on segment identity and sequencing, but not on syllable structure. Experiment 1 tested for independent effects of syllable structure on intervocalic stop-liquid consonant duration. Three speakers produced stimuli in which sequence order, boundary strength, and stress were manipulated. The results showed that stimuli with word-internal sequences were produced with similar stop-liquid duration ($S$:$L$) ratios regardless of sequence order or stress. When a word boundary split the consonants, $S$:$L$ ratios were much smaller, primarily because liquids were longer. Experiment 2 was aimed at distinguishing between alternate explanations for the pattern observed in Experiment 1. Data from three Russian speakers suggested that the English pattern was due to word-internal liquid reduction rather than to boundary lengthening. The $S$:$L$ ratios in Russian were not affected by sequence order or boundary strength. Further, the ratios were significantly different from the word-internal $S$:$L$ ratios of English, but not from the word-peripheral ones. Overall, the results argue against a causal relationship between syllable structure and articulatory timing, and for word-sized articulatory units in English.


**5pSC3. Perception of coarticulated speech with contrastively enhanced spectrotemporal patterns.** Travis Wade and Lori Holt (Dept. Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, twade@andrew.cmu.edu)


High-level contrastive mechanisms cause perception of auditory events to be influenced by spectral and temporal properties of surrounding acoustic context, and may play a role in perceptual compensation for coarticulation in human speech. However, it is unknown whether auditory contrast is incorporated optimally to compensate for different speakers, languages and situations or whether amplification of the processes involved would provide additional benefit, for example, in the perception of hypoarticulated speech, under adverse listening conditions, or in an incompletely acquired language. This study examines effects of artificial contrastive modification of spectrotemporal trajectories on the intelligibility of connected speech in noise by native and non-native listeners. Adopting methods known to improve automatic classification of speech sounds, we model contrast-providing context as an averaged estimated vocal tract function (LPC-derived log area ratio coefficient vector) over a Gaussian-weighted temporal window. Local coefficient values are adjusted from this context based on previously observed contrastive perceptual tendencies, and the intelligibility of the resulting speech is compared with that of unmodified trajectories across listener language backgrounds. Results are discussed with respect to implementation and applicability of general auditory processes.

**5pSC4. Lip and jaw closing gesture durations in syllable final voiced and voiceless stops.** Willis Warren (Univ. of Texas Dept of Linguist., Calhoun 501, 1 University Station B5100, Austin, TX 78712-0198, warisill@mail.utexas.edu) and Adam Jacks (Univ. of Texas, Austin, TX 78712)

It is generally assumed that the longer vowel durations in voiced final consonant contexts can be attributed to longer vowel steady state durations wherein the jaw remains stable for a longer period of time (see diagrams in Chen, 1970; Summers, 1987). Chen (1970) argued that the rate of closure duration was the best explanation for the vowel length effect. However, the closure durational differences he reported were very small relative to the reported differences in vowel duration with an average of 17 ms to 92 ms. Lip and jaw movements for two speakers were measured and synchronized with acoustic data in the CSD department at the University of Texas using the head-mounted lip-jaw movement transducer (HML-JMT) of Barlow *et al.* (1983). This study provides much stronger evidence for Chen's hypothesis, suggesting that there is a more dynamic arc to the jaw. While Chen (1970) gave data showing the closing gesture accounted for less than 20% of the overall difference, the data given here, shows that the duration of the closing gesture can account for 80% of the total vowel durational differences, and the remainder can be accounted for in the opening gesture and the relative timing of vocal cord cessation.

**5pSC5. Assimilation context effects as cue grouping phenomena.** Nicole Couture, Sarah Godbois (Dept. of Psych., Salem State College, 352 Lafayette St., Salem, MA 09170, gow@helix.mgh.harvard.edu), and David W. Gow, Jr. (Massachusetts General Hospital, Boston, MA 02114)

Listeners show bidirectional context effects in the perception of spontaneously assimilated speech. One account of these phenomena, feature cue parsing theory, suggests that both regressive and progressive context effects result from the same perceptual grouping process of aligning feature cues with assimilated word offsets and assimilating word onsets. This hypothesis was examined through two form priming experiments that explored the interpretation of spontaneously assimilated items under manipulations that either favored or disfavored the perceptual grouping of feature cues associated with assimilated and assimilating items. In both experiments, listeners heard phrases such as right berries in which a word final coronal segment (/t/) assimilated the non-coronal place of a subsequent word onset (labial /b/) to form an item that sounded like one of its lexical competitors (ripe). In Experiment 1 spatial continuity was manipulated through the use of same and different side lateralized presentation of assimilated words and their contexts. In Experiment 2 continuity in speaker identity was manipulated through cross-splicing. Analyses contrasted the pattern of priming found in continuous versus non-continuous contexts. The results are discussed in the terms of competing interpretations of assimilation context effects. [Work supported by the NIH.]

**5pSC6. Relationships among perceived sexual orientation, perceived height, and perceived speech clarity.** Benjamin Munson and Elizabeth C. McDonald (Dept. Speech-Lang.-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455 Munso005@umn.edu)

Previous research [B. Munson, Linguist. Soc. Am. (2005)] showed that listeners rate self-identified gay/lesbian/bisexual (GLB) talkers as more-GLB sounding than self-identified heterosexual talkers when listening to readings of single words in isolation. In this investigation, we examine relationships among measures of perceived sexual orientation (PSO) and two other perceptual measures made from the same talkers' speech, perceived height and perceived speech clarity. Ratings were collected from three independent groups of listeners. Regression analyses showed that approximately 80% of the variance in PSO of women's voices was predicted by measures of perceived clarity (women rated as speaking more clearly were more likely to be rated as heterosexual-sounding than women rated as speaking less clearly) and perceived height (women rated as sounding tall were more likely to be rated as GLB-sounding than women rated to sound short). In contrast, 50% of the variance in men's sexual orientation was predicted by measures of perceived clarity (men rated as speaking more clearly were more likely to be rated as GLB-sounding than men rated as speaking less clearly). These results suggest that ratings of PSO from speech may be mediated by judgments of other parameters that are robustly coded in short samples of speech.

**5pSC7. Relative contributions of feedback and editing in language production: Behavioral & articulatory evidence.** Corey T. McMillan, Martin Corley (Univ. of Edinburgh, PPLS, 7 George Square, Rm. F23, Edinburgh, EH8 9JZ, UK, Corey.McMillan@ed.ac.uk), Robin J. Lickley (Queen Margaret Univ. College, Edinburgh, EH12 8TS, UK), and Robert J. Hartsuiker (Univ. of Ghent, 9000 Ghent, Belgium)

Psychologists normally attribute the surfacing of phonological speech errors to one of two factors: editing of the speech plan [Levelt (1989)] or feedback between word and phoneme levels [Dell (1986)]. This paper assesses the relative contributions of each factor, focusing on the perception and articulation of elicited speech errors. Experiments one and two measure the likelihood of phonological exchange errors as a function of phonetic similarity [Frisch (1996)], using the SLIP paradigm and a tongue-twister task. Both experiments show that error likelihood increases with phonetic similarity between intended and actual utterance, an effect easy to account for in terms of feedback but not in terms of editing. Experiment three uses EPG to analyze the tongue-twister utterances: many errors occur at the articulatory level but are not easily perceived in the speech signal. Preliminary analysis suggests three patterns of error: (1) substitution of segments, which may be the result of editing; (2) simultaneous double articulation, hypothesized to be the result of residual activation due to feedback; and (3) overlapping double articulation, representing partial execution of one articulation before substitution with another. Taking these findings together, we hope to evaluate the relative contributions of editing and feedback to phonological speech errors.

**5pSC8. Acoustical study of the development of stop consonants in children.** Annika K. Imbrie (Speech Commun. Group, MIT, Rm. 36-545, 77 Massachusetts Ave., Cambridge, MA 02139, imbrie@mit.edu)

This study focuses on the acoustic patterns of stop consonants and adjacent vowels as they develop in young children (ages 2;6-3;3) over a six month period. Over forty different acoustic measurements were made on each utterance, including durational, amplitude, spectral, formant, and harmonic measurements. These acoustic data are interpreted in terms of the supraglottal, laryngeal, and respiratory actions that give rise to them. Data show that some details of the child's gestures are still far from achieving the adult pattern. Significant findings include a high number of bursts in children's stop productions, possibly due to greater compliance of the active articulator. The incidence of multiple bursts decreases over the period of the study. Measurements of the burst spectra show that the production of all three places of articulation is close to normal. Finally, children's voiced stop productions display a delayed onset of voicing for all three places of articulation, indicating difficulty with coordination of closure of the glottis and release of the primary articulator. This measurement is found to decrease toward adult values over six months. Analysis of longitudinal data on young children will result in better models of the development of motor speech production. [Work supported by NIH grant DC00075.]

**5pSC9. Relations between speech sensorimotor adaptation and perceptual acuity.** Virgilio Villacorta, Joseph Perkell, and Frank Guenther (Res. Lab of Electron., Mass. Inst. of Tech., 50 Vassar St., Ste. 36-511, Cambridge, MA 02139)

The goal of this research is to study the auditory component of feedback control in speech production. This experiment investigates auditory sensorimotor adaptation (SA) as it relates to speech production: the pro-

cess by which speakers alter their speech production in order to compensate for perturbations of normal auditory feedback. Specifically, the first formant frequency ($F1$) was shifted in the auditory feedback heard by naive adult subjects as they produced vowels in single syllable words. Results presented previously indicate that the subjects demonstrate compensatory formant shifts in their speech. The current study investigates the relationship between compensatory adaptation and speaker perceptual acuity. Subjects from the SA study were tested for their ability to discriminate vowel tokens differing in $F1$ frequency. Preliminary results indicate that subjects with greater perceptual acuity also demonstrated greater ability to adapt, with a significant cross-subject correlation ($p$ less than 0.05). The relation between these experimental findings and DIVA, a neurocomputational model of speech motor planning by the brain, will also be discussed. [Work supported by NIDCD Grant R01-DC01925.]

**5pSC10. Glottal-wave and vocal-tract-area-function estimations from vowel sounds based on realistic assumptions and models.** Huiqun Deng, Rabab K. Ward, Micheal P. Beddoes (Elec. and Comput. Eng. Dept., Univ. of British Columbia, Canada), Murray Hodgson, and Bryan Gick (Univ. of British Columbia, Canada)

Estimating glottal waves by inverse filtering vowel sounds and deriving vocal-tract area functions (VTAFs) from vocal-tract filter (VTF) estimates require that VTF models be realistic and that VTF estimates contain no effects of open glottises and glottal waves. In this study, VTFs are modeled to have lip reflection coefficients with low-pass frequency responses; to minimize the effects of open glottises and glottal waves on the estimates, VTFs are estimated from sustained vowel sounds over closed glottal phases, assuming that the glottal waves are periodically stationary random processes. Since incomplete glottal closures are common, VTF estimates may contain the effects of glottal loss. To eliminate the effects of glottal loss in the VTF estimates, lip-opening areas must be known. Theoretically, estimates of glottal waves and VTAFs corresponding to large-lip-opening vowel sounds are less affected by the glottal loss than those corresponding to small-lip-opening vowel sounds. The VTAFs and glottal waves estimated from vowel sounds produced by several subjects are presented. The normalized VTAFs estimated from large-lip-opening sounds are similar to that measured from an unknown subjects magnetic resonance image. Over closed glottal phases, the glottal waves are non-zero. They increase during vocal-fold colliding, and decrease or even increase during vocal-fold parting.

**5pSC11. Cross-modal perception of vowels, stops, and approximants using reduced-formant stimuli.** Philip Rubin (Haskins Labs., 270 Crown St., New Haven, CT 06511-6204 and Yale Univ., School of Medicine, New Haven, CT, rubin@haskins.yale.edu), Catherine Best (Haskins Labs., New Haven, CT 06511-6204), Gordon Ramsay, Stephen Frost, and Bruno Repp (Haskins Labs., New Haven, CT 06511-6204)

This paper explores the relationship between audio and video representations of speech, by examining how information in particular formants is complemented by facial information, for different classes of speech sounds. Specifically, we investigate whether acoustic information removed by deleting formant combinations that usually signal contrasts between vowels ($F1$,$F2$), stops ($F2$) and approximants ($F3$) can be replaced by optical information derived from the talking face. Simultaneous audio and video recordings were made of speakers uttering CVC nonsense syllables constructed from vowels, stops, and approximants exhibiting varying degrees of facial motion and varying dependence on individual formant cues. Synthetic stimuli were then resynthesized from combinations of one, two, and three formants using a parallel formant synthesizer. Subjects performed an auditory identification task with stimuli presented in an audio-only condition, followed by a separate block of stimuli presented in an audio-visual condition. Source effects were examined using an inverse-filtered source and a synthetic source with constant source characteristics. There were no differences between source conditions, but differences were obtained between the audio and audio+video conditions that reflect the relationship between facial and formant dynamics. These results complement recent research on perception of reduced video information and complete audio information [L. Lachs and D. B. Pisoni, J. Acoust. Soc. Am. **116**, 507–518 (2004)]. [Work supported by NIH.]

**5pSC12. Analysis of vowel and speaker dependencies of source harmonic magnitudes in consonant-vowel utterances.** Markus Iseli, Yen-Liang Shue, and Abeer Alwan (UCLA, 405 Hilgard Ave., Los Angeles, CA 90095, iseli@ee.ucla.edu)

It is assumed that voice quality characteristics are mainly manifested in the glottal excitation signal. As [Holmberg *et al.*, J. Speech Hear. Res. **38**, 1212–1223 (1995)] showed, there is a correlation between low-frequency harmonic magnitudes of the glottal source spectrum and voice quality parameters. In this study, we assess the influence of vowel and speaker differences on the difference between the first and the second harmonic magnitudes, $H_1-H_2$. The improved harmonics correction formula introduced in [Iseli *et al.*, Proceedings of ICASSP, Vol. 1 (2004), pp. 669–672] is used to estimate source harmonic magnitudes. $H_1-H_2$ is estimated for consonant-vowel utterances where the vowel is one of the three vowels /a,i,u/ and the consonant is one of the six plosives in American English /b,p,d,t,g,k/. Several repetitions of each of the utterances, spoken by two male and two female talkers, are analyzed. Other measurements, such as fundamental frequency, $F_0$, and energy are estimated pitch synchronously. Results are compared to current literature findings. [Work supported by NSF.]

**5pSC13. Pharynx depth and tongue height in relationship to intrinsic $F0$ of vowels.** D. H. Whalen, Leonardo Oliveira, and Bryan Gick (Haskins Labs, 270 Crown St., New Haven, CT 06511, whalen+AEA@haskins.yale.edu)

Intrinsic $F0$ (I$F0$) is the tendency for high vowels to have higher $F0$s than low vowels. It is universal and apparently automatic, but the exact mechanism is still unclear. Here, ultrasound measurements of the tongue were used to determine whether vowel height is the best predictor of I$F0$. Ten native speakers of English spoke keywords for the eleven non-diphthongal vowels while the lips, jaw, and tongue were imaged with the HOCUS ultrasound/Optotrak system [D. H. Whalen *et al.*, J. Sp. Lang. Hear. Res. (in press)], which tracks the head and lips with optical markers while the tongue is imaged with ultrasound. The tongue edge can be extracted and rotated relative to the head, indicating changes in tongue height and pharyngeal depth. Preliminary analysis indicates that I$F0$ is better correlated with the most anterior point on the tongue root (indicating pharyngeal depth) than with the highest point on the tongue. It thus seems likely that the shape of the pharynx is more of a determinant than tongue height for I$F0$. Because pharynx depth and vowel height covary in most languages (including English), the earlier description is somewhat accurate; I$F0$ may be more accurately described as correlated to the tongue root.

**5pSC14. Acoustically-guided articulation patterns for vowel production.** Brad Story (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, P.O. Box 210071, Tucson, AZ 85721, bstory@u.arizona.edu)

The purpose of this study was to investigate the link between the vocal tract deformation patterns obtained from statistical analyses of articulatory data (i.e., measured area functions) and the acoustic properties of a realistic neutral vocal tract shape. The starting point was a previously reported statistical analysis (i.e., PCA) of a speaker-specific set of measured vowel area functions. It resulted in a mean area function, and two basis functions that perturb the vocal tract into vowel-like shapes. In the present study, acoustic sensitivity functions were calculated for the same mean area function. Then, four separate sum and difference combinations of the sensitivity functions were used directly to perturb the mean area function. The results showed that these sum and difference combinations created deformation patterns with nearly the same variation along the vocal tract length as the basis functions derived from articulatory data. The procedure was then applied to other speaker-specific sets of area functions and the results were similar. [Work supported by NIH R01-DC04789.]

**5pSC15. The effect of spatial frequency information in central and peripheral vision on natural gaze patterns and audiovisual speech perception.** Julie N. Buchan, Amanda Wilson, Martin Paré, and Kevin G. Munhall (Queen's Univ., Kingston, ON, Canada, 2jnb@qlink.queensu.ca)

It has been known for some time that visual information plays an important role in how we perceive speech. The present research examines how this visual information is processed in central and peripheral vision. The perception of visual images is carried out in the nervous system by a set of spatial frequency-tuned channels and the sensitivity to spatial resolution varies across the retina. The experiment was conducted using a gaze-contingent display system that records a person's eye position and then displays different information to their peripheral and central vision. The amount of high spatial frequency information presented in the central and peripheral visual fields was manipulated and the effect on both audiovisual speech perception and natural gaze patterns was measured. Preliminary results show that when peripheral information is no longer adequate, gaze patterns become altered to gather important visual information. This suggests that information across the whole visual field influences audiovisual speech behavior.

**5pSC16. Perception of a Japanese moraic obstruent in an identification task.** Shigeaki Amano, Ryoko Mugitani, and Tessei Kobayashi (NTT Commun. Sci. Labs., NTT Corp., 2-4 Hikari-dai, Seika-cho, Souraku-gun, Kyoto 6190237, Japan)

Native Japanese speakers perceive a moraic obstruent when the closure duration between successive moras is longer than a normal obstruent. However, it has not been well clarified how the perception of the moraic obstruent relates to the closure and neighboring moras' duration. To investigate this point, a perceptual experiment was conducted with stimuli consisting of two-mora nonsense syllables with a closure between the moras (/bipa/, /guku/, /kuku/, /kuto/, and /tapi/). The closure duration was shortened or lengthened in 10-ms steps. The duration of the first mora was modified by changing its vowel duration to 50%, 100%, and 150% by the STRAIGHT method. Forty native Japanese speakers identified whether the nonsense syllable contained a moraic obstruent with the 2AFC method. Results showed that the perceptual boundaries of the moraic obstruent were 182 ms $(SD=13.2\ ms)$, 204 ms $(SD=14.7\ ms)$, and 222 ms $(SD=16.6\ ms)$, respectively, with 50%, 100%, and 150% vowel durations in the first mora, and that a logistic function fits the identification ratio very well. The results indicated that the perception of a moraic obstruent depends on both closure duration and the duration of neighboring moras. They also suggest that the perception would be categorical.

**5pSC17. Nasals and nasalization: The interplay between segmental duration and coarticulation.** Samantha E. Sefton and Patrice S. Beddor (Dept. of Linguist., Univ. of Michigan, 4080 Frieze Bldg., Ann Arbor, MI 48109, beddor@umich.edu)

Two experiments were conducted on English to test the hypothesis that the temporal extent of coarticulatory nasalization is inversely related to the duration of the source of coarticulation. Experiment 1 investigated patterns of carryover (NVC) and anticipatory (CVN) vowel nasalization, and the overall timing of nasalization across the syllable. Acoustic analysis of CVN and NVC words (e.g., seen/niece, pin/nip, tame/mate, ten/net) produced by five speakers showed that, although vowel nasalization is temporally more extensive in CVN than in NVC sequences, *N* is shorter in CVN than in NVC, so that there was no significant overall difference in total nasalization between the two syllable types. Experiment 2 tested similar timing patterns in CVNC syllables, where *N* duration varied as a function of the voicing of the following *C* (e.g., wince/wins, sent/send, can't/canned). Acoustic measures of productions of five English speakers again showed that the shorter the *N*, the temporally more extensive the vowel nasalization. Findings for selected other languages (Thai, Italian) provide further evidence of the inverse relation. These findings have implications for theories of coarticulation and theories of phonological change. [Work supported by NSF.]

**5pSC18. Perception of a Japanese moraic obstruent in a discrimination task.** Tessei Kobayashi, Ryoko Mugitani, and Shigeaki Amano (NTT Commun. Sci. Labs., NTT Corp., 2-4, Hikaridai, Seika-cho, Soraku-gun, Kyoto, Japan 619-0237)

Previous research with an identification task suggests that native Japanese speakers perceive a moraic obstruent categorically (Amano et al., 2005). However, it is still unclear whether the suggestion would be confirmed by other tasks. This study used an AX discrimination task to investigate whether the performance is highly sensitive around the perceptual boundaries that were obtained from the identification task. Native Japanese speakers $(N=40)$ were presented with a pair of two-mora nonsense syllables (/bipa/, /guku/, /kuku/, /kuto/) and then required to judge whether the stimulus pair was acoustically identical. The stimulus set was produced by reducing or increasing the closure duration between the successive moras in 5-ms steps ($-95$ to 240 ms). Results showed that the participants discriminated more precisely around closures duration ranging from approximately 150 to 180 ms (/bipa/: 180 ms, /guku/: 160 ms, /kuku/: 155 ms, /kuto/: 175 ms). Although the results indicate, to some degree, the possibility of perceptual categorization of the moraic obstruent, the high sensitivity peak in each stimulus type was slightly shifted from the perceptual boundaries in the identification task.

**5pSC19. Intrinsic factors of releasing motions in an articulator: On assuming segmental input in speech-production models.** Victor J. Boucher (Univ. of Montreal, C.P. 6128 succ. Centre-ville, Montreal, QC, Canada J3C 3J7)

Prominent models of speech production use serial input that is assumed to be commensurate with linguistic segments. The view is that such units underlie a serial activation of aperture motions such as closing and opening motions of the lips in articulating a bilabial stop. This is incompatible with conventional EMG observations showing a single burst of activity of labial adductors at the onset of a close-open cycle. The present study examines the spring-like effects of bilabial compression and pressure on labial opening (release) following a relaxation of the orbicularis oris muscle. Using reiterative series [papapapa] produced at increasing intensities, the range and velocity of opening motions of the lower-lip were correlated with lip compression and oral-pressure. The results for three speakers show that pressure and compression are correlated and that these factors account for 45% to 66% of the variance in velocity of lower-lip opening, and for 47% to 73% of the variance in the range of lower-lip opening. These results complement earlier findings of Abbs and Eilenberg (1976) showing the intrinsic effects of muscle elasticity on opening motions of the lips. Close-open cycles in articulators may not reflect segment-by-segment serial activation.

**5pSC20. Kinematic properties of stop and fricative production.** Anders Lofqvist (Haskins Labs., 270 Crown St., New Haven, CT 06511, lofquist@haskins.yale.edu)

This study examined movement kinematics in stop and fricative consonants, looking for potential evidence of different movement characteristics related to precision requirements. It has often been claimed that fricatives require more precision in their production than stops due to the management of airflow. However, the empirical evidence for this claim is very limited and mostly based on different developmental trends. Tongue movements were recorded using a magnetometer system. Speed and movement paths were used to examine movement kinematics, including different measures of stiffness. Preliminary results do not show any consistent pattern across subjects, although some subjects did have systematic differences in movement stiffness with stop consonants having stiffer movements that fricatives. These results most likely reflect that fact that speech is an over learned skill, so that any differences related to precision are very small, or nonexistent, in adults, and probably only apparent in the early stages of speech development. [Work supported by NIH.]

**5pSC21. Haptic information enhances auditory speech perception.** Kristin M. Johannsdottir, Diana Gibrail, Gick Bryan, Ikegami Yoko (Dept. of Linguist., Univ. of British Columbia, Canada), and Jeff Muehlbauer (Univ. of British Columbia, Canada)

Studies of tactile enhancement effects on auditory speech perception [Reed *et al.*, JSHR, 1978, 1982, 1989] have traditionally used experienced, pre-trained subjects. These studies have left unanswered the basic questions of whether tactile enhancement of speech is a basic component of human speech perception or a learned association [Fowler, JPhon, 1986; Diehl and Kluender, Ecol. Psych., 1989], and which aspects of the signal are enhanced through this modality. The present study focuses exclusively on tactile enhancement effects available to naive speakers. In a speech-masked environment, naive subjects were tasked with identifying consonants using the Tadoma method. Half of the stimuli were presented with tactile input, and half without. The results show that all subjects gained considerably from tactile information, although which part of the tactile signal was most helpful varied by subject. The features that contributed most to consonant identification were aspiration and voicing. [Work supported by NSERC.]

**5pSC22. Categorization of spectrally complex non-invariant auditory stimuli in a computer game task.** Lori Holt and Travis Wade (Dept. Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, twade@andrew.cmu.edu)

This study examined perceptual learning of spectrally complex non-speech auditory categories in an interactive multi-modal training paradigm. Participants played a computer game in which they navigated through a three-dimensional space while responding to animated characters encountered along the way. Characters appearances in the game correlated with distinctive sound category distributions, exemplars of which repeated each time the characters were encountered. As the game progressed, the speed and difficulty of required tasks increased and characters became harder to identify visually, so quick identification of approaching characters by sound patterns was, although never required or encouraged, of gradually increasing benefit. After thirty minutes of play, participants performed a categorization task, matching sounds to characters. Despite not being informed of audio-visual correlations, participants exhibited reliable learning of these patterns at post-test. Categorization accuracy was related to several measures of game performance and category learning was sensitive to category distribution differences modeling acoustic structures of speech categories. Category knowledge resulting from the game was qualitatively different from that gained from an explicit unsupervised categorization task involving the same stimuli. Results are discussed with respect to information sources and mechanisms involved in acquiring complex, context-dependent auditory categories, including phonetic categories, and to multi-modal statistical learning.

**5pSC23. Acoustic correlates of non-modal phonation in telephone speech.** Tae-Jin Yoon (Dept. of Linguist., Univ. of Illinois, 4080 FLB, 707 S. Mathews Ave., MC-168, Urbana, IL 61801, tyoon@uiuc.edu), Jennifer Cole, Mark Hasegawa-Johnson, and Chilin Shih (Univ. of Illinois, Urbana, IL 61801)

Non-modal phonation conveys both linguistic and paralinguistic information, and is distinguished by acoustic source and filter features. Detecting non-modal phonation in speech requires reliable $F0$ analysis, a problem for telephone-band speech, where $F0$ analysis frequently fails. An approach is demonstrated to the detection of creaky phonation in telephone speech based on robust $F0$ and spectral analysis. The $F0$ analysis relies on an autocorrelation algorithm applied to the inverse-filtered speech signal and succeeds in regions of non-modal phonation where the non-filtered $F0$ analysis typically fails. In addition to the extracted $F0$ values, spectral amplitude is measured at the first two harmonics ($H1$, $H2$) and the first three formants ($A1$, $A2$, $A3$). $F0$ and spectral tilt are measured from 300 samples of modal and creaky voice vowels, selected from Switchboard telephone speech using auditory and visual criteria. Results show successful $F0$ detection in creaky voice regions, with distinctive low $F0$, and statistically significant differences between modal and creaky voice in measures of spectral amplitude, especially for measures based on $H1$. Our current work develops methods for the automatic detection of creaky voicing in spontaneous speech based on the analysis technique shown here. [Work supported by NSF.]

**5pSC24. A consonant-vowel-consonant display as a computer-based articulation training aid for the hearing impaired.** Fansheng Meng, Eugen Rodel, and Stephen Zahorian (Dept. of Elec. and Comput. Eng., Old Dominion Univ., Norfolk, VA 23529)

Computer-based visual speech training aids are potentially useful feedback tools for hearing impaired people. In this paper, a training aid for the articulation of short Consonant-Vowel-Consonant (CVC) words is presented using an integrated real-time display of phonetic content and loudness. Although not yet extensively tested with hearing-impaired listeners, real-time signal processing has been developed, flow-mode displays have been developed, and accuracy tests have been completed with a large database of CVC short words. A combination of Hidden Markov Models and time-delay neural networks are used as the primary method for the acoustic-phonetic transformations used in the display. [Work partially supported by NSF grant BES-9977260.]

**5pSC25. Stop consonant perception in silent-center syllables.** Mark DeRuiter (Univ. of Minnesota, 164 Pillsbury Dr. S.E., 115 Shevlin Hall, Minneapolis, MN 55455), Virginia S. Ramachandran, and Sheryl J. Rosen (Wayne State Univ., Detroit, MI 48202)

Silent-center (SC) syllables are consonant-vowel-consonant syllables with the steady-state vowel portion excised from them. These syllables are appealing for research of stop-consonants because they retain the complexity, brevity, and rapid changes inherent in formant transitions while presumably eliminating temporal masking by the vowel. However, questions exist as to whether or not SC syllables are processed in the same manner as their full-vowel (FV) counterparts (i.e. are they processed as speech units or as sounds?). Data is reviewed from a series of experiments which examined listeners discrimination, labeling, and response time for synthesized consonant-vowel-consonant syllables in FV and SC conditions. Results from 3 experiments with typical listeners reveal that: (1) discrimination is significantly better in the SC condition; (2) consonant labeling on a /bab/ to /dad/ continuum is poorer in the SC condition and is significantly poorer at the category boundary; (3) discrimination response-time (RT) is significantly shorter in the SC condition. Labeling and discrimination results reveal that listeners processed the stop-consonants in these SC syllables in a less-categorical manner than in the FV syllables. Taken together with significantly different response-times, these results may indicate that listeners utilize a different mode of processing for SC syllables.

**5pSC26. Voice $F0$ responses elicited by perturbations in pitch of auditory feedback during English speech and sustained vowels.** Jay J. Bauer (Dept. of Commun. Sci. & Disord., Univ. of Wisconsin–Milwaukee, P.O. Box 413, Milwaukee, WI 53201, jbauer@uwm.edu) and Charles R. Larson (Northwestern Univ., Evanston, IL 60208)

Twenty-one native speakers of American English were asked to repeatedly inflect the pitch of either the first or second syllable of an English speech phrase or produce non-word sustained vowels while listening to amplified auditory feedback. Brief upward and downward perturbations in pitch of auditory feedback were introduced in real time shortly after vocal onset. Resultant changes in voice $F0$ due to perturbations in pitch were compared across vocal conditions and perturbation direction. Data indicated that auditory feedback was used in real-time to stabilize voice $F0$ during both normal English speech and sustained vowels. Incomplete compensatory corrections in voice $F0$ due to the perturbations in pitch of auditory feedback were prevalent during both dynamic speech (gain: $\sim 12.7\%$) and static sustained vowel phonations (gain: $\sim 19.3\%$), and appear to be regulated by similar corrective mechanisms. However during dynamic speech, voice $F0$ was modulated across syllable boundaries in a task-dependent manner according to syllable pitch inflection. Thus, voice $F0$ responses appear to help maintain the underlying suprasegmental meaning of a speech phrase by delaying the onset of the voice $F0$ response to coincide with the pitch-inflected syllable. [Work supported by NIH Grant No. F31 DC005874.]

**5pSC27. The perception of speech rhythm: An investigation of inter and intra rhythmic class variability using delexicalized stimuli.** Volker Dellwo (Dept. of Phonet. and Linguist., University College London, 4 Stephenson Way, London NW1 2HE, UK) and Petra Wagner (Universitaet Bonn, D-53115 Bonn, Germany)

It has been demonstrated that stress- and syllable-timed languages are differentiable in terms of parameters based on vocalic and consonantal interval durations, and that stress-timed languages are more variable in terms of these parameters. There is also significant between- and within-subject variability on these parameters, but it is unclear whether this production variability is perceptually salient in terms of rhythm. This issue was investigated in a series of experiments using delexicalized stimuli, in which the vocalic intervals were substituted by sinusoidal tones and the consonantal intervals by white noise. Subjects were asked to classify the stimuli according to the regularity of the tone sequences. The results thus far indicate that intra rhythm class variation, as measured by consonantal and vocalic interval duration variation, can be well perceived.

**5pSC28. Palato-lingual contact patterns during Korean obstruent production; lax, aspirated and forced.** Hirohide Yoshioka and Kim Soone (Inst. of Human Sci., Univ. of Tsukuba, Japan)

In Korean obstruents, there is a three-way distinction in manner of articulation; lax, aspirated and forced. The present study is to investigate the temporal and spatial details of tongue contacts with the hard palate during these three types of consonant production. The subject, a 30-year-old female of the Seoul dialect was instructed to read a set of various words beginning with these consonants, embedded in a frame sentence i-geo-/Cal/-ita (C=consonant). The palato-lingual contact patterns were recorded using dynamic electro-palatography. The results show that the area and duration of the palato-lingual contact are clearly different among these three types of consonants. As for stops, the palato-lingual contact is wider and longer in the forced stop production. In the affricate production, the patterns of the maximum contact area and the duration are as follows: wide and long in the forced type, middle in the aspirated, and narrow and short in the lax. These data strongly suggest that the temporal and spatial details of the palato-longual contact may play an important role in manifesting the Korean three-way distinction, together with other articulatory adjustments, such as laryngeal and respiratory controls.

**5pSC29. Vowel production and perception in French blind and sighted adults.** Sophie Dupont and Lucie Menard (Universite du Province Quebec a Montreal, Departement de linguistique, CP 8888, succ. Centre-Ville, Montreal, Canada H3C 3P8, menard.lucie@uqam.ca)

It is well known that visual cues play an important role in speech perception and production. Early in life, blind speakers, who do not have access to visual information related to lip and jaw movements, show a delay in the acquisition of phonological contrasts based on these visible articulatory features [A. E. Mills, *Hearing by eye: The psychology of lip-reading*, pp. 145–161 (1987)]. It has also been claimed that blind speakers have better auditory discrimination abilities than sighted speakers. The goal of this study is to describe the production-perception relationships involved in French vowels for blind and sighted speakers. Six blind adults and six sighted adults served as subjects. The auditory abilities of each subject were evaluated by auditory discrimination tests (AXB). At the production level, ten repetitions of the ten French oral vowels were recorded. Formant values and fundamental frequency values were extracted from the acoustic signal. Measures of contrasts (Euclidean distances) and dispersion (standard deviations) were computed and compared for each feature (height, place, roundedness) and group (blind, sighted). Regression analyses between discrimination scores and produced contrasts were carried out. Despite between-speaker variability, results show an effect of speakers group (blind versus sighted) on the produced contrast distances.

**5pSC30. The role of intonational phrasing in the elicitation of speech errors.** Marianne Pouplier (TAAL, Univ. of Edinburgh, UK and Haskins Labs, New Haven, CT, pouplier@ling.ed.ac.uk), Mark Tiede (Haskins Labs, New Haven, CT), Stefanie Shattuck-Hufnagel (MIT R.L.E., Cambridge, MA), Man Gao, and Louis Goldstein (Yale Univ., New Haven, CT)

Recent work has proposed that some speech errors arise from an underlying coordination process in speech production in which gestures can assume grammatically illegal but dynamically stable coordination modes. The general premise is that shared gestural structure within a prosodic domain sets up the conditions under which stable coordination modes spontaneously emerge and produce errors. If certain gestures (e.g., coda consonants) recur every word but others recur less frequently (e.g., alternating initial consonants), the lower frequency gestures will increase in occurrence, resulting in gestural intrusion errors. This suggests that manipulation of the rhythmic domain through intonational phrasing should determine error patterns differentially. Since the gestural coupling which lies at the heart of the rhythmic synchronization approach takes time to establish, it is predicted that continuous repetitions of e.g. ‘‘cop tap tube cub’’ will lead to a maximum buildup of coupling strength and thus the most errors. If however phrasal grouping is imposed, as in ‘‘cop tap, tube cub,’’ fewer errors are hypothesized to occur. We test these predictions by having subjects repeat the same word string in different prosodic grouping conditions. Our initial results confirm that the more alternating consonants an intonational phrase contains, the more errors occur.

**5pSC31. Effect of aspiration noise and spectral slope on perceived breathiness in vowels.** Rahul Shrivastav and Maria Pinero (Dept. of Comm. Sci. & Disord., Univ. of Florida, 336 Dauer Hall, Gainesville, FL 32611)

Breathy voice quality is frequently encountered in both normal and dysphonic voices. Breathiness has been associated with an increase in the intensity of aspiration noise as well as changes in spectral slope [Childers and Ahn, 1995; Fischer-Jorgenson, 1967; Huffman, 1987; Klatt and Klatt, 1990]. Shrivastav and Sapienza (2003) found that subjective ratings of breathiness obtained from a group of listeners were highly correlated with certain measures calculated from the auditory spectrum of the vocal acoustic signal. One of these measures, the partial loudness of the harmonic energy, was related to both the aspiration noise and the spectral slope of the voices. In the present experiment, 10 young adult listeners with normal hearing were presented with voices that varied systematically in terms of their aspiration noise and spectral slope. The stimuli, five male and five female voices, were generated using the Klatt synthesizer and were modeled after naturally occurring voices. Listeners rated breathiness for each of these stimuli using a 7-point rating scale. Results show the relative contribution of spectral slope and aspiration noise to the perception of breathiness.

**5pSC32. Intensity variation in vowels across acoustic and auditory spectra.** Ewa Jacewicz and Robert Allen Fox (Dept. of Speech and Hearing Sci., The Ohio State Univ., Columbus, OH 43210, jacewicz.1@osu.edu)

Changes to vowel intensity are affected primarily by an increased physiological effort of the speaker. A second source of intensity variation comes from immediate consonant environment of the vowel as a result of coarticulation [House & Fairbanks, J. Acoust. Soc. Am. **22**, 105–113 (1953)]. Variation in intensity distribution across the vowel spectrum was measured for eight American English vowels in ten symmetrical CVC environments (consonants included both stops and fricatives) at four different locations (at the vowels rms peak and at points corresponding to 20%, 50%, and 80% of the vowels duration). Three types of spectral intensity distribution were examined: (1) the relative amplitudes of formants $F1$-$F4$; (2) the summed intensity in four contiguous frequency bands of 0–0.5, 0.5–1.0, 1.0–2.0 and 2.0–4.0 kHz [Sluijter and Van Heuven, J. Acoust. Soc. Am. **100**, 2471–2485 (1996)]; and (3) the intensity distribution following three stages of auditory processing (using an auditory filter bank consisting of 33 filters, equal loudness pre-emphasis, and intensity-loudness compression). The nature and size of the effects of spectral intensity variation as a function of consonant environment in the acoustic spectrum (formants) and in auditory pre-processing of the acoustic spectrum will be compared. [Work supported by NIDCD R03 DC005560-01.]

**5pSC33. Voice onset time is shorter in high-frequency words.** Mark VanDam and Robert Port (Dept. of Linguist., Indiana Univ, Bloomington, IN 47405)

Frequency of occurrence is known to have many effects on speech production [see J. Bybee, *Phonology and Language Use* (Cambridge, 2001)] including vowel quality, overall duration, rate of deletion, assimilation, coarticulation, etc. The current work addresses voice-onset time (VOT) in words with differing lexical frequency estimates from published materials and addresses whether words in a list exhibit similar effects to words in sentential context. Four talkers produced 20 low frequency words and 10 high frequency words four times each in isolation and again in non-idiomatic, sentential context. VOT was measured in monosyllabic content words with initial /t/. Results show that frequent words (e.g., *talk, table*) have a mean VOT roughly 10 ms shorter than less frequent words (e.g., *talc, taint*) ($p<0.01$). The effect was significantly stronger for words in a sentence (e.g., *He will talk to his supervisor*) than in a list. These findings are consistent with linguistic theories that propose detailed auditory representations for words and finely controlled productions, but are in conflict with traditional, abstract phonological representations.

**5pSC34. Tension asymmetries in a finite element model of the vocal folds.** Greg S. Davidson (Univ. of Chicago, Chicago, IL 60637) and Fariborz Alipour (Univ. of Iowa, Iowa City, IA 52242)

Tension asymmetries in a finite element model of the vocal folds were examined as a function of lung pressure. The vocal fold model had an asymmetry in the tension ratio ranging from 1.1 to 2.0, corresponding to a 10%–100% increase in tension of the left fold compared to the right, and lung pressure was increased over the range 5–40 cm $H_2O$. For tension ratios greater than 1.6, oscillation was not supported over most of this range. For tension ratios less than 1.6, the fundamental frequency was found to increase up to approximately 22 cm $H_2O$. Mean vocal fold contact area showed a broad maximum over the range 8–13 cm $H_2O$ and decreased between 13 and 22–25 cm $H_2O$. Open quotient decreased between 7.5 and 13–15 cm $H_2O$ and increased afterwards to reach a maximum value between 20 and 23 cm $H_2O$. For these three parameters, the behavior above the upper thresholds was tension ratio dependent. Mean glottal area linearly increased with lung pressure with differing slopes over different pressure ranges. Taken together, the four parameters provide evidence for different vibratory domains. [Work supported by NIDCD grant No. DC03566.]

**5pSC35. Not just any contour: Intonation marking semantic domain narrowing.** Karsten A. Koch (Univ. of British Columbia, Vancouver, BC, Canada)

*Not just any* statements (*Natalie won't drink just ANY wine from France*) carry an intonational contour. Previous studies have characterized the contour as fall-rise [D. Robert Ladd, *The structure of intonational meaning* (1980)], but more intricate acoustic analysis that considers the semantics of the construction has not been performed. Preliminary data shows that for speakers of Canadian English, the contour contains a rise-fall and stress accent on *any*, followed by a rise on the sentence-final syllable. If the stressed rise-fall on *any* indicates a topic accent, then the meaning of the construction can be calculated semantically: the presence of the topic accent indicates a disputable topic (in this case, Which wines will Natalie drink?) [following Daniel Buering, Linguistics and Philosophy **20**, 175–194 (1997) for German]. This operation narrows the domain of *any wine* from all wine from France to some smaller amount perhaps Natalie only drinks expensive cabernets. Furthermore, the same intonational contour occurs in other quantificational statements, like *ALL gamblers aren't addicted*. Here, the disputable topic is: How many gamblers *are* addicted? Again, the intonation serves to narrow the domain to a subset of all the gamblers. Thus, documenting this intonational contour identifies a more general semantic process.

**5pSC36. The role of stress accent in the understanding of sentences in noise.** Pierre Divenyi and Alex Brandmeyer (Speech and Hearing Res., VA Medical Ctr., 150 Muir Rd., Martinez, CA 94553)

When a target sentence is presented in noise, the stressed syllables therein will constitute epochs with a high likelihood of S/N increase. Since in spontaneous speech stressed syllables have significantly higher information content than unstressed syllables [S. Greenberg *et al.*, in Proc. 2d Intern. Conf. Human Lang. Technol. Res. (2002), pp. 36–43], it follows that masking the stressed syllables will negatively impact the perception of neighboring unstressed syllables. This hypothesis was tested by varying the S/N of only the stressed syllables in sentences embedded in otherwise unmodulated speech-spectrum noise. Results suggest that, compared to conditions with absent or decreased-S/N stressed syllables, increasing the S/N during stressed syllables increases intelligibility of the unstressed syllable or syllables that follow. Since the unstressed syllables are rarely masked in the classic, energetic sense, the phenomenon reported constitutes an instance of informational masking. The masking can affect perception at the phonemic as well as on higher linguistic levels, depending on the information provided by sentence context.

# Session 5pUW

## Underwater Acoustics: Autonomous Underwater Vehicles (AUV's), Sensors, and Arrays

Brian T. Hefner, Chair
*Applied Physics Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698*

### *Contributed Papers*

**1:30**

**5pUW1. An autonomous underwater vehicle towed hydrophone array for ocean acoustic measurements.** Jason D. Holmes, William M. Carey (Boston Univ., Boston, MA 02215), James F. Lynch, Arthur E. Newhall, and Amy Kukulya (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

An autonomous underwater vehicle (AUV) with a towed hydrophone array (THA) can provide the capability of mobile-single-ship operation for both short-range single path and long range synthetic aperture experiments. A low noise towed array for an AUV (REMUS) has been developed to demonstrate the feasibility and utility of such measurements. Previous measurements of AUV radiated noise indicated levels that would limit measurements by hull arrays providing a rational for a THA. A small-diameter fluid-filled prototype hydrophone array was developed and tested to ensure adequate sensitivity and system noise levels. The digital recording system (DRS) consisted of mini-disc recorders with a band width of 20 kHz housed in a tube attached to the AUV. This combined system (REMUS, DRS, and THA) was used to conduct a proof of concept test in Dodge Pond. This paper presents the results that show, in the Sea State 0 noise field of Dodge Pond, array system self noise was less than the ambient and vehicle noise was manageable. The test included three pingers and a beacon sound source. The acoustic results were at a high signal-to-noise ratio and show the array tow stability and coherent processing capability. [Work supported by Boston University, Woods Hole, and ONR.]

**1:45**

**5pUW2. Response pattern control in spite of failed array elements.** Henry Cox and Hung Lai (Lockheed Martin Orincon, 4350 North Fairfax Dr., Ste. 470, Arlington, VA 22203)

Large arrays of many elements usually are shaded to achieve low sidelobes. However, in practice these arrays typically operate with some element failures that significantly degrade sidelobe performance. First, the effects of different numbers of failed elements on sidelobe levels are analyzed and illustrated with simulations. Statistics and bounds for sidelobe levels with failed elements are presented. The inability and limitations of conventional shading modifications to compensate for failed elements is discussed and illustrated. Missing elements give rise to large inter-element spacings that in turn result in grating effects that cannot be compensated for using modified shading weights. An adaptive compensation approach for failed elements is then presented. This involves narrowband interpolation from nearby elements to estimate the missing data prior to applying the original shading weights. Failures of both isolated and neighboring elements are considered. When there is a strong source present it is relatively easy to estimate its value on missing elements. But this is exactly the case of interest, when sidelobe levels are important. The adaptive interpolation approach provides good bearing response patterns so that strong signals are well isolated in bearing. Results are illustrated with large array simulations with different failure patterns and multiple strong sources.

**2:00**

**5pUW3. Underwater acoustic measurements with the Liberdade/X-Ray flying wing glider.** Gerald L. D'Spain, Scott A. Jenkins, Richard Zimmerman (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA 93940-0701), James C. Luby (Univ. of Washington, Seattle, WA 98105), and Aaron M. Thode (Scripps Inst. of Oceanogr., La Jolla, CA 93940-0701)

An underwater glider based on a flying wing design (Jenkins *et al.*, 2003) presently is under development by the Marine Physical Laboratory, Scripps Institution of Oceanography and the Applied Physics Laboratory, University of Washington. This design maximizes the horizontal distance between changes in buoyancy to minimize mechanical power consumed in horizontal transport. The prototype wing has a 6.1 m wing span and is 20 times larger by volume than existing gliders. Initial at-sea tests indicate that the lift-to-drag ratio is 17/1 at a horizontal speed of about 1.8 m/s for a 38-liter buoyancy engine. Beamforming results using recordings of the radiated noise from the deployment ship by two hydrophones mounted on the wing verify aspects of the prototype wing flight characteristics. The payload on the new glider will include a low-power, 32-element hydrophone array placed along the leading edge of the wing for large physical aperture at midfrequencies (above 1 kHz) and a 4-component vector sensor. Data previously collected by these types of arrays illustrate the performance of narrow-band detection and localization algorithms. Flight behaviors are being developed to maximize the arrays' detection and localization capabilities. [Work sponsored by the Office of Naval Research.]

**2:15**

**5pUW4. Adaptive rapid environmental assessment simulation framework.** Ding Wang and Henrik Schmidt (Acoust. Lab, Ocean Eng. Dept., MIT, 77 Mass Ave 5-435, Cambridge, MA 02139, prolog@mit.edu)

The coastal environment is characterized by variability on small spatial scales and short temporal scales. These environmental uncertainties translate—in a highly non-linear fashion—into uncertain acoustic propagation properties, often severely affecting the sonar performance prediction capabilities. Conventional oceanographic measurement systems cannot capture these environmental uncertainties due to their limited predictability and the lack of mobility of the traditional measurement platforms. Taking advantage of the mobility of modern autonomous underwater vehicle technology, the Adaptive Rapid Environmental Assessment (AREA) concept has been developed to optimally use the available resources to capture the acoustic uncertainty by adaptive and rapid *in situ* measurement of the environmental properties most significant to the actual sonar system. The ocean area of interest is usually fairly large, and the *in situ* measurement resources are limited, and the adaptive sampling strategy in AREA can therefore make a significant difference in capturing and mitigating the sonar performance uncertainties. To determine an optimal or sub-optimal sampling strategy and test the optimization effect before doing costly on-site experiments, an Adaptive Rapid Environmental Assessment Simulation Framework (AREASF) has been developed, based on state-of-the-art forecasting frameworks and acoustic propagation models. The simulation framework has been applied to investigate the performance of AREA under different environmental conditions, and based on these

results, the feasibility of performing real-time adaptive environmental assesment using AREA under realistic ocean conditions will be discussed. [Work supported by ONR.]

**2:30**

**5pUW5. Optimizing multistatic sonobuoy placement.** Donald R. DelBalzo, Erik R. Rike, and David N. McNeal (Neptune Sci. Div. of Planning Systems, Inc., 40201 Hwy 190 E, Slidell, LA 70461, delbalzo@neptunesci.com)

Sonobuoy patterns for monostatic sensors were developed during the Cold War for deep, uniform underwater environments, where a simple median detection range defined a fixed inter-buoy spacing (usually along staggered lines). Oceanographic and acoustic conditions in littoral environments are so complex and dynamic that spatial and temporal variability of low-frequency signal and noise fields destroys the basic homogeneous assumption associated with standard tactical search concepts. Genetic Algorithms (GAs) have been applied to this problem to produce near-optimal, non-standard search tracks for monostatic mobile sensors that maximize probability of detection in such inhomogeneous environments. The present work describes a new capability, SCOUT (Sensor Coordination for Optimal Utilization and Tactics), to simulate multistatic distributed-sensor geometries and to optimize the locations of multistatic active sonobuoys in a complex, littoral environment. This presentation reviews the GA approach, discusses the new chromosome structure, and introduces a new target-centric geometry. The results show that (a) standard patterns are not optimal even for a homogeneous environment, (b) small distributed sensor clusters are preferred, and (c) standard patterns are grossly ineffective in inhomogeneous environments where 20% improvements in detection are achieved with SCOUT. [Work supported by NAVAIR.]

**2:45**

**5pUW6. Acoustic particle velocity and intensity calculations from tri-axial pressure gradient measurements.** Melanie E. Austin and Alex O. MacGillivray (JASCO Res. Ltd., 2101-4464 Markham St., Victoria, BC, Canada V8Z 7X8, melanie@jasco.com)

In July 2004 Fisheries and Oceans Canada supported a study to investigate effects of seismic airgun signals on hearing organs of freshwater fish in the Mackenzie River at Inuvik, NWT Canada. The study required particle velocity measurements for correlation with observed biological effects. JASCO Research built a pressure gradient measurement apparatus consisting of four hydrophones mounted at the vertices of a triangular-pyramid frame. The system was used to measure differential pressure from the airgun events simultaneously in three perpendicular axial directions. An attached depth-compass sensor monitored the depth and orientation of the system. Hydrophone separations were chosen to be small relative to the acoustic wavelength so that measured differential pressures correctly approximated the pressure gradients along each axis. Particle accelerations were computed directly from pressure gradients following Euler's linearized momentum equation, and particle velocities were computed by integrating particle accelerations. Acoustic intensity was computed from the product of acoustic pressure and particle velocity. The hydrophone precision imposed a limit on accuracy of particle velocity measurements at low frequencies. Likewise the fixed hydrophone spacings defined an upper frequency limit for applicability of this method.

**3:00**

**5pUW7. Macroscopic sonic crystals in a wave guide.** Dalcio K. Dacol, Gregory J. Orris, and David C. Calvo (Acoust. Div. Naval Res. Lab., Washington, DC 20375-5350)

Periodical arrays of long, parallel cylinders have been shown in the published literature to exhibit acoustical properties which are analogous to the electronic properties of certain crystalline solids. Thus there are frequency bands for which no acoustic propagation through the array is allowed, with those band gaps being directionally dependent. The properties of such arrays in wave guides are examined in this work. Particular emphasis is put in investigating the properties of those arrays in wave guides that are idealized models of shallow water oceanic wave guides. The effects of those arrays on the propagation of normal modes is investigated in detail. The possibility of using such arrays in the construction of large scale underwater devices such as filters and lenses is also discussed. [Work supported by ONR.]