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**Session 4aAAa****Architectural Acoustics and Speech Communication: Speech Intelligibility in Buildings and Metrics for the Prediction and Evaluation of Intelligibility**

Kenneth P. Roy, Cochair

*Innovation Center, Armstrong World Industries, P.O. Box 3511, Lancaster, Pennsylvania 17604*

Peter A. Mapp, Cochair

*Peter Mapp Associates, 101 London Road, Copford, Colchester CO3 4JZ, United Kingdom***Chair's Introduction—8:00*****Invited Papers*****8:05****4aAAa1. Is the Speech Transmission Index (STI) a robust measure of sound system speech intelligibility performance?** Peter Mapp (Peter Mapp Assoc., 101 London Rd., Copford, Colchester C03 4JZ, UK, Petermapp@btinternet.com)

Although RaSTI is a good indicator of the speech intelligibility capability of auditoria and similar spaces, during the past 2–3 years it has been shown that RaSTI is not a robust predictor of sound system intelligibility performance. Instead, it is now recommended, within both national and international codes and standards, that full STI measurement and analysis be employed. However, new research is reported, that indicates that STI is not as flawless, nor robust as many believe. The paper highlights a number of potential error mechanisms. It is shown that the measurement technique and signal excitation stimulus can have a significant effect on the overall result and accuracy, particularly where DSP-based equipment is employed. It is also shown that in its current state of development, STI is not capable of appropriately accounting for a number of fundamental speech and system attributes, including typical sound system frequency response variations and anomalies. This is particularly shown to be the case when a system is operating under reverberant conditions. Comparisons between actual system measurements and corresponding word score data are reported where errors of up to 50 implications for VA and PA system performance verification will be discussed.

**8:25****4aAAa2. Correlation study of predictive and descriptive metrics of speech intelligibility.** Abigail Stefaniw, Yasushi Shimizu, and Dana Smith (Prog. in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY 12180)

There exists a wide range of speech-intelligibility metrics, each of which is designed to encapsulate a different aspect of room acoustics that relates to speech intelligibility. This study reviews the different definitions of and correlations between various proposed speech intelligibility measures. Speech Intelligibility metrics can be grouped by two main uses: prediction of designed rooms and description of existing rooms. Two descriptive metrics still under investigation are Ease of Hearing and Acoustical Comfort. These are measured by a simple questionnaire, and their relationships with each other and with significant speech intelligibility metrics are explored. A variety of rooms are modeled and auralized in cooperation with a larger study, including classrooms, lecture halls, and offices. Auralized rooms are used to conveniently provide calculated metrics and cross-talk canceled auralizations for diagnostic and descriptive intelligibility tests. Rooms are modeled in CATT-Acoustic and auralized with a multi-channel speaker array in a hemi-anechoic chamber.

**8:45****4aAAa3. The social effects of poor classroom acoustics on students and The District of Columbia Public Schools demonstration of support through mandating the ANSI Classroom Acoustics standard.** Donna Ellis (The District of Columbia Public Schools, 1709 3rd St. NE, Washington, DC 20002)

The effects that poor acoustics have on students extend beyond the classroom. This paper is to discuss the immediate and long-term results that inadequate acoustical design in the educational setting has on academic and social development and how the District of Columbia Public Schools (DCPS) are contributing to the classroom acoustic movement. DCPS is taking a pro-active stance in educational acoustics by mandating the ANSI Draft S12.60-200X classroom acoustic standard in the transformation of ten schools

a year for the next ten to fifteen years. Synthesizing the ANSI S12 standard with the DCPS Design Guidelines describes explanation of how to design for appropriate acoustics in all core-learning spaces. Examples of the existing conditions of the facilities and acoustical remediation for new and historical preservation projects will be demonstrated. In addition, experience will be shared on the International Building Code Council hearings for classroom acoustics.

### *Contributed Papers*

9:05

**4aAAa4. Assessment of language impact to speech privacy in closed offices.** Yong Ma Ma, Daryl J. Caswell (Dept. of Mech. & Manufacturing Eng., Univ. of Calgary, Calgary, AB T2N 1N4, Canada), Liming Dai (Univ. of Regina, Regina, SK S4S 0A2, Canada), and Jim T. Goodchild (Smed Intl., Calgary, AB T2C 4T5, Canada)

Speech privacy is the opposite concept of speech intelligibility and can be assessed by the predictors of speech intelligibility. Based on the existing standards and the research to date, most objective assessments for speech privacy and speech intelligibility, such as articulation index (AI) or speech intelligibility index (SII), speech transmission index (STI), and sound early-to-late ratio (C50), are evaluated by the subjective measurements. However, these subject measurements are based on the studies of English or the other Western languages. The language impact to speech privacy has been overseen. It is therefore necessary to study the impact of different languages and accents in multiculturalism environments to speech privacy. In this study, subjective measurements were conducted in closed office environments by using English and a tonal language, Mandarin. Detailed investigations on the language impact to speech privacy were carried out with the two languages. The results of this study reveal the significant evaluation variations in speech privacy when different languages are used. The subjective measurement results obtained in this study were also compared with the objective measurement employing articulation indices.

9:20

**4aAAa5. An analysis and retrofit of the acoustics at Image Creators Health and Beauty Salon.** Donna Ellis (521 Manor Rd., Severna Park, MD 21146)

This paper discusses the analysis and retrofit of the acoustics in a high-volume beauty salon in Severna Park, MD. The major issues in what was designed to be a serene environment are reverberation times of 1–1.68 s in the mid- to upper-frequency range. Employee and customer complaints include heightened stress, vocal strain, headaches, and poor intelligibility. Existing analysis and acoustical retrofit solutions will be demonstrated.

9:35

**4aAAa6. A speech perception test for children in classrooms.** Sergio Feijóo, Santiago Fernández, and José Manuel Álvarez (Dept. de Física Aplicada, Fac. de Física, Univ. de Santiago, 15782 Santiago de Compostela, Spain)

The combined effects of excessive ambient noise and reverberation in classrooms interfere with speech recognition and tend to degrade the learning process of young children. This paper reports a detailed analysis of a speech recognition test carried out with two different children popu-

lations of ages 8–9 and 10–11. Unlike English, Spanish has few minimal pairs to be used for phoneme recognition in a closed set manner. The test consisted in a series of two-syllable nonsense words formed by the combination of all possible syllables in Spanish. The test was administered to the children as a dictation task in which they had to write down the words spoken by their female teacher. The test was administered in two blocks on different days, and later repeated to analyze its consistency. The rationale for this procedure was (a) the test should reproduce normal academic situations, (b) all phonological and lexical context effects should be avoided, (c) errors in both words and phonemes should be scored to unveil any possible acoustic base for them. Although word recognition scores were similar among age groups and repetitions, phoneme errors showed high variability questioning the validity of such a test for classroom assessment.

9:50

**4aAAa7. Speech intelligibility in the community mosques of Dhaka City.** Sheikh Muhammad Najmul Imam (Dept. of Architecture, Bangladesh Univ. of Eng. and Technol. (BUET), Dhaka 1000, Bangladesh, najmul@buet.ac.bd)

A mosque facilitates a Muslim community through different religious activities like congregational prayers, recitation and theological education. Speech in a mosque usually generates through bare voice though sound amplification system is also applied. Since no musical instrument is used in any liturgy, a mosque involves only speech acoustics. The community mosques of Dhaka city, the densely populated capital of Bangladesh, are usually designed and constructed by common people inspired from religious virtues. Seeking consultancy for acoustical design is almost never done. As an obvious consequence, there is a common crisis of speech intelligibility in different mosques, except those saved for their smaller volume and other parameters generated by chance. In a very few cases, a trial and error method is applied to solve the problem. But in most of the cases, the problem remains unsolved, putting the devotees in endless sufferings. This paper identifies the type and magnitudes of the prevailing crisis of speech intelligibility of these community mosques through instrumental measurements and questionnaire survey. This paper is also intended to establish certain research rationale and hypothesis for further research, which will propose certain parameters in acoustical design for mosques of Dhaka city in particular and of Bangladesh in general.

**Session 4aAAb****Architectural Acoustics, Musical Acoustics and Physical Acoustics: Ancient Acoustics II  
Vases and Theater Spaces**

Per V. Brüel, Cochair

*Brüel Acoustics, Gl Holtevej 97, Holte DK-2840, Denmark*

Sergio Beristain, Cochair

*IMA ESIME, P.O. Box 75805, 07300, D.F. Mexico****Invited Papers*****10:30****4aAAb1. Models of ancient sound vases.** Per V. Bruel (Bruel Acoust., GI Holtevej 97, Holte DK-2840, Denmark, pvb@bruela.dk)

Models were made of vases described by Vitruvius in Rome in about the year 70 A.D. and of sound vases (lydpotter) placed in Danish churches from 1100–1300 A.D. Measurements of vase's resonant frequencies and damping (reradiation) verified that the model vases obeyed expected physical rules. It was concluded that the excellent acoustical quality of many ancient Greek and Roman theaters cannot be ascribed to the vases placed under their seats. This study also found that sound vases placed in Nordic churches could not have shortened the reverberation time because there are far too few of them. Moreover, they could not have covered a broad frequency range. It remains a mystery why vases were installed under the seats of ancient Greek theaters and why, 1000 years later, Danes placed vases in their churches.

**10:50****4aAAb2. Acoustics of Chinese traditional theatres.** J. Q. Wang (Inst. of Acoust., Tongji Univ., Shanghai 200092, PROC)

Chinese traditional theatre is unique in the world as an architectural form. The Chinese opera evolved into maturity as early as the Song and Yuan Dynasties, 11th–14th centuries, and Chinese theatrical buildings developed accordingly. As the Chinese opera plays on the principle of imaginary actions, no realistic stage settings are required. But Chinese audiences have placed great demands on vocal performance since ancient times. Pavilion stages that are small in area, open on three sides, and thrusting into the audience area are commonly found in traditional theatres, both the courtyard type and auditorium type. The pavilion stage is backed with a wall and a low ceiling (flat or domed). The stage functions as a reflecting shell, which enhances the sound to the audience and provides self-support to the singer. Numerous theatres of this kind exist and function in good condition to the present time. Acoustical measurements show that the sound strength and clarity in audience areas is satisfactory in moderate size courtyard traditional theatres. [Work supported by NSF.]

**11:10****4aAAb3. An acoustical performance space in ancient India: The Rani Gumpha.** C. Thomas Ault and Umashankar Manthravadi (Dept. of Theatre and Dance, Indiana Univ. of Pennsylvania, Waller Hall, Indiana, PA 15705)

The Rani Gumpha, or Queen's Cavern, was built by artist-king of Kalinga, Kharavela (*ca.* 200–100 B.C.). It is a rock cut structure, carved into Udayagiri hill. As in ancient Greek and Roman theaters, the entire performance space of the Rani Gumpha is backed by a decorated facade, and it is remarkably similar to Greek theaters of the Hellenistic period, having both an upper and lower level for playing. There are acoustical chambers behind each level as well as on either side, and a special "cantor's chamber" stage left on the lower level. The effect on the voice is astonishing. This is a rock cut acoustical installation analogous to that described by Vitruvius in Book V, Chaps. 5 and 8, of his *de Architectura*, where he speaks of vessels placed in Greek and Roman theaters for the same purpose. We have created a computerized model of the Ranim Gumpha, using CATT Acoustic. We have taken acoustic measurements of the site, using Aurora Software package. Our results indicate that the Rani Gumpha is an acoustical performance site, sharing characteristics of the classical Greek and Roman theaters of approximately the same period.

## Contributed Papers

11:30

**4aAAb4. Ethnographic model of acoustic use of space in the southern Andes for an archaeo-musicological investigation.** Jose Perez de Arce (Museo Chileno de Arte Precolombino, Bandera 361, Santiago, Chile, jperezdearce@museoprecolombino.cl)

Studies of ritual celebrations in central Chile conducted in the past 15 years show that the spatial component of sound is a crucial component of the whole. The sonic compositions of these rituals generate complex musical structures that the author has termed “multi-orchestral polyphonies.” Their origins have been documented from archaeological remains in a vast region of southern Andes (southern Peru, Bolivia, northern Argentina, north-central Chile). It consists of a combination of dance, space walk-through, spatial extension, multiple movements between listener and orchestra, and multiple relations between ritual and ambient sounds. The characteristics of these observables reveal a complex schematic relation between space and sound. This schema can be used as a valid hypothesis for the study of pre-Hispanic uses of acoustic ritual space. The acoustic features observed in this study are common in Andean ritual and, to some extent are seen in Mesoamerica as well.

11:45

**4aAAb5. General considerations regarding the importance of a study of the acoustic conditions of churches with historic and artistic value in Mexico. The case of baroque architecture.** Fausto Rodriguez Manzo and Eduardo Kotasek Gonzalez (Div. de Ciencias y Artes para el Diseno, Universidad Autonoma Metropolitana-Azcapotzalco, Mexico, D.F., Mexico, rfme@correo.azc.uam.mx)

This paper deals with the problem of old churches in Mexico with historic and artistic value. Some churches in the baroque period in Mexico are examples of a kind of building used as concert and conference halls. However there is a lack of acoustic knowledge of these kinds of architectural spaces in Mexico. This knowledge will also augment that of the historic, artistic and architectural point of view, and be a foundation for high-quality renovation and preservation projects for this type of building. The particular characteristics of some of the most significant churches of this period are presented as well as a first architectural acoustics review of them.

THURSDAY MORNING, 5 DECEMBER 2002

CORAL KINGDOM 2 AND 3, 8:30 TO 11:35 A.M.

### Session 4aAB

#### Animal Bioacoustics: Topics in Animal Bioacoustics

Whitlow W. L. Au, Chair

Hawaii Institute of Marine Biology, P.O. Box 1106, Kailua, Hawaii 96734

## Contributed Papers

8:30

**4aAB1. A general spike coding model for natural echoes.** Rolf Müller (Biosonar Lab, Dept of. Animal Physiol., Tübingen Univ., Morgenstelle 28, D-72076 Tübingen, Germany)

Bats navigating in natural environments face a challenging task in identifying landmarks (e.g., vegetation) that are composed of many reflecting facets and hence generate random echo waveforms. The biological substrate for these tasks is a spike code. Informative code features were identified by a computational approach which encoded a natural stimulus ensemble (84 800 echoes from four deciduous tree species) using a parsimonious model of spike generation. Interspike intervals spanning more than one cycle at the auditory bandpass filter's center frequency proved convenient classification substrates because of the following properties: (a) almost certain presence in each echo representation; (b) simple probabilistic description (uncorrelated); (c) undemanding constraints on firing threshold locations and spike time resolution. An  $m$ -ary sequential probability ratio test using a first-order characterization of these intervals (number plus sample mean of duration and threshold location) demonstrated good classification performance (0.1% to 1% error probability at <10 expected echoes) as well as remarkable robustness against changes in the spike-model parameters. Distant-cycle intervals are therefore hypothesized to be a primal sketch building block analogous to visual edges, with the notable difference that they do not delineate deterministic shapes [Work supported by DFG, SFB 550 B6.]

8:45

**4aAB2. Narrowband *Phocoena*-like clicks produced by a delphinid species.** Marc O. Lammers and Whitlow W. L. Au (Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734, lammers@hawaii.edu)

A distinction is traditionally made between the echolocation clicks of the whistling members of the family *Delphinidae* and those of nonwhistling smaller odontocetes, which include the family *Phocoenidae* and members of genus *Cephalorhynchus* and *Kogia*. The larger, whistling delphinids typically emit clicks that are short (40–70  $\mu$ s) and broadband (30–60 kHz) with peak frequencies ranging from 40 to 130 kHz. Clicks of the smaller nonwhistling odontocetes are generally longer (>100  $\mu$ s), more narrowband (15–25 kHz) with peak frequencies between 90 and 160 kHz. Here we report findings of long (~135  $\mu$ s), narrowband (10–25 kHz) clicks produced by a whistling species of delphinid, the Hawaiian spinner dolphin (*Stenella longirostris*). These clicks typically have peak frequencies between 90 and 100 kHz and predominantly occur in a doublet pattern with clicks consistently spaced 13 ms apart within a doublet and 23 ms between doublets. The vast majority of free-ranging spinner dolphin clicks conform to the more traditional delphinid form. Narrowband clicks appear to be produced rarely, having been observed on only 4 out of 25 recording sessions. It is presently not clear whether these clicks represent echo location or communication signals.

9:00

**4aAB3. Features of echoes bottlenose dolphins use to perceive object properties.** Caroline M. DeLong, Whitlow W. L. Au (Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734, delong@hawaii.edu), and Heidi E. Harley (New College of Florida, Sarasota, FL 34243)

Echolocating bottlenose dolphins can discriminate among objects that vary in size, shape, and structure. Our purpose in this project is to investigate the features of echoes dolphins may use to determine each of these object properties (e.g., target strength, number, and position of highlights). A dolphin performed a cross-modal matching task where he was presented with an object in one modality (e.g., vision) and then asked to choose the same object from among a group of three objects using another modality (e.g., echolocation). The dolphin was presented with object sets in which the objects within the set varied along one feature (size or shape or material or texture). These objects were later ensonified with dolphin-like clicks and the object echoes were collected with a binaural (two hydrophone) system. To mimic the dolphin's ability to scan across objects, echoes were captured as the objects were rotated. Differences in echoic features within object sets were examined along with the dolphin's error. In addition, an artificial neural network (ANN) was used to classify objects using the object echoes. The ANN was utilized to explore the importance of spectral versus time domain features, and single versus multiple object orientations in object classification.

9:15

**4aAB4. Target strengths of two gillnets and predictions of biosonar detection ranges.** T. Aran Mooney, Paul E. Nachtigall, and Whitlow W. L. Au (Hawaii Inst. of Marine Biol., Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, mooneyt@hawaii.edu)

Small cetaceans are often caught as bycatch in gillnet fisheries. One potential method that has been shown to significantly reduce harbor porpoise incidental take is the use of barium sulfate enhanced gillnets. One hypothesis states barium sulfate nets reduce bycatch because they are acoustically more reflective than monofilament nets, thus easier to detect by echolocating odontocetes. This premise remains untested. To determine acoustic reflectivity, this study compared the target strength of a barium sulfate gillnet with a comparable monofilament gillnet at 0, 10, 20, 30, and 40. At normal incidence (0) the monofilament target strength measured slightly greater than the barium sulfate net and biosonar detection ranges are estimated to be greater for the monofilament. As the angle of incidence increased from normal to 20, the barium sulfate net echo strength remained the same while the nylon net echo strength decreased significantly at 10. Biosonar detection ranges are estimated to be greater for the barium sulfate net at increased angles. Echolocating odontocetes may utilize acoustic reflections of both nets but due to the weak target strength of both nets, other mechanisms may also mitigate reduced odontocete bycatch. Current psychoacoustic research is investigating the net detection ranges of captive bottlenose dolphins.

9:30

**4aAB5. Chiroptera-inspired robotic cephaloid (CIRCE): A next generation biomimetic sonar head.** Rolf Müller, Andrew J. Mubeezimagoola (Biosonar Lab, Dept. of Animal Physiol., Tübingen Univ., Morgenstelle 28, D-72076 Tübingen, Germany), Herbert Peremans (Antwerp Univ., B-2000 Antwerpen, Belgium), John Hallam (Univ. of Edinburgh, Edinburgh EH1 2QL, Scotland), Simon Jones, James Flint (Loughborough Univ., Loughborough LE11 3TU, UK), Dominiek Reynaerts, Herman Bruyninckx (Katholieke Universiteit Leuven, B-3001 Leuven, Belgium), Reinhard Lerch, and Alexander Streicher (Friedrich Alexander Univ., D-91052 Erlangen, Germany)

The dynamic interplay between signal processing in the acoustic and neural domain renders the biosonar system of bats, a prime example of intelligent, integrated systems in biology. Previous attempts at technical reproductions of this system have studied a few functional aspects but did

not produce convincing replicas of the integrated system. In particular, none of them could provide for beamforming pinnae, ear mobility, signal bandwidth and an adequate number of elements in the primary signal representation at the scale of the biological paragon. The CIRCE (<http://www.circe-project.org>) project brings together a multidisciplinary consortium with all competences necessary for addressing these aspects in conjunction. Well-integrated solutions to the following technological key challenges are therefore sought: (a) Evolution of pinna shapes to optimize performance in natural biosonar tasks. (b) Adaptation of state-of-the-art ultrasonic transducer technology for matching bats in terms of bandwidth and sound pressure level/sensitivity. (c) Actuation of mobile pinnae (including the possibility of nonrigid transformations) within the scale constraints of a bat's head. (d) Custom-designed DSP hardware providing a qualitative and quantitative reproduction of the neural signal representation at the level of the auditory nerve. [Work supported by the European Commission, LPS Initiative.]

9:45

**4aAB6. Target range discrimination by the bottlenose dolphin.** L. Gennadi Zaslavskiy (Univ. Authority for Appl. Res., Tel-Aviv, P.O. Box 3359, Hadera 38100, Israel)

In order to measure range accuracy of the dolphin's sonar, a range discrimination experiment was conducted. To prevent simultaneous exposure of both targets to each dolphin's click, targets were submerged under water at different ranges in sequence with a 0.3–0.5-sec delay. Otherwise, if presented simultaneously, the targets could be distinguished by difference in the echo arrival times, without measuring the target ranges. Couples of identical spheres as well as targets of a different size and shape were tested. Nevertheless, it proved to be impossible to make the dolphin's measure the distance from themselves to targets. Both dolphins solved the task by measuring a time interval between the target echo and an echo from a reference object, which could be the pool walls or even a net. There was no consistent change in the range accuracy with the change of the target range whereas it regularly increased when the targets were moved closer to the reference object. The dolphins did not keep the distance from the targets as short as possible. Instead, by backing off from the targets they tried to position the target echoes as close as possible to the reference echo. It appeared that the dolphins were incapable of measuring a target range.

10:00–10:20 Break

10:20

**4aAB7. Click train encoding in a song bird: A psychometric function showing two mechanisms of pitch perception.** Jeffrey Cynx (Dept. of Psych., Vassar College, 124 Raymond Ave., Poughkeepsie, NY 12604)

In humans, the pitch of click trains composed of pulses with the same and alternating polarity can vary depending on whether the pulse sequences are below 100 Hz or above 200 Hz. Below 100 Hz, the pitch is equal to the pulse rate regardless of polarity. Above 200 Hz, the pitch is determined by the fundamental frequency and the pattern of click polarity [J. L. Flanagan and N. Guttman, *J. Acoust. Soc. Am.* **32**, 1308–1319 (1960)]. European starlings (*Sturnus vulgaris*), a species of songbird, were trained on an operant procedure with bipolar stimuli on a two tone discrimination, then tested with unipolar stimuli. We then built a psychometric function, especially covering the ambiguous region between 100 and 300 Hz, and the upper frequency limits.



**4aAB8. Hearing is an active process in mosquitoes.** Daniel Robert and Martin C. Goepfert (Biological Sci., Woodland Rd., Univ. of Bristol, Bristol BS8 1UG, UK, d.robert@bristol.ac.uk)

Auditory sensitivity and frequency selectivity are enhanced by active mechanics. Known from vertebrates, active audition is established for an invertebrate. The auditory mechanics of antennal sound receivers in mosquitoes exhibits key diagnostic features of active mechanics: (a) physiologically vulnerable sensitivity and tuning. The mechanical sensitivity and tuning sharpness of the receiver decrease after death. (b) Vulnerable nonlinear mechanical response. Nonlinearity is expressed as amplitude-dependent damping: lower stimulus intensities result in larger response gain and sharper resonance peaks. This band-limited nonlinearity disappears post-mortem. (c) Hypoxia sensitivity. In females, transient exposure to CO<sub>2</sub> reversibly reduces tuning sharpness. In males, the effect is opposite: CO<sub>2</sub> sharpens tuning and elicits large vibrations in the absence of acoustic stimulation. Hypoxia-induced vibrations are unrelated to external forces; they are autonomous and self-sustained, and forcibly generated by an internal motor. (d) Autonomous antennal vibration (AV). AV is also induced by the injection of dimethylsulfoxide (DMSO). DMSO-induced AV persists (90 min) at large amplitudes (>400 nm). Antennal deflection shapes rule out the involvement of muscles, narrowing down the source of mechanical activity to the auditory sense organ at the antennae base. [Work supported by the Leopoldina Academic Society, the Swiss National Science Foundation, BBSRC and the University of Bristol.]

**4aAB9. Micromechanics of *Drosophila* audition.** Martin C. Goepfert and Daniel Robert (Biological Sci., Woodland Rd., Univ. of Bristol, Bristol BS8 1UG, UK, m.goepfert@bristol.ac.uk)

In *Drosophila melanogaster*, hearing is mediated by the antenna, vibrations of which are transduced by Johnston's organ. The receiver's mechanical response exhibits key features of active mechanics. Limit cycle oscillations (LCO) can be measured in untreated animals and after the injection of dimethylsulphoxide. LCO are entrained by acoustic stimuli and reversibly vanish during hypoxia. The mechanical response is nonlinear, exhibiting an increase in resonance frequency with stimulus intensity. This nonlinear stiffness is reversibly affected by hypoxia and disappears post mortem. The origin of mechanical nonlinearity and LCO was traced in auditory mutants no-mechanoreceptor-potential A. NompA protein links the receptor neurone dendrites to the extracellular caps. Mutations in the *nompA* gene thus mechanically isolate the auditory receptors from antennal structures, resulting in conductive hearing loss. An antennal response in *nompA* flies lacks LCO and the nonlinear stiffness alteration of controls. Also, LCO and intensity-dependent stiffness depend on functional mechanotransduction channels: mutations in *nompC* linearize antennal mechanics. Similar effects are measured in touch-insensitive-larva B (*ti1B*) and beethoven (*btv*) mutants, indicating that mechanical activity in *Drosophila* ensues from the interplay between *nompC* channels and receptor cilia, possibly involving ciliary motility. [Work supported by the Leopoldina Academic Society, the Swiss Science Foundation, BBSRC, and the University of Bristol.]

**4aAB10. Acoustic communication by ants.** Robert Hickling (Sonometrics, Inc., 8306 Huntington Rd., Huntington Woods, MI 48070, sonomet@aol.com) and Richard L. Brown (Mississippi State Univ., MS 39763)

Effective communication is essential in highly organized ant societies. Currently myrmecologists believe that pheromones play the central role in ant communication and that relatively little is accomplished with sound. However many ant species make sounds by stridulating, i.e., running a scraper over ridges on their abdomen. Hence, it seems likely that ants communicate using a combination of pheromones and sound. Because ants appear insensitive to airborne sound, myrmecologists have concluded that stridulatory signals are transmitted through the substrate. However, this conclusion is untenable for a number of reasons. How then can ants transmit signals through the air when apparently they are insensitive to airborne sound? The answer lies in the fact that ants are much smaller than the wavelength of the sound they generate, so there is a relatively significant near field surrounding a stridulating ant. The near field is characterized by a major increase in the particle velocity of sound, which can be detected by hair-like sensilla on the apical segments of the ants' antennae. By sensing the relative difference in particle sound velocity between the antennae, an ant can detect stridulation signals from a nearby ant, yet appear deaf to extraneous airborne sound.

**4aAB11. Behavioral responses to acoustical stimulation of individuals and colonies of the termite *Reticulitermes virginicus*.** JulieAnn M. Chambers and Roger Hasse (Univ. of Mississippi, Natl. Ctr. for Physical Acoust., University, MS 38677)

This study examined the effects of acoustical stimulation on *Reticulitermes virginicus* termites and describes the results of experiments conducted to analyze behavioral responses to various frequencies of sound. Termites were studied for individual response to sound stimulation and for colony response. Individuals were mounted to rods and responses were recorded to sound stimulations and magnetic vibrations. Colonies were established and allowed to acclimate in simulated nesting environments containing fine porous sand and rotten wood shims. A transducer was buried in the sand close to the colony and driven by a function generator using variable frequencies at fixed amplitude. Two types of sound stimuli were applied to each colony: tone bursts (TB) and continuous waves (CW). Behaviors indicated that the termites were agitated by the sound stimulus and termite activity was interpreted as an avoidance response. Once stimuli were removed the termites returned to normal activity near the aquarium walls. Observed responses are being examined for consistency and characterization of behaviors. Further study to be presented will quantify the nature of acoustic transmission in the aquaria as well as to understand the termite responses to separate aspects of the stimuli.

## Session 4aBB

## Biomedical Ultrasound/Bioresponse to Vibration: Ultrasound-Mediated Drug Delivery and Gene Transfection

Junru Wu, Chair

Department of Physics, University of Vermont, Burlington, Vermont 05405

### Invited Papers

8:00

**4aBB1. Ultrasound mediated transdermal drug delivery.** Ahmet Tezel, Ashley Sens, and Samir Mitragotri (Chemical Eng. Dept., UCSB, Santa Barbara, CA 93106)

Low-frequency sonophoresis (LFS) is a noninvasive method of transdermal drug delivery and diagnostics. In this method, a short application of ultrasound is used to permeabilize skin for a prolonged period of time. During this period, ultrasonically permeabilized skin may be used as a permeable interface to the body. This interface may be used for the extraction of analytes or the delivery of drugs. Our main objective in this study is to understand the mechanisms of skin permeability due to LFS. Although the principal mechanism of LFS is known to be cavitation, little is known about how cavitation does actually enhance skin permeability. We are specifically interested in understanding the interactions of cavitation bubbles with the skin surface. We are also interested in assessing the structural alterations in the skin induced by ultrasound. We hypothesize that ultrasound generates defects that are responsible for the enhanced delivery of large hydrophilic solutes such as proteins (e.g., insulin). In this study we sought to determine the dependence of LFS transport pathways on ultrasound parameters such as application frequency and energy density.

8:20

**4aBB2. Transdermal drug delivery by sono-macroporation.** Ludwig Weimann (Ultra-Sonic Technologies, LLC, P.O. Box 319, St. Albans, VT 05478, ljw@together.net) and Junru Wu (Univ. of Vermont, Burlington, VT 05405)

A feasibility study of using high amplitude ultrasound to deliver large molecules transdermally was undertaken. Ultrasound (20 kHz) of intensity in the range between 2 to 50 W/cm<sup>2</sup> was used to increase the permeability of skin *in vitro* to poly-L-lysines as models of drug molecules. When 20 kHz, 5% duty cycle ultrasound at the spatial average and pulse average intensity  $I_{SAPA} = 19$  W/cm<sup>2</sup> was applied for 10 min the skin permeability was calculated to be  $0.5 \pm 0.2$  cm/h and  $8.5 \pm 4.2$  cm/h, respectively, for poly-L-lysine-FITC (51 kDa) and octa-L-lysine-FITC (2.5 kDa). Without application of ultrasound the skin permeability of the above-mentioned molecules was essentially zero. A transdermal flux enhancement occurred during the process reported here was much higher than that due to sonophoresis ( $I_{SAPA} < 2$  W/cm<sup>2</sup>) as reported in literature (Mitragotri *et al.*, 1995). Reported here are experimental results from transdermal flux kinetics, and confocal microscopic cross sectional and optical images suggest the formation of pores in the stratum corneum whose size varies with skin samples and may be in the range of 1–100  $\mu$ m. The confocal images also suggest the formation of micron-size pathways in epidermis during ultrasound exposure.

8:40

**4aBB3. Ultrasound-enhanced gene transfer: Comparison of contrast agents and ultrasound modalities *in vitro* and *in vivo*.** Sorin V. Pislaru, Gulati Rajiv, Cristina Pislaru, Randall R. Kinnick, Ripudamanjit Singh, James F. Greenleaf, and Robert D. Simari (Mayo Clinic and Foundation, Rochester, MN 55905)

Ultrasound (US) enhancement of plasmid-based gene transfer is an emerging technique. Our hypothesis was that two contrast agents (Optison and PESDA), and two US exposure modalities (dedicated continuous wave system and diagnostic scanner) may have different effects. Luciferase plasmid with or without contrast agent was added to vascular smooth muscle cells and endothelial cells, followed by US exposure. Luciferase activity was measured 24 h later. US exposure consistently induced higher transfection rates than all controls. PESDA was superior to Optison in both cell lines. *In vitro*, continuous wave and diagnostic US were not significantly different. *In vivo*, Lux and PESDA were injected into skeletal muscles of rats (IM or intra-arterial) followed or not by US exposure. In separate animals, adenovirus encoding for luciferase was injected IM and was not followed by US exposure. Gene transfer efficacy was 8–10 fold higher with US and PESDA than with plasmid alone, but 2 fold lower than with adenovirus. However, as opposed to adenovirus, US-enhanced plasmid gene transfer was highly localized to the injected muscle, with no expression at distal sites. Our results support the hypothesis that contrast agents and exposure modalities are not equivalent with regard to gene transfer efficacy.

9:00

**4aBB4. Gene transfection by echo contrast agent microbubbles.** Katsuro Tachibana (Dept. of Anatomy, Fukuoka Univ. School of Medicine, 7-45-1 Nanakuma, Jonan Fukuoka, Japan, k-tachi@cis.fukuoka-u.ac.jp)

*In vitro* and *in vivo* experiments have demonstrated that various echo contrast agent microbubbles can be intentionally ruptured by diagnostic and therapeutic ultrasound. Violent microstreaming are produced during microbubble collapse. Researchers have hypothesized that these microjets or microstreaming could be applied to promote diffusion of drugs into various tissues and lesions. The most exciting application of this method is probably delivery of genes into cells. As various genes are currently under investigation for the



purpose of treating diseases, ultrasound and microbubbles may be used as a modality to promote better outcome for gene therapy. Recent studies have shown that different gases contained within the bubbles greatly influence the degree of gene transfection. Also, the outer layer of the microbubbles can be custom-made for binding to target tissue. Recent advance on this topic will be discussed.

9:20

**4aBB5. Bubbles, membranes and molecules: sequence of events from sonication to intracellular delivery.** Mark R. Prausnitz, Pavel P. Kamaev, Robyn K. Schlicher, and Vladimir G. Zarnitsyn (School of Chemical Eng., Georgia Tech, Atlanta, GA 30332-0100, mark.prausnitz@che.gatech.edu)

To elucidate mechanisms and control bioeffects for ultrasound-mediated drug and gene delivery, we carried out an experimental study to quantitatively measure the effects of ultrasound and other physical parameters on the sequence of events leading from sonication to drug and gene delivery. Using a Coulter counter, the number and size distribution of bubbles was measured as a function of ultrasound pressure and time, and found to decrease as a function of acoustic energy exposure. The effects of these bubbles on cells were measured by electron and confocal microscopy, which indicated that cavitation created cell membrane defects that could be actively repaired and permitted the intracellular transport of molecules. Using flow cytometry, levels of molecular uptake and cell viability were quantified over a broad range of conditions and correlated with acoustic energy, the ratio of cells-to-cavitation nuclei and the size distribution of bubbles. Finally, levels of gene expression were quantified as a function of acoustic and other parameters and related to cavitation bubble dynamics. By examining each of the steps, rather than only measuring endpoints, we can understand and ultimately control the process by which ultrasound delivers drugs and genes into cells.

9:40–10:00 Break

### Contributed Papers

10:00

**4aBB6. Shock induced jetting of micron sized bubbles.** Claus-Dieter Ohl, Roy Iking, Detlef Lohse (Dept. of Appl. Phys., WSL, TU Twente, Postbus 217, 7500 AE Enschede, The Netherlands), and Andrea Prosperetti (Johns Hopkins Univ., Baltimore, MD 21218)

Gas bubbles having a radius between 10  $\mu\text{m}$  and 100  $\mu\text{m}$  and rising freely in water when being subjected to a shock front exhibit a liquid jetting phenomenon. The jet points in the direction of the propagating shock wave. A linear relationship between the jet length and the bubble radius is found and a lower bound of the averaged velocity of the liquid jet can be estimated to be between 50 m/s and 300 m/s increasing linearly for larger bubbles. In a later stage the jet breaks up and releases micron sized bubbles. In the course of shock wave mediated cell permeabilization this observation suggests a microinjection mechanism responsible for cell transfection when minute gas bubbles are present and exposed together with cells to shock waves.

10:15

**4aBB7. Interaction of ultrasound driven microbubbles and lipid membranes.** Philippe Marmottant and Sascha Hilgenfeldt (Appl. Phys., Univ. Twente, P.O. Box 217, 7500AE Enschede, The Netherlands, p.marmottant@tn.utwente.nl)

Micron-sized bubbles show pronounced oscillations when submitted to ultrasound, leading to increased scattering and improved echographic contrast. It has been reported that this excitation can also alter nearby cell membranes [M. Ward, J. Wu, and J. F. Chiu, *Ultrasound Med. Biol.* **26**, 1169–1175 (2000)], and increase the permeability for drug delivery. To elucidate the mechanisms at work in these sonoporation experiments, we developed a setup that allows for a controlled study of the interaction of single microbubbles with single lipid bilayer vesicles. Substituting vesicles for cell membranes is advantageous because the mechanical properties of vesicles are well-known. Microscopic observations reveal that vesicles near a bubble undergo vivid motion, being periodically accelerated toward and away from the bubble. This “bouncing” of vesicles is a vivid motion that we attribute to a streaming flow field set up by the bubble oscillation. Some vesicles undergo dramatic deformations as they follow the flow, reflecting the high shear rates attained. Break-up of vesicles could also be observed.

10:30

**4aBB8. Evaluation of polymer solubility and cavitation on ultrasound-polymer synergy.** Tyrone M. Porter, Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound (CIMU), Univ. of Washington, Seattle, WA 98105), Patrick S. Stayton, and Allan S. Hoffman (Univ. of Washington, Seattle, WA 98105)

Poly(propylacrylic acid) is a pH-sensitive membrane disruptive polymer designed to release therapeutic molecules from endosomes to the cell cytoplasm before degradation. In mildly acidic environments, the polymer becomes more hydrophobic and less soluble in aqueous medium. Previous research has demonstrated the capacity of polyelectrolytes and therapeutic ultrasound to synergistically disrupt red blood cell membranes, resulting in hemolysis [Mourad (2000)]. In this investigation, the ability of therapeutic ultrasound and PPAA to disrupt cell membranes and deliver fluorescently labeled polymer to the cell cytoplasm was demonstrated. The ultrasound/polymer synergy was dependent upon the solubility of PPAA and acoustic cavitation activity. The solubility of PPAA in phosphate buffered saline was quantified by measuring variations in liquid/gas interfacial tension with polymer concentration and acidity. Reductions of surface tension up to 28% were measured using the Wilhemy plate technique. Acoustic cavitation activity was measured using a passive cavitation detection scheme, quantified, and correlated with the number of fluorescent cells. This study demonstrates the potential of combining therapeutic ultrasound and membrane-disruptive polymers for drug delivery.

10:45

**4aBB9. Ultrasound mediated gene transfection.** Rene G. Williamson, Robert E. Apfel (Dept. of Mech. Eng., Yale Univ., New Haven, CT 06520-8286, rgw27@pantheon.yale.edu), and Janet L. Brandsma (Yale Univ., New Haven, CT 06520)

Gene therapy is a promising modality for the treatment of a variety of human diseases both inherited and acquired, such as cystic fibrosis and cancer. The lack of an effective, safe method for the delivery of foreign genes into the cells, a process known as transfection, limits this effort. Ultrasound mediated gene transfection is an attractive method for gene delivery since it is a noninvasive technique, does not introduce any viral

particles into the host and can offer very good temporal and spatial control. Previous investigators have shown that sonication increases transfection efficiency with and without ultrasound contrast agents. The mechanism is believed to be via a cavitation process where collapsing bubble nuclei permeabilize the cell membrane leading to increased DNA transfer. Our research is focused on the use of pulsed wave high frequency focused ultrasound to transfect DNA into mammalian cells *in vitro* and *in vivo*. Results from further *in vitro* experiments with a 2.5 MHz transducer and Optison as microbubble nuclei will be presented, as well as a discussion of the design of future *in vivo* experiments.

11:00

**4aBB10. Quantification and optimization of transfection efficiency and cell viability by acoustic cavitation.** Vladimir G. Zarnitsyn and Mark R. Prausnitz (School of Chemical Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0100, mark.prausnitz@che.gatech.edu)

Previous studies have shown that ultrasound-mediated cavitation can introduce large numbers of biological macromolecules, such as DNA, into cells. However, optimization of ultrasound conditions is complicated because elevated levels of cavitation increase uptake of molecules, but also decrease cell viability. The goal of this study was to optimize DNA transfection by identifying acoustic and physical conditions that promote DNA uptake without excessive viability loss. To achieve this, suspensions of DU145 prostate cancer cells were sonicated at 24 and 500 kHz in the presence of plasmid encoding green fluorescent protein (GFP) over a range of pressures, exposure times, cell concentrations, media composition, and nucleation site (Optison) concentration. Expression of GFP and cell viability were independently measured by flow cytometry one day after sonication. We found that above a threshold value, transfection increased, went through a maximum and then decreased with increasing pressure, exposure time, and energy exposure. Cell concentration, media composition, and nucleation site concentration also affected transfection levels and were adjusted to optimize DNA expression. Although exact conditions identified in this study may not be optimal in other biological scenarios, we believe the trends observed here can broadly guide optimization of transfection.

11:15

**4aBB11. Characterization of ultrasound-mediated cellular effects.** Robyn K. Schlicher, Monali Desai, Mark R. Prausnitz (Wallace H. Coulter Dept. of Biomed. Eng., Georgia Inst. of Technol., Atlanta, GA 30332), and Robert P. Apkarian (Emory Univ., Atlanta, GA)

Ultrasound can temporarily make cells and tissue more permeable, an effect that could be used for enhanced and targeted drug delivery. Increased permeability is believed to involve creation of transient disruptions in cell membranes. This study seeks to characterize these disruptions and the mechanisms by which they are created, reseal and permit intracellular transport. To achieve this, DU145 prostate cancer cells were exposed to 24 kHz ultrasound with 0.1 s pulse length and 10% duty cycle for 2 s total exposure at pressures from 0.36 to 0.89 MPa. Disruptions were estimated to be at least 50 nm in diameter with lifetimes of 1–2 min using a range of fluorescent molecules with known molecular radii studied using flow cytometry. Cell morphological effects were examined using scanning

electron, transmission electron, and laser scanning confocal microscopies after rapid fixation (within seconds after exposure). Images indicate that cell death from ultrasound exposure occurs due to a combination of apoptosis, necrosis and mechanical fragmentation and uptake may occur through physical disruptions in cell membrane structure. Using red blood cell ghosts and ATP-depleted prostate cancer cells, it was found that molecular uptake into viable cells requires active cellular processes which infers that cell recovery is an energy-dependent process.

11:30

**4aBB12. Measurement of cavitation bubble lifetime and size distribution and its correlation with bioeffects.** Pavel Kamaev and Mark Prausnitz (Dept. of Chemical Eng., Georgia Inst. of Technol., 315 Ferst Dr., Atlanta, GA 30332)

Ultrasound-induced disruption of cell membranes can be used to deliver drugs and genes into cells. Because ultrasound bioeffects are governed by cavitation, measurement and control of cavitation bubble lifetime and size distribution should afford greater control over ultrasound's effects on cells. In this study, 500 kHz ultrasound was focused onto suspensions of albumin-stabilized gas bubbles (Optison) either with or without DU145 prostate cancer cells bathed in calcein. Bubble size and lifetime were determined by Coulter counting, while uptake of calcein and cell viability (i.e., bioeffects) were quantified by flow cytometry. We found that the lifetime and size of Optison bubbles decreased with increasing acoustic energy exposure. Moreover, bioeffects were shown to correlate well with disappearance of bubbles. For example, cell viability remained above 90% until approximately 75% of bubbles were destroyed; viability then dropped dramatically as more bubbles disappeared. Additionally, Optison solutions pre-sonicated to destroy all detectable bubbles also caused significant bioeffects. These observations suggest that bioeffects were caused by the cavitation dynamics of free, short-lived, and/or very small daughter bubbles liberated from albumin-stabilized Optison parent bubbles. Regulation of bubble size distribution, possibly by pre-sonication of Optison solutions, could provide a means to optimize high uptake and cell viability.

11:45

**4aBB13. Nonlinear behavior of encapsulated bubbles relevant to ultrasound mediated gene transfection applications.** Junru Wu, Jason Pepe, and William Dewitt III (Dept. of Phys., Univ. of Vermont, Burlington, VT 05405)

Ultrasound contrast agents (UCA) have been widely used therapeutic applications in medical research and clinical practice. Recently, UCA have been shown to be effective *in vitro* in ultrasound mediated sonoporation and gene transfection and treating vascular thrombosis. The possible primary reasons proposed for the therapeutic applications of UCA are acoustic cavitation [W. J. Greenleaf *et al.*, *Ultrasound Med. Biol.* **24**, 587–595 (1998)] and acoustic microstreaming surrounding acoustically driven UCA [J. Wu, *Ultrasound Med. Biol.* **28**, 125–129 (2002)]. Our study has shown that low frequency (0.6 MHz) is more effective than high frequency (2 MHz) in effects related to the acoustic cavitation, and the opposite is true for the microstreaming due to the nonlinear oscillation behavior of encapsulated bubbles of UCA.

**Session 4aEA****Engineering Acoustics: Metrology Standards and Calibration on Acoustics, Ultrasound and Vibrations**

George S. K. Wong, Cochair

*Institute for National Measurement Standards, National Research Council, 1500 Montreal Road, Ottawa, Ontario K1A 0R6, Canada*

Jose Salvador Echeverria, Cochair

*Centro Nacional de Metrología, km 4.5, Carretera a los cues el Marques, Queretaro 76900, Mexico*

Marc Antonio Nabucode Arauj, Cochair

*Centro Nacional de Metrología, km 4.5, Carretera a los cues el Marques, Queretaro 76900, Mexico***Chair's Introduction—7:55*****Invited Papers*****8:00****4aEA1. Microphone interlaboratory comparison in the Americas.** George S. K. Wong and Lixue Wu (Inst. for Natl. Measurement Standards, Natl. Res. Council, Ottawa, ON K1A 0R6, Canada)

The final results of a Sistema Interamericano de Metrología (SIM) interlaboratory comparison on microphone calibration are presented. Initially the comparison involved NORAMET countries: USA, Canada, and Mexico. Later, the comparison was extended to include Argentina and Brazil, resulting in a SIM AUV.A-K1 microphone interlaboratory comparison. The National Metrology Institutes (NMIs) of the five American countries that participated were the Institute for National Measurement Standards (INMS—Canada), National Institute of Standards and Technology (NIST—USA), Centro Nacional de Metrología (CENAM—Mexico), Instituto Nacional de Metrología, Normalização e Qualidade Industrial (INMETRO—Brazil) and Unidad Técnica Acústica, (INTI—Argentina). INMS, Canada was the pilot laboratory that provided the data for the final report. The maximum rms deviation for the two LS1P laboratory standard microphones measured by the above participants is 0.037 dB that may be considered as the key comparison reference value.

**8:15****4aEA2. Issues concerning international comparison of free-field calibrations of acoustical standards.** Victor Nedzelnitsky (Natl. Inst. of Standards and Technol. [NIST], 100 Bureau Dr., Stop 8221, Sound Bldg. [233], Rm. A147, Gaithersburg, MD 20899-8221, Victor.Nedzelnitsky@nist.gov)

Primary free-field calibrations of laboratory standard microphones by the reciprocity method establish these microphones as reference standard devices for calibrating working standard microphones, other measuring microphones, and practical instruments such as sound level meters and personal sound exposure meters (noise dosimeters). These primary, secondary, and other calibrations are indispensable to the support of regulatory requirements, standards, and product characterization and quality control procedures important for industry, commerce, health, and safety. International Electrotechnical Commission (IEC) Technical Committee 29 Electroacoustics produces international documentary standards, including standards for primary and secondary free-field calibration and measurement procedures and their critically important application to practical instruments. This paper addresses some issues concerning calibrations, standards activities, and the international key comparison of primary free-field calibrations of IEC-type LS2 laboratory standard microphones that is being planned by the Consultative Committee for Acoustics, Ultrasound, and Vibration (CCAUV) of the International Committee for Weights and Measures (CIPM). This comparison will include free-field calibrations by the reciprocity method at participating major national metrology laboratories throughout the world.

**8:30****4aEA3. A BIPM/CIPM key comparison on microphone calibration—defining the state of the art.** Richard G. Barham (Acoust. Metrology Group, Natl. Physical Lab., Teddington, Middlesex TW11 0LW, UK, richard.barham@npl.co.uk)

The formation of the Consultative Committee on Acoustics, Ultrasound and Vibration (CCAUV) now gives acoustical quantities the same status as more established metrics like mass or voltage. The principle work of CCAUV is to establish degrees of equivalence between member states of the Metre Convention. This is achieved through key comparisons and subsequent regional comparison. For sound-in-air, the subject of this key comparison is most appropriately the calibration of laboratory standard microphones. The first such project considered the pressure calibration of IEC type LS1P microphones in the frequency range from 63 Hz to 8 kHz. Twelve national laboratories took part in the key comparison, piloted by the UK's National Physical Laboratory. The project has now been completed. The results for the measured pressure sensitivity level have a standard deviation of around 0.02 dB at frequencies below 1 kHz, rising to a maximum of 0.04 dB at 8 kHz. The mean of these results normalized to 0 dB at each frequency is considered as the key comparison reference value (KCRV) and the standard deviation provides an estimate of its standard uncertainty. The KCRV then defines the datum which enables the performance of all laboratories to be related.

**4aEA4. Sound power measurement standards at laboratory and field environments.** Lucía Taibo, Jorge Riganti, and Waldemar Dittmar (U.T. Acústica, Instituto Nacional de Tecnología Industrial, INTI, c.c. 157, 1650 Prov. Buenos Aires, Argentina)

Sound power level measurements, SWLs, of equipment and machinery through direct or comparison methods at laboratory or field conditions, are of increasing importance in terms of regulations and industrial competitiveness. SWLs constitute the input data when modeling the real world, in complex industrial premises for example, as a means for diagnosis and SPL predictions. SWLs have a major technical and economical impact, trying to keep the measurement uncertainties as low as possible, while balancing the testing complexity, time consumption, and costs. In the present work, a description about the calibration of a Reference Sound Source, RSS, B&K 4204 [ISO 6926] is made. The testing survey of normal equipment emitting steady noise and the RSS at different acoustical environments, that range from an hemi-anechoic room to a typical industrial facility is also described. Some measuring methods such as ISO 3745, ISO 3744, ISO 9614-2 are compared for the different sources and environments, yielding some discrepancies between standards as is extensively referred to in literature. A proposal based on a new multiple height microphone array is encouraged at laboratory conditions. Regarding field tests, sound intensity measurements by scanning is extensively adopted for practical reasons, with acceptable results according to our experience.

**4aEA5. The current status of measurement standards for acoustics and vibration at INMETRO.** Gustavo P. Ripper and Walter E. Hoffmann (INMETRO, DIAVI, Av. N. S. das Graças, 50, Xerém-D. Caxias, Rio de Janeiro 25250-020, Brazil, lavib@inmetro.gov.br)

The Division of Acoustics and Vibration of INMETRO (DIAVI) establishes, validates and maintains the Brazilian national measurement standards used for the realization of the units of physical quantities related to the field of acoustics and vibration. The basic vibration quantity realized by DIAVI is translational acceleration, from which the other motion quantities, i.e., velocity and displacement can be derived. Acoustics physical quantities include sound pressure and sound power. The national measurement standards comprise both absolute (primary) and comparison (secondary) calibration setups, which are used to cover the wide range of calibration and testing services demanded by the Brazilian society, trade, industry, and science. This paper outlines the current status of these measurement standards and presents the measurement capabilities of INMETRO in the field of acoustics and vibration.

**4aEA6. The influence of standards on acoustic transducer design and calibration.** Gunnar Rasmussen (G.R.A.S. Sound & Vibration ApS, Skelstedet 10B, 2950 Vedbaek, Denmark, gr@gras.dk)

The Western Electric 640AA microphone set the standard for measurement microphones and influenced their development for many decades. It has made its mark on  $1 - \frac{1}{2} - \frac{1}{4} - \frac{1}{8}$  in. microphones. It has also had an influence on calibration techniques and calibrators. Developments in microphones—calibration and standards will be discussed. The microphone is an important part of a sound level meter and of the complete measurement setup. In particular, measurements in the outdoor environment are greatly influenced by the requirements set by sound level meter standards.

**4aEA7. Validity of acoustic reference standards.** E. Frederiksen (Bruel & Kjaer, Skodsborgvej 307, 2850 Naerum, Denmark and Danish Primary Lab. of Acoust.)

Today, acoustic calibrations and associated uncertainties are hot topics. Countries all over the world establish new calibration facilities and coordinate their efforts to make sure that they can calibrate correctly and have their results internationally accepted. This work goes on within accreditation, between National Calibration Institutes, and within standardization. The working groups of the technical committee IEC/TC29, which today make essentially all acoustic instrument and calibration standards, thus expand existing standards with detailed test procedures and related uncertainty requirements. The new requirements are partly associated with the primary pressure and free-field calibration of Reference Standards performed by National Calibration Institutes, and partly with the testing of instruments in Calibration Service Centers, which are the main users of the references. This paper deals with calibration of Reference Standards, which generally needs to be analyzed and elaborated in connection with the ongoing international work. Thousands of microphone and pistonphone calibration results of the Danish Primary Laboratory of Acoustics have been analyzed. Resulting data including uncertainty, reproducibility, and repeatability will be presented. The data are based on calibrations performed over several decades and covering the range from 20 Hz to 25 kHz.

### Contributed Papers

**4aEA8. Uncertainties related to fast determination of free-field response for sound level meters.** Johan Gramtorp (Bruel & Kjaer, Skodsborgvej 307, 2850 Naerum, Denmark)

As a part of the ongoing work on the new standard IEC 61672 Electroacoustics—Sound level meters, the TC29 WG4 working group is writing the specifications for the periodic verification of sound level meters. Recent investigations of historical data have shown that the fre-

quency response for the microphone is the most likely cause for the rejection of a sound level meter under test. It is therefore very important to measure the frequency response of the sound level meter with acoustical input signals. In order to keep the cost of the test down, it is essential to be able to choose a fast and sufficiently accurate method. In this paper some of the possible measurement methods such as free-field comparison, electrostatic actuator, multifrequency calibrator, and comparison coupler are described. The focus is on their advantages, drawbacks, and their associated uncertainties.

10:00

**4aEA9. Perspectives on anechoic chamber qualification.** Kenneth A. Cunefare and Van B. Biesel (School of Mech. Eng., Georgia Tech, Atlanta, GA 30332-0405)

The qualification of a new anechoic chamber requires demonstration that the chamber produces a free-field environment within some tolerance bounds and over some acceptable volume. At the most basic level, qualification requires measurement of sound levels at increasing distances from a test source, and then comparing the levels to a theoretical free-field decay. While simple in concept, the actual performance of a qualification

test is problematic in implementation, with troublesome issues relevant to the nature of the sound source, test signal (broadband or pure tone), spatial resolution of measurements (e.g., measurements at discrete locations or spatially continuous), and comparison of the data to a theoretical decay. This presentation will provide a brief historical perspective on chamber qualification and review current practice. It will demonstrate the inadequacy of broadband noise and widely spaced discrete measurements for qualification purposes. It will demonstrate that pure tone signals and spatially continuous measurements provide a rigorous test of a chambers performance.

### *Invited Papers*

10:15

**4aEA10. IEC measurement standards for ultrasonic hydrophones and radiation force balances.** K. Beissner (Physikalisch-Technische Bundesanstalt, 38116 Braunschweig, Germany)

Acoustic output measurements of medical ultrasonic equipment are usually performed in water and using two widely established methods: ultrasonic power measurement with a radiation force balance and acoustic pressure measurement with an ultrasonic hydrophone. The International Electrotechnical Commission (IEC) has published a number of International Standards on both methods, prepared by its Technical Committee 87 "Ultrasonics." These are discussed and future standardization trends are presented. The basic standard for ultrasonic power measurements is IEC 61161. It recommends using a radiation force balance and specifies the relevant technical properties and sources of uncertainty. It is currently envisaged to revise this standard and to include particular recommendations for the power range up to 20 W which have recently been worked out in an international cooperation of national metrology institutes. Hydrophone standards have continuously been developed over many years. The most recent one is IEC 62092 which extends the upper frequency limit of hydrophone standardization from 15 MHz to 40 MHz. The current trend is to reorganize all existing hydrophone standards and to find a new scheme of three parallel standards dealing with hydrophone properties, hydrophone use, and hydrophone calibration.

10:30

**4aEA11. A BIPM/CIPM key comparison covering the calibration of ultrasonic hydrophones over the frequency range 1 MHz to 15 MHz.** Bajram Zeqiri and Nigel D. Lee (Acoustical Metrology Group, Natl. Physical Lab., Teddington, Middlesex TW11 0LW, UK, bajram.zeqiri@npl.co.uk)

A central objective of the Mutual Recognition Arrangement (MRA), signed by national measurement institute (NMI) directors in 1999, is the establishment of the degrees of equivalence of national measurement standards held by each institute. International comparisons, known as key comparisons, represent the sole mechanism for establishing these degrees of equivalence. In this paper we describe a key comparison, undertaken under the auspices of the BIPM/CIPM Consultative Committee for Acoustics, Ultrasound, and Vibration, related to the realization of the acoustic pascal in water at ultrasonic frequencies. This is most appropriately achieved through a comparison of calibrations of stable transfer standard hydrophones; 1 mm active element bilaminar membrane hydrophones, being chosen for this purpose. With NPL acting as the pilot laboratory, two hydrophones were calibrated using the NPL primary standard laser interferometer and circulated sequentially to participant NMI laboratories in Germany, China, The Netherlands, and Denmark. Laboratories were asked to report values for the hydrophone open-circuit free-field sensitivity over the frequency range 1 MHz to 15 MHz. The principal calibration methods used by the NMIs were optical interferometry and/or two-transducer reciprocity. The key comparison process, its results, and the analysis used to derive the key comparison reference values, are all described in detail.

10:45

**4aEA12. ISO standardization to ensure traceability of vibration and shock acceleration measurements.** Hans-Jürgen von Martens (Physikalisch Technische Bundesanstalt PTB, Bundesallee 100, 38116 Braunschweig, Germany, hans-juergen.v.martens@ptb.de)

Traceability to the International System of Units (SI) is increasingly demanded for vibration and shock measurements as specified in international standards, recommendations, and regulations to ensure product quality, health, and safety. Hierarchies of measurement standards have been established and are operated at the National Metrology Institutes (NMIs) as well as accredited and nonaccredited calibration laboratories. To meet the increasing demands, a revision of the ISO 5347 series, renumbered to ISO 16063, started in 1995 (ISO TC 108). Improved and new standard methods for vibration and shock calibration have been specified at the different traceability levels. For primary vibration calibration by laser interferometry at the NMI level, ISO 16063-11:1999 has extended the frequency range (0,4 Hz–10 kHz) and included absolute-phase shift measurement. ISO 16063-13:2001 provides interferometric primary shock



calibration ( $100 \text{ m/s}^2$ – $100\,000 \text{ m/s}^2$ ). For vibration and shock calibration at lower levels, parts 21 and 22 of ISO 16063 provide upgraded comparison methods. ISO standards for the calibration of angular transducers by laser interferometry (part 15) and by comparison (part 23) are under development. Using the methods specified, national standards and calibration equipment have been developed for calibration laboratories, offering many new and improved calibration and measurement capabilities and thus international traceability.

11:00

**4aEA13. On the uncertainties of accelerometer calibration by laser interferometry.** Guillermo Silva-Pineda (Centro Nacional de Metrología, km 4,5 Carr. a los Cus, El Marques, Qro. CP 76241, Mexico)

Depending on both measurement standards and measurement methods, derived quantities such as acceleration are traceable to the basic quantities in the International System of Units, SI. The uncertainty of measurement is often related to the length of the traceability chain up to the standards of reference quantities and the accuracy of the methods used in every calibration step. The traceability chart of the experimental arrangements can be used to visualize the different uncertainty levels and their relationships from the application of measurement to the standards of basic quantities in the SI. For the case of accelerometer calibration, experimental results of the fringe-counting method using a Michelson interferometer, which has standards traceable to basic units of length and time, are shown. Also, the physical principles and the alignment of the laser interferometer are discussed, and how the measurement accuracy can be improved. The classical approach of the uncertainty budget, based on a first-order Taylor series approximation, for the sensitivity calibration of an accelerometer is compared with a model based on higher-order terms in the Taylor series approximation. Some concluding remarks obtained from the comparison of these two approaches are given.

11:15

**4aEA14. International documentary standards and comparison of national physical measurement standards for the calibration of accelerometers.** David J. Evans (Natl. Inst. of Standards and Technol., 100 Bureau Dr., M.S. 8221, Gaithersburg, MD 20899-8221)

The documentary standards defining internationally adopted methodologies and protocols for calibrating transducers used to measure vibration are currently developed under the International Organization for Standardization (ISO) Technical Committee 108 Sub Committee 3 (Use and calibration of vibration and shock measuring instruments). Recent revisions of the documentary standards on primary methods for the calibration of accelerometers used to measure rectilinear motion have been completed. These standards can be, and have been, used as references in the technical protocols of key international and regional comparisons between National Measurement Institutes (NMIs) on the calibration of accelerometers. These key comparisons are occurring in part as a result of the creation of the Mutual Recognition Arrangement between NMIs which has appendices that document the uncertainties, and the comparisons completed in support of the uncertainties, claimed by the National Laboratories that are signatories of the MRA. The measurements for the first international and the first Interamerican System of Metrology (SIM) regional key comparisons in vibration have been completed. These intercomparisons were promulgated via the relatively new Consultative Committee for Acoustics, Ultrasound and Vibration (CCAUV) of the International Committee for Weights and Measures (CIPM) and SIM Metrology Working Group (MWG) 9, respectively.

### *Contributed Papers*

11:30

**4aEA15. High-frequency acceleration comparison among NIST, CENAM, and INMETRO.** Gustavo P. Ripper (INMETRO, DIAVI/LAVIB, Av. N. S. das Graças, 50, Xerém-D. Caxias, Rio de Janeiro 25250-020, Brazil, lavib@inmetro.gov.br), Beverly F. Payne (NIST, Gaithersburg, MD 20899-8221), and Guillermo Silva-Pineda (CENAM, C.P. 76900, Querétaro, Mexico)

This paper presents the results obtained in an international laboratory comparison among NIST/USA, CENAM/Mexico, and INMETRO/Brazil. This acceleration comparison was focused on the calibration of the charge sensitivity of a single-ended standard accelerometer in the frequency range from 3 kHz to 10 kHz. The measurements were carried out by laser interferometry in accordance with the standard ISO 16063-11:1999 and with a preagreed protocol. This comparison supplements the former comparison SIM.AUV.V-K1, which covered the frequency range from 50 Hz to 5 kHz.

11:45

**4aEA16. Charge amplifier calibration comparison among NIST, CENAM, and INMETRO.** Gustavo P. Ripper (INMETRO, DIAVI, Av. N. S. das Graças, 50, Xerém-D. Caxias, Rio de Janeiro 25250-020, Brazil, lavib@inmetro.gov.br), Beverly F. Payne (NIST, Gaithersburg, MD 20899-8221, payne@nist.gov), and Guillermo Silva-Pineda (CENAM, Querétaro, Mexico)

This paper presents the results obtained in an international laboratory comparison among NIST/USA, CENAM/Mexico and INMETRO/Brazil which was focused on the electric calibration of a charge amplifier B & K model 2626 from 50 Hz to 5 kHz. One of the main concerns of this comparison was to conclude all measurements within the period of 1 month, in order to minimize the influence of long-term temporal instabilities. Each participating laboratory carried out the measurements in accordance with a calibration protocol, which was established to avoid the effect of additional sources of uncertainty. The results are compared with the ones obtained during the former comparison SIM.AUV.V-K1 and the conclusions are presented.

**Session 4aED****Education in Acoustics: Virtual Labs, Workshops and Multimedia in Acoustics Education**

Ralph T. Muehleisen, Chair

*Civil, Environmental and Architectural Engineering, University of Colorado, Boulder, Colorado 80309-0428***Chair's Introduction—8:30*****Invited Papers*****8:35****4aED1. Concerthalls.org: A webpage for architectural acoustics education.** Lily M. Wang and Jessica M. Hall (Architectural Eng. Prog., Univ. of Nebraska–Lincoln, 200B Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0681, lwang@unl.edu)

A website focusing on concert hall acoustics ([www.concerthalls.org](http://www.concerthalls.org)) has been developed under the Schultz Grant from the Newman Student Award Fund. The website includes historical information, discussions on a variety of architectural acoustical measures, links to other websites of interest including a collection of concert hall webpages, and a comprehensive reference list. Of particular interest are the many listening examples provided on the website, which help students and other visitors to understand each subjective quality audibly and give insight on how to measure and control the associated objective measure. Examples are provided for reverberation, clarity, intimacy, warmth, loudness, and spaciousness. Various samples will be played during this presentation. Instructors of architectural acoustics are encouraged to introduce the site to their students and incorporate it into their instructional materials. [Work supported by Schultz Grant from the Newman Student Award Fund.]

**8:55****4aED2. Animations and auralizations for noise control education.** Ralph T. Muehleisen (Univ. of Colorado, Boulder, CO 80309-0428)

For students just starting out in acoustics, some basic concepts can be terribly difficult to understand. Ideas like plane, spherical, travelling, standing, and evanescent waves are easily described mathematically but can be difficult to understand physically. The use of animation to show the particle motion, velocity, and/or pressure fields associated with these wave motions helps to solidify the understanding of the basic physics. In noise control, students often have problems conceiving the effects of reverberation on acoustic signals and the perceived noise reduction from various acoustic treatments. These acoustic effects can be simulated through the use of auralization. Animations and auralizations developed using MATLAB, CATT Acoustics, and Cooledit for a senior level architectural engineering undergraduate classes in building noise control and acoustical room design will be presented.

**9:15****4aED3. Acoustics workshops for teachers in elementary schools and in secondary school science classes.** Uwe J. Hansen (Dept. of Phys., Indiana State Univ., Terre Haute, IN 47809)

A number of acoustics workshops have been conducted throughout the US with support from ASA technical initiative and education funds. Emphasis in elementary schools is on using music to introduce science concepts. Concepts discussed include wave properties such as wavelength, frequency, propagation speed, and standing waves. The idea of resonance is experienced musically and physically, and the relationship between whole number frequency multiples and the musical harmonic series is developed. Computer programs such as COOL-EDIT are used extensively in secondary teacher workshops to experience frequency analysis, frequency synthesis, and resonances in open and closed tubes. Workshop teaching approaches for these topics will be illustrated in this presentation.

**9:35****4aED4. Computer simulations enhance experimental demonstrations in the underwater acoustics and sonar course.** Murray S. Korman (Dept. of Phys., U.S. Naval Acad., Annapolis, MD 21402)

Underwater acoustics and sonar (SP411) is a 3 hour course that is offered to midshipmen in their senior year. Typically, general science and oceanography majors, totaling 110 students/yr, enroll. Since this course is offered without a lab, the “in class” experience has been enhanced with the development (over many years) of our demo carts and computer workstations which surround the classroom. In a studio classroom atmosphere, students perform a variety of experiments in small groups. How can scientific visualizations best develop learning of complex interactions? Two examples are presented. PC–IMAT (personal curriculum interactive multisensor analysis training) simulations of multielement array steering support the theory and enhance the experiments that are performed in class such as the two-element array. Mathematica simulations involving the programming and animation of a point source in a rigid–rigid infinite parallel wave guide are used to stress the method of images, superposition, group and phase velocity and far-field modal pattern that is observed as a function of depth and source frequency. Later, students have fun using a ripple tank with an eye dropper to generate a point source between two adjustable parallel boundaries, and their understanding of “underwater sound” is greatly enhanced.

## Contributed Papers

10:05

**4aED5. Experiences on Science, Technology and Society (ScTS) in teaching environmental noise.** Pablo Roberto Lizana Paulin, Jose de Jesus Negrete Redondo, and Elena Romero Rizo (Acoust. Lab., ESIME, IPN, Mexico, plizana98@hotmail.com)

The ScTS (Science, Technology and Society) is a new work field for a critical and interdisciplinary way to evaluate the social dimension of science and technology with regard to the social and environmental effects, in this case of environmental noise contamination. This new way to see science promotes reflection and modification of personal and professional values. Teams were organized to research the environmental noise impact on society and the human being. Then discussions were carried out in order to prepare classroom presentations. After the presentations and proposals, further discussion and conclusions about the problem were made, getting the students involved in the possible solutions. (To be presented in Spanish.)

10:20

**4aED6. Help! There are 60 screaming kids in my lab!** Corinne Darvennes (Dept. of Mech. Eng., Tennessee Technolog. Univ., Box 5014, Cookeville, TN 38505, CDarvennes@tntech.edu)

What do you do when you have a large noise control facility available and you want to introduce local children to science and engineering? You invite them in! This presentation will describe a day of hands-on activities for 60 fifth graders in the Acoustics and Vibrations Laboratory at TTU. This includes the logistics of having 60 kids in the lab and keeping their attention, a description of each activity, and the equipment used. Activities included learning about hearing and hearing loss, propagating and standing waves, ultrasound, vibrations, and resonance. Students used slinkies, sound level meters, a strobe light, Vernier LabPro data collection systems and sensors (ultrasonic motion detectors and microphones). They popped balloons in the reverberation chamber to listen to the sound decay. Their most memorable experience, however, was the screaming contest in the anechoic room!

10:35

**4aED7. Didactic toy for teaching the Fourier principle.** Miguel Medina-Villanueva, Victor Medina-Tamez, Javier Medina-Tamez, and Adrian Garcia-Mederez (Acoust. Lab., FIME, Univ. A. de Nuevo Leon, P.O. Box 28 "F," Cd. Universitaria, San Nicolas, 66450 N.L., Mexico)

The mathematical tool of Fourier analysis is used in many areas like vibrations, communications, optics, electronics, etc. The understanding of this subject sometimes causes frustration in students. The main purpose of this presentation is to propose a didactic toy that calculates the harmonic magnitudes through the discrete values of analog periodic signals. This device shows the rotative vectors in a physical way that makes the principle of Fourier understandable.

10:50

**4aED8. Using computers to overcome math-phobia in an introductory course in musical acoustics.** Andrew A. Piasek (Central Washington Univ., Ellensburg, WA 98926)

In recent years, the desktop computer has acquired the signal processing and visualization capabilities once obtained only with expensive specialized equipment. With the appropriate A/D card and software, a PC can behave like an oscilloscope, a real-time signal analyzer, a function generator, and a synthesizer, with both audio and visual outputs. In addition, the computer can be used to visualize specific wave behavior, such as superposition and standing waves, refraction, dispersion, etc. These capa-

bilities make the computer an invaluable tool to teach basic acoustic principles to students with very poor math skills. In this paper I describe my approach to teaching the introductory-level Physics of Musical Sound at Central Washington University, in which very few science students enroll. Emphasis is placed on how visualization with computers can help students appreciate and apply quantitative methods for analyzing sound.

11:05

**4aED9. Lab experiment using physical models of the human vocal tract for high-school students.** Eri Maeda, Takayuki Arai, Noriko Saika, and Yuji Murahara (Dept. of Elec. and Electron. Eng., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, m-eri@sophia.ac.jp)

Recently, the development of educational tools for acoustics has become popular in Japan. We believe that physical models of the human vocal tract are particularly useful for teaching acoustics. Formerly we proposed three models of the vocal tract corresponding to the Japanese vowels, /i/, /e/, /a/, /o/, and /u/. We presented cylindrical, nasalized, and plate type models. The models were made of transparent acrylic resin, enabling the configurations of the oral cavity to be seen from the outside of the model. In this presentation, we will discuss the results of a lab experiment in which we used these tools to teach the mechanism of vowel production to high-school students who had just finished studying basic acoustics. By manipulating the plates in the plate type model, students were able to simulate constrictions at nodes and antinodes, and they were able to hear the shift in formant frequencies. The exercise helped students to understand vowel production. We received positive feedback from those who participated in the experiment.

11:20

**4aED10. Physical models of the human vocal tract as tools for education in acoustics.** Takayuki Arai, Eri Maeda, Noriko Saika, and Yuji Murahara (Dept. of Elec. and Electron. Eng., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan)

We recently replicated the Chiba and Kajiyama physical models of the human vocal tract and found that they are extremely effective as a manipulative tool for education in acoustics. This study is an overview of our previous development of the models and their application for education. The models we developed simulate the configuration of the vocal tract, and are of two major types: cylinder and plate. The interior, bottle shape of the cylindrical models was based on the original measurements of Chiba and Kajiyama. The diameters of the holes in the 10–17 movable plates of the plate-type model were also based on the Chiba and Kajiyama measurements, but the radius curves were approximated at 10 mm resolution in a stepwise fashion. We also developed a nasalized model, which consists of a side branch attaching to a cylindrical model. Our experience confirmed that when used in a classroom environment, the models increase student understanding of the acoustic theory of speech production.

11:35

**4aED11. Musical appreciation.** Maria del Consuelo Medina (Departamento de Musica, Centro Universitario de Arte Arquitectura y Diseo, Universidad de Guadalajara, Morelos 181, Zona Centro, Guadalajara, Jal. Mexico)

Pre-school listening to music is the principal way that leads to the appreciation of music that later facilitates knowledge and pleasure in the history of music. At the preschool age it is a very important aspect of education, and reasons and suggestions will be given. The activities must be brief, the teachers of music can at the most develop the activity every

five minutes, leaving time for rest or expansion. Another suitable way to bring the child to music is through stories, which please all children; let them go to an unreal and fantastic world and listen to a story or an exciting adventure. The story then, should be brief, simple, with action, with fa-

miliar characters, but with some mystery; some repetitive element; and an ending both surprising and happy. It is preferable to include small folkloric tales from the universal repertoire, with works of simple and clear structure.

THURSDAY MORNING, 5 DECEMBER 2002

CORAL GARDEN 2 AND 3, 7:55 A.M. TO 12:10 P.M.

### Session 4aNS

## Noise and Structural Acoustics and Vibration: Aircraft Interior Noise

David Reed, Cochair

*The Boeing Company, P.O. Box 3707, Seattle, Washington 98124-2207*

Kevin P. Shepherd, Cochair

*Structural Acoustics Branch, NASA Langley Research Center, Hampton, Virginia 23681*

Chair's Introduction—7:55

### Invited Papers

8:00

**4aNS1. Recent advances in active control of aircraft cabin noise.** Gopal Mathur (The Boeing Co., 5301 Bolsa Ave., MC: H013-B308, Huntington Beach, CA 92647) and Christopher Fuller (Virginia Polytechnic Inst. & State Univ., Blacksburg, VA 24061)

Active noise control techniques can provide significant reductions in aircraft interior noise levels without the structural modifications or weight penalties usually associated with passive techniques, particularly for low frequency noise. Our main objective in this presentation is to give a review of active control methods and their applications to aircraft cabin noise reduction with an emphasis on recent advances and challenges facing the noise control engineer in the practical application of these techniques. The active noise control method using secondary acoustic sources, e.g., loudspeakers, as control sources for tonal noise reduction is first discussed with results from an active noise control flight test demonstration. An innovative approach of applying control forces directly to the fuselage structure using piezoelectric actuators, known as active structural acoustic control (ASAC), to control cabin noise is then presented. Experimental results from laboratory ASAC tests conducted on a full-scale fuselage and from flight tests on a helicopter will be discussed. Finally, a hybrid active/passive noise control approach for achieving significant broadband noise reduction will be discussed. Experimental results of control of broadband noise transmission through an aircraft structure will be presented.

8:20

**4aNS2. Recent advances in active noise and vibration control at NASA Langley Research Center.** Gary P. Gibbs, Randolph H. Cabell, Daniel L. Palumbo, Richard J. Silcox, and Travis L. Turner (Structural Acoust. Branch, NASA LaRC, Hampton, VA 23681)

Over the past 15 years NASA has investigated the use of active control technology for aircraft interior noise. More recently this work has been supported through the Advanced Subsonic Technology Noise Reduction Program (1994–2001), High Speed Research Program (1994–1999), and through the Quiet Aircraft Technology Program (2000–present). The interior environment is recognized as an important element in flight safety, crew communications and fatigue, as well as passenger comfort. This presentation will overview research in active noise and vibration control relating to interior noise being investigated by NASA. The research to be presented includes: active control of aircraft fuselage sidewall transmission due to turbulent boundary layer or jet noise excitation, active control of interior tones due to propeller excitation of aircraft structures, and adaptive stiffening of structures for noise, vibration, and fatigue control. Work on actuator technology ranging from piezoelectrics, shape memory actuators, and fluidic actuators will be described including applications. Control system technology will be included that is experimentally based, real-time, and adaptive.

8:40

**4aNS3. Interior noise control through tuned loudspeaker absorption.** Robert L. Clark and Mark A. McEver (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., 301 Hudson Hall, Durham, NC 27708-0300)

In this work we describe the development and application of an interior noise control approach based on active loudspeaker tuning for sound absorption. Loudspeakers are tuned to minimize global sound pressure at specific acoustic modes through diaphragm velocity feedback. An experimental model of the acoustic space used for controller design is obtained through system identification.  $H_2$  optimal controllers are designed to minimize a performance microphone output by feeding back the tuned loudspeaker diaphragm velocity. By sensing speaker velocity rather than far-field sound pressure, the loudspeaker is actively tuned to minimize reflected sound at specific acoustic modes and therefore reduce the global pressure field. Experimental results, obtained from an acoustic enclosure modeled after a rocket fairing, are used to demonstrate the effectiveness of the tuned loudspeaker at global attenuation. Details of the design procedure and experimental applications are discussed.

9:00

**4aNS4. Cabin noise and weight reduction program for the Gulfstream G200.** C. Kearney Barton (Gulfstream Aerosp. Corp., P.O. Box 2206, Savannah, GA 31402)

This paper describes the approach and logic involved in a cabin noise and weight reduction program for an existing aircraft that was already in service with a pre-existing insulation package. The aircraft, a Gulfstream G200, was formally an IAI Galaxy, and the program was purchased from IAI in 2001. The approach was to investigate every aspect of the aircraft that could be a factor for cabin noise. This included such items as engine mounting and balancing criteria, the hydraulic system, the pressurization and air-conditioning system, the outflow valve, the interior shell and mounting system, antennae and other hull protuberances, as well as the insulation package. Each of these items was evaluated as potential candidates for noise and weight control modifications. Although the program is still ongoing, the results to date include a 175-lb weight savings and a 5-dB reduction in the cabin average Speech Interference Level (SIL).

9:20

**4aNS5. An in-flight study of cabin buzz-saw noise.** David Reed, Stefan Uellenberg, and Evan Davis (The Boeing Co., P.O. Box 3707-MC 67-ML, Seattle, WA 98124)

This paper examines the characteristics of multiple-pure-tone noise generated by high-speed turboprops under conditions of supersonic fan tip speeds, especially as it is observed in an aircraft passenger cabin. This phenomenon, also known as buzz-saw noise, is an important noise source in commercial airplane passenger cabins and has proved to be difficult to treat with sound-absorbing materials. Recent flight test experiments by The Boeing Company have demonstrated extraordinary success in suppressing cabin buzz-saw noise by strategic placement of engine inlet acoustic linings. The observed behavior is explained by a propagation and radiation model, which is validated by in-flight measurements made with a phased microphone array mounted on the fuselage skin of a Boeing 777. A structural acoustic model is also offered to explain the different transmission characteristics of buzz-saw noise and turbulent boundary layer excitation. Correlation length scales measured on the fuselage surface for these two noise sources are key inputs to the structural model.

9:40

**4aNS6. Structural acoustic modeling of aircraft fuselage structures.** Ralph Buehrle, Jacob Klos, Jay Robinson (NASA Langley Res. Ctr., Structural Acoust. Branch, M.S. 463, Hampton, VA 23681), and Ferdinand Grosveld (Lockheed Martin Eng. and Sci., Hampton, VA)

Recent studies at NASA Langley Research Center have examined the development and validation of finite element and boundary element modeling techniques for the prediction of structural acoustic response of aircraft fuselage structures. The goal of this work is to provide increased confidence in the modeling techniques so that interior noise criteria can be incorporated early in the design process. These efforts have focused on the development and validation of high-fidelity physics-based numerical models for structural acoustic predictions into the kilohertz region. Finite element models were developed based on the geometric and material properties of the aircraft fuselage structures. Experimental modal analysis and point force frequency response functions were used to validate and refine the finite element models. Once validated, the finite element predictions of the velocity response were used as boundary condition input for boundary element predictions of the radiated sound power. Experiments in the Structural Acoustic Loads and transmission (SALT) Facility at NASA Langley were used to validate the acoustic predictions. Numerical and experimental results will be presented for conventional aluminum rib and stringer-stiffened aircraft structures, a honey comb composite sidewall panel, and damped acrylic windows. Numerical predictions were in good agreement with the experimental data.

10:00–10:10 Break

10:10

**4aNS7. Uncertain structural dynamics of aircraft panels and fuzzy structures analysis.** Victor W. Sparrow (Grad. Prog. in Acoust., Penn State Univ., 316B Leonhard Bldg., University Park, PA 16802, vws1@psu.edu) and Ralph D. Buehrle (NASA Langley Res. Ctr., Hampton, VA 23681)

Aircraft fuselage panels, seemingly simple structures, are actually complex because of the uncertainty of the attachments of the frame stiffeners and longitudinal stringers. It is clearly important to understand the dynamics of these panels because of the subsequent radiation into the passenger cabin, even when complete information is not available for all portions of the finite-element model. Over the last few years a fuzzy structures analysis (FSA) approach has been undertaken at Penn State and NASA Langley to quantify the uncertainty in modeling aircraft panels. A new MSC.Nastran [MSC Software Corp. (Santa Ana, CA)] Direct Matrix Abstraction Program (DMAP) code was written and tested [AIAA paper 2001-1320, 42nd AIAA/ASME/ASCE/AHS/ASC Structures, Structural Dynamics, and Materials Conf., Seattle, WA, 16 April 2001] and was applied to simple fuselage panel models [J. Acoust. Soc. Am. **109**, 2410(A) (2001)]. Recently the work has focused on understanding the dynamics of a realistic aluminum fuselage panel, typical of today's aircraft construction. This presentation will provide an overview of the research and recent results will be given for the fuselage panel. Comparison between experiments and the FSA results will be shown for different fuzzy input parameters. [Work supported by NASA Research Cooperative Agreement NCC-1-382.]



10:30

**4aNS8. New analysis methods for the calculation of interior noise.** Donald B. Bliss and Linda P. Franzoni (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708, dbb@duke.edu)

Recently developed analysis methods applicable to the prediction of aircraft interior noise are described. Analytical-numerical matching (ANM) and Local-global homogenization (LGH) are methods for the low- to mid-frequency range. Energy-intensity boundary elements are applicable for high frequencies. ANM is a method that replaces the effect of structural discontinuities (ribs, stringers, attachments) with dynamically equivalent smooth force distributions, thereby removing the need for high-resolution numerics in the overall computational problem. LGH, which applies for periodic and quasiperiodic structures, provides a means to solve directly for the long wavelength spatial content of the structure. This part of the structural motion is strongly coupled most to the interior sound field. Energy-intensity boundary elements provide a very efficient means to predict high-frequency broadband sound fields in a fundamentally correct manner. Current research on this boundary element method involves its extension to handle coupling with structural surfaces vibrating at high frequency. The strengths and limitations of each of the above methods is discussed in the context of the interior noise problem. Innovative methods for the reduction of interior noise, as suggested by these approaches, are also mentioned.

10:50

**4aNS9. A prototype spherical array for interior noise investigations.** Earl G. Williams, Brian H. Houston (Code 7130, Naval Res. Lab., Washington, DC 20375), and Peter C. Herdic (SFA, Inc., Landover, MD)

A spherical array of radius 0.2 m with 18 uniformly distributed microphone elements on a spherical surface has been constructed. Using the theory of nearfield acoustical holography the instantaneous pressure, velocity, and intensity fields are reconstructed throughout a volume of 0.6 m radius centered at the array origin in the frequency range of 0–600 Hz. Since the measurements are instantaneous the array is intended to be used to measure distributed sources that need not be coherent or stationary, ideally suited for source identification in the interior of an in-flight aircraft, an automobile or in interior spaces in naval vessels. Preliminary experiments with a loudspeaker demonstrate the accuracy of the volume-field reconstruction of the instantaneous fields. This array is a prototype of an array with many more elements which will have a wider frequency range and finer spatial resolution. [Work supported by NASA and ONR.]

### *Contributed Papers*

11:10

**4aNS10. Recent developments in statistical energy analysis for aircraft interior noise.** Paul Bremner, Michael Bloor (Vibro-Acoust. Sci., Inc., 12555 High Bluff Dr., Ste. 310, San Diego, CA 92130), and John F. Wilby (Wilby Assoc.)

Price and Crocker [J. Acoust. Soc. Am. **47** (1970)] have previously shown that Statistical Energy Analysis (SEA) is a robust tool for sound transmission loss problems in room acoustics. This paper presents recent developments in application of SEA to the structural-acoustic design of aircraft for control of interior noise. Developments for this application include the means to model boundary layer excitation of the fuselage; the means to model the dynamic response of both metallic-ribbed and composite sandwich pressurized shells; and the means to model the full noise and vibration reduction performance of multilayered interior trim and insulation treatments. The method has proven effective for acoustic design evaluation of new materials and constructions of aircraft structure or interiors but is equally attractive for optimization of the cost, weight, and packaging space of noise control treatments on aircraft of more conventional construction. Results of recent validation studies will be presented.

11:25

**4aNS11. Application of statistical energy analysis to the design of crew rest compartments.** Mark M. Gmerek (The Boeing Co., Seattle, WA 98124, mark.m.gmerek@boeing.com)

Longer flight times for modern commercial aircraft have led to the need for crew rest compartments. Noise levels in the crew rest compartments must be conducive to proper rest and recuperation. Statistical Energy Analysis (SEA) has been used to develop a 777 crew rest compartment design that achieves appropriate noise levels at acceptable weight and cost. In this paper the design of a 777 overhead crew rest compartment is outlined using SEA design tools and methods. Noise data were gathered in flight to distinguish airplane source components and develop model inputs. Crew rest panels, the airplane fuselage, and acoustic volumes were modeled as SEA subsystems by taking into account geometry, material

properties, modulus, and damping. A model was built, excited with inputs, and analyzed to determine energy flow paths and acoustic pressure at receiver locations. Prospective add-on treatments were then assessed to engineer an effective noise control package. The model development was supplemented by laboratory sound transmission loss testing of individual components. The good agreement between the laboratory tests and individual SEA models of the components increased confidence in the approach. Once the crew rest was installed on the airplane, the measured in-flight noise levels closely matched the SEA estimates.

11:40

**4aNS12. The analysis of a state-switched absorber design concept.** Anne-Marie Albanese and Kenneth Cunefare (School of Mech. Eng., Georgia Tech, Atlanta, GA 30332-0405)

A tuned vibration absorber (TVA) is a spring-damper-mass system used in many industries for the suppression of a specific vibration frequency, and has application for the suppression of aircraft fuselage vibration. A state-switched absorber (SSA) is similar to a TVA, except that one or more components in the SSA is able to instantaneously and discretely change properties, thus increasing the effective bandwidth of vibration suppression. In order to design a replacement SSA for the classic TVA, the SSA must operate in the appropriate frequency range, be lightweight, and compact. An optimal SSA will also have a maximal frequency range between which it can switch. This paper discusses the development of a magnetorheological (MR) silicone gel used as the SSA switching element, the shape required to maintain a magnetic flux path, and the contribution of the magnet-mass to frequency shifting. The MR gel is iron-doped silicone, cured in the presence of a magnetic field. During operation, the applied magnetic flux is modified to change the natural frequency. The applied flux requirement forces the SSA to be a small ring. The SSA is designed to operate below 100 Hz.

**4aNS13. Experimental evaluation of sound transmission through a thin-wall structure produced by turbulent pressure fluctuations.** Boris M. Efimtsov (Acoust. Div., Central AeroHydrodynamics Inst. (TsAGI), 17 Radio St., 105005 Moscow, Russia, ayzverev@hotmail.com) and Sergey N. Baranov (RUSAVIA Res. Group., 125299 Moscow, Russia)

Problems of physical modeling of sound transmission through a thin-wall structure produced by turbulent pressure fluctuations under excitation

are presented. A method of evaluating the sound transmission loss of a thin-wall structure from turbulent pressure fluctuations on the basis of experimental data is proposed. The experiment is associated with full scale tests of structure in two adjacent reverberation chambers and with model tests in two adjacent model reverberation chambers and in the wind tunnel equipped with a similar model sound chamber. An example of such method realization is presented and this indicates a possibility of its practical realization in predicting the noise inside the passenger aircraft, produced by the turbulent boundary layer on its exterior surface. [Work supported by INTAS Grant No. 99-0088.]

THURSDAY MORNING, 5 DECEMBER 2002

CORAL GALLERY FOYER, 9:00 TO 11:40 A.M.

## Session 4aPAa

### Physical Acoustics: Laser Ultrasonics

Todd W. Murray, Chair

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

#### Invited Papers

9:00

**4aPAa1. Ultrafast optoacoustic wave generation effects on picosecond-timescale reflectivity measurements in metal thin films.**

James B. Spicer (Johns Hopkins Univ., Rm. 102, Maryland Hall, 3400 N. Charles St., Baltimore, MD 21218, spicer@jhu.edu) and Christopher J. K. Richardson (Univ. of Maryland, College Park, MD 20740)

The increase in temperature at the surface of a metal during absorption of optical energy is a dynamic, multistep process. High-time-resolution, ultrafast measurements of thermal and elastic transients have proven to be useful for investigating the structure, thermophysical properties, and elastic properties of thin films [Paddock and Eesley, *J. Appl. Phys.* **60**, 285–290 (1986)]. During ultrafast optical excitation of metals, the rapid deposition of energy causes the absorbing electrons to enter a superheated state, dispersing beyond the excitation volume before energy is deposited to the lattice with the rate of energy transfer between electrons and phonons being described using the electron–phonon coupling parameter. Typically, ultrafast reflectivity measurements taken immediately after excitation can be modeled using a completely thermal analysis if reduced values for the electron–phonon coupling parameter are used. These reduced values often vary substantially [Gusev and Wright, *Phys. Rev. B* **57**, 2878–2888 (1998)] from accepted values that are experimentally determined from measurements using the cooling rate of superheated electrons. By including elastic contributions to the modeled signal at all times, agreement with measured signals can be obtained for short and long times using values for materials parameters that are in agreement with those obtained from other techniques.

9:25

**4aPAa2. Dispersion and soliton formation from longitudinal acoustic phonons in crystalline solids.** Humphrey J. Maris, Hsin-Yi Hao, and Wisit Singhsomroje (Dept. of Phys., Brown Univ., Providence, RI 02912)

Experiments are described in which the picosecond ultrasonic technique is used to study the effects of dispersion and elastic nonlinearity on the shape of longitudinal acoustic pulses propagating in crystalline solids. In these experiments, a subpicosecond light pulse (pump pulse) is used to generate an acoustic pulse at one surface of the sample. After propagation through the sample, the shape of this acoustic pulse is modified. When the amplitude of the pump pulse is sufficiently low, nonlinear effects can be neglected and a measurement of the acoustic pulse shape can be used to determine the phonon dispersion. This measured phonon dispersion is compared with various lattice dynamical models. When the pump pulse intensity is high, the nonlinear effects become strong enough to balance the dispersion, and acoustic solitons emerge from the original acoustic pulse. Measurements of the characteristics of these solitons are in agreement with the results of computer simulations based on the Korteweg-de Vries equation.

9:50

**4aPAa3. Picosecond transient grating acoustic measurements of thin films, microfluidic networks, and microphononic bandgap structures.** John Rogers (Bell Labs., 600 Mountain Ave., Rm. 1D-332, Murray Hill, NJ 07974)

We describe some new methods for transient grating (TG) photoacoustic measurements, and summarize their use in high frequency acoustic waveguide analysis of a variety of samples with dimensions smaller than 100 microns. This research is motivated in part by the growing interest in micro- and nanostructures whose mechanical and acoustic properties are important either for basic study or device design. Although the conventional TG technique provides a convenient method for exciting and detecting acoustic waves in bulk samples, it is not well suited to microsystems, where signal levels can be low and acoustic dispersion (i.e., variation in acoustic velocity with wavelength) can be pronounced. Here we illustrate how specially designed diffractive optics and imaging lenses remove these limitations by enabling convenient means for: (i) generating and overlapping many (up to 10 has been demonstrated)

excitation pulses for launching complex, user-definable acoustic waveforms that enable single-shot measurements of dispersion; and (ii) providing a phase-stabilized heterodyne detection scheme that enhances the sensitivity of the measurement and simplifies many of the alignment procedures. We describe our application of these methods to a range of samples, including thin films and multilayer assemblies, optical fibers, microfluidic networks, and microphononic bandgap structures.

#### 10:15–10:30 Break

#### 10:30

**4aPAa4. Laser ultrasonic system for on-line steel tube gauging and process control.** Daniel Levesque, Marc Choquet, Christian Padioleau, Christian Neron, Christian Corbeil, Richard Talbot, Abdelhakim Bendada, Jean-Pierre Monchalin (Industrial Mater. Inst., Natl. Res. Council of Canada, 75 de Mortagne Blvd., Boucherville, QC J4B 6Y4, Canada), Robert V. Kolarik II, Gerald V. Jeskey, Erich D. Dominik, Larry J. Duly, Kenneth J. Samblanet, Steven E. Agger, Kenneth J. Roush, and Michael L. Mester (The Timken Co., Canton, OH 44706)

A laser ultrasonic system has been installed on a seamless tubing production line at The Timken Company and is being used to measure on-line the wall thickness of tubes during processing. The seamless process consists essentially in forcing a mandrel through a hot cylindrical billet in rotation and results in wall thickness variations that should be minimized and controlled to respect specifications. The system includes a Q-switched Nd-YAG laser for the generation of ultrasound by ablation, a long pulse very stable Nd-YAG laser for detection coupled to a confocal Fabry–Perot interferometer. The lasers, data acquisition, and processing units are housed in a cabin off-line and connected to a front coupling head located over the passing tube by optical fibers. The system also includes a fiber-coupled pyrometer to measure tube temperature profile and two fiber-coupled optical velocimeters to measure the coordinates at the probing location on the surface of the passing, rotating hot tube. During the presentation further details of the system will be disclosed, as well as typical results and examples of its diagnostic capability. [Work partially supported by Department of Energy under Award No. DE-FC07-99ID 13651.]

### Contributed Papers

#### 10:55

**4aPAa5. Rayleigh scattering and heterodyne detection as a nonintrusive microphone.** Cesar Aguilar, Carlos Azpeitia, Andres Porta, and Catalina Stern (Lab. de Acustica, Depto Fisica, Fac. Ciencias, UNAM, Mexico)

Heterodyne detection of a monochromatic laser beam scattered by molecules in a transparent gas gives a signal that is proportional to the spatial Fourier transform of density fluctuations for a wave vector determined by the optics. In a turbulent air jet, density fluctuations are due to either acoustic waves or to entropy fluctuations. This technique captures both. The method can then be used as a nonintrusive microphone to study the propagation of acoustic waves inside the jet. At each point in the flow the direction of propagation of the acoustic wave is determined, and the acoustic field can be described. This will eventually help to localize the sources of the waves in the flow and hopefully determine the hydrodynamic event that generates these waves.

#### 11:10

**4aPAa6. Laser ultrasonic in-process inspection of paper for elastic properties.** Paul Ridgway, Richard Russo (Lawrence Berkeley Natl. Lab., 1 Cyclotron Rd., Berkeley, CA 94720), Emmanuel Lafond, Charles Habeger, Jr., and Ted Jackson (Inst. of Paper Sci. and Technol., Atlanta, GA 30318)

A laser-ultrasonic (LUS) sensor has been developed that allows measurement of the bending stiffness (BS) and shear rigidity (SR) of paper and paperboard as it is being made on the papermaking machine. A prototype system was recently tested in a paper mill at web speeds up to 5000 ft/min with excellent precision and accuracy. The LUS technique performs well on paper and board with basis weights up to 130 g/m<sup>2</sup>. Several laboratory methods exist for measuring the bending stiffness in small samples of paper and board. Currently, no commercial method exists for nondestructively measuring this property on the papermaking machine at production speeds. Commercial instruments using contact transducers measure “tensile strength orientation” (TSO) on heavier boards, where

marking of the sheet by the contact transducers is not of concern. Unlike contact ultrasonic techniques, LUS does not visibly mark even the lightest grade papers. Contact ultrasonic measurements correlate approximately to the tensile strength of the sheet and can be used to calculate an approximate value for BS. LUS measurements are directly related to BS and should yield more accurate determinations. Optimum use of feed stock, reduced waste, and decreased energy consumption are potential benefits of the LUS technology.

#### 11:25

**4aPAa7. Characterization of graded coatings using laser generated acoustic waves.** Oluwaseyi O. Balogun, Todd W. Murray (Aerosp. and Mech. Eng. Dept., Boston Univ., 110 Cummington St., Boston, MA 02215), and Soumendran N. Basu (Boston Univ., Brookline, MA 02446)

Theoretical and experimental results are presented for the pulsed laser generation of ultrasound in functionally graded coating materials. The case of 9–12 m compositionally graded mullite (3Al<sub>2</sub>O<sub>3</sub>:2SiO<sub>2</sub>) environmental barrier coatings in which the Al composition varies as a function of depth is considered in detail. The laser source is seen to generate both the lowest order Rayleigh mode and Sezawa mode in the coatings. The waveforms are processed using the 2-D FFT technique to extract the dispersion curves. Theoretical dispersion curves and time domain signals generated in coatings with linear, sigmoidal, and exponential compositional gradations will be presented and the sensitivity of the wave modes to elastic property gradation explored. The graded coatings are treated as multilayer materials with a sufficient number of layers chosen to achieve convergence of the dispersion curves for a given elastic property gradation. The (depth dependent) elastic properties of the coatings are evaluated using the simplex optimization routine and the functional form of the property variation is correlated with the measured compositional profiles. This work demonstrates that laser based ultrasonic measurements of acoustic modes in functionally graded EBCs provides a useful tool for extracting the depth dependent elastic properties.

**Session 4aPAb****Physical Acoustics: Global Infrasound Monitoring I**

Henry E. Bass, Chair

*National Center for Physical Acoustics, University of Mississippi, Coliseum Drive, University, Mississippi 38677***Chair's Introduction—10:10****Invited Papers****10:15****4aPAb1. The verification system of the Comprehensive Nuclear-Test-Ban Treaty.** Gerardo Suarez (Intl. Monitoring Systems, Vienna Intl. Ctr., P.O. Box 1200, A-1400 Vienna, Austria, gerardo.suarez@ctbto.org)

The Comprehensive Nuclear-Test-Ban Treaty was opened for signature in September 1996. To date, the treaty has been signed by 165 countries and ratified by 93; among the latter, 31 out of the 44 whose ratification is needed for the treaty to enter into force. The treaty calls for the installation and operation of a verification system to ensure compliance. The verification system is composed of the International Monitoring System (IMS), the International Data Centre (IDC), and the On Site Inspection Division (OSI). The IMS is a global network of 321 stations hosted by 90 countries. The primary network is composed of 50 seismic stations, 31 of which are seismic arrays and 19 three-component, broad-band stations, 11 hydroacoustic stations, 60 infrasound arrays, and 80 radionuclide monitoring stations measuring radioactive particulates and noble gases in the atmosphere. The radionuclide network is supported by 16 laboratories. The auxiliary network of 120 seismic stations is interrogated on request by the IDC to improve the accuracy of the locations. The data from the 321 stations and from the laboratories is transmitted to the IDC in Vienna via a dedicated Global Communication Infrastructure (GCI) based on VSAT antennas. The IDC collects and processes the data collected from the four technologies and produces bulletins of events. The raw data and bulletins are distributed to state signatories. Upon entry into force, an on-site inspection may be carried out if it is suspected that a nuclear explosion has taken place. Since mid-1997, when the Provisional Technical Secretariat responsible for the implementation of the verification system began its work in Vienna, over 86% of the sites have been surveyed and the final location of the stations selected. By the end of 2002 this number will reach about 90%, essentially completing this phase. To date, 131 stations have been built or upgraded, and 80 are now sending data to the IDC; 112 others are under construction or under negotiation. Over 392 authorized users from 53 state signatories are now receiving data and products from the IDC. In addition, under the auspices of the signatories of the treaty, several projects are being funded to improve the calibration of seismic stations, the software used routinely at the IDC, and the analysis of radionuclides, among others.

**10:35****4aPAb2. Monitoring of atmospheric nuclear explosions with infrasonic microphone arrays.** Charles R. Wilson (Geophysical Inst., Univ. of Alaska, Fairbanks, AK 99775)

A review is given of the various United States programs for the infrasonic monitoring of atmospheric nuclear explosions from their inception in 1946 to their termination in 1975. The US Atomic Energy Detection System (USAEDS) monitored all nuclear weapons tests that were conducted by the Soviet Union, France, China, and the US with arrays of sensitive microbarographs in a worldwide network of infrasonic stations. A discussion of the source mechanism for the creation and subsequent propagation around the globe of long wavelength infrasound from explosions (volcanic and nuclear) is given to show the efficacy of infrasonic monitoring for the detection of atmospheric nuclear weapons tests. The equipment that was used for infrasound detection, the design of the sensor arrays, and the data processing techniques that were used by USAEDS are all discussed.

**10:55****4aPAb3. Processing and analysis of infrasound data at the Comprehensive Nuclear Test Ban Treaty International Data Center.** Robert G. North and Robert L. Woodward (Sci. Appl. Intl. Corp., Ctr. for Monitoring Res., 1300 N. 17th St., Ste. 1450, Arlington, VA 22209)

The Center for Monitoring Research (CMR) in Arlington, VA served as a prototype of the International Data Center (IDC) that is specified in the Comprehensive Nuclear Test Ban Treaty (CTBT). The CMR developed the software that is now being used by the IDC that has been established in Vienna to acquire, process, and analyze data from the global networks of hydroacoustic, infrasonic, radionuclide, and seismic sensors that are being installed to assist verification of the CTBT. The prototype IDC tested this software and related procedures very extensively with seismic and radionuclide data, but the infrasonic aspects of the system are less advanced due to the limited number of sensors available and a less complete knowledge of sources and signal propagation. The infrasonic processing

system attempts to detect signals of interest by combining spatial coherence and power, and estimates signal arrival time, direction of approach, velocity, and other parameters. After a signal is detected, further processing is carried out to try to identify “phases” that correspond to reflections from different atmospheric layers. Finally, detections at multiple sensors of all types (seismic and hydroacoustic as well as infrasonic) are combined to estimate the time and location of potential sources.

### Contributed Papers

11:15

**4aPAb4. Status report on the establishment of the Comprehensive Nuclear-Test-Ban Treaty (CTBT) International Monitoring System (IMS) infrasound network.** J. A. Vivas Veloso, D. R. Christie, P. Campus, M. Bell, T. L. Hoffmann, A. Langlois, P. Martysevich, E. Demirovik, J. Carvalho, A. Kramer, and A. Moleró Alonso (CTBTO, VIC, Wagramer Str. 5, 1400 Vienna, Austria)

The infrasound component of the International Monitoring System (IMS) for Comprehensive Nuclear-Test-Ban Treaty verification aims for global detection and localization of low-frequency sound waves originating from atmospheric nuclear explosions. The infrasound network will consist of 60 array stations, distributed as evenly as possible over the globe to assure at least two-station detection capability for 1-kton explosions at any point on earth. This network will be larger and more sensitive than any other previously operated infrasound network. As of today, 85% of the site surveys for IMS infrasound stations have been completed, 25% of the stations have been installed, and 8% of the installations have been certified and are transmitting high-quality continuous data to the International Data Center in Vienna. By the end of 2002, 20% of the infrasound network is expected to be certified and operating in post-certification mode. This presentation will discuss the current status and progress made in the site survey, installation, and certification programs for IMS infrasound stations. A review will be presented of the challenges and difficulties encountered in these programs, together with practical solutions to these problems.

11:30

**4aPAb5. Post-installation activities in the Comprehensive Nuclear Test Ban Treaty (CTBT) International Monitoring System (IMS) infrasound network.** J. A. Vivas Veloso, D. R. Christie, T. L. Hoffmann, P. Campus, M. Bell, A. Langlois, P. Martysevich, E. Demirovik, J. Carvalho, A. Kramer, and A. Moleró Alonso (CTBTO, VIC, Wagramer Str. 5, 1400 Vienna, Austria)

The provisional operation and maintenance of IMS infrasound stations after installation and subsequent certification has the objective to prepare the infrasound network for entry into force of the Comprehensive Nuclear-Test-Ban Treaty (CTBT). The goal is to maintain and fine tune the technical capabilities of the network, to repair faulty equipment, and to ensure that stations continue to meet the minimum specifications through evaluation of data quality and station recalibration. Due to the globally dispersed nature of the network, this program constitutes a significant undertaking that requires careful consideration of possible logistic approaches and their financial implications. Currently, 11 of the 60 IMS infrasound stations are transmitting data in the post-installation Testing & Evaluation mode. Another 5 stations are under provisional operation and are maintained in post-certification mode. It is expected that 20% of the infrasound network will be certified by the end of 2002. This presentation will focus on the different phases of post-installation activities of the IMS infrasound program and the logistical challenges to be tackled to ensure a cost-efficient management of the network. Specific topics will include Testing & Evaluation and Certification of Infrasound Stations, as well as Configuration Management and Network Sustainment.

THURSDAY MORNING, 5 DECEMBER 2002

CORAL GALLERY 1, 8:00 TO 11:20 A.M.

### Session 4aSA

## Structural Acoustics and Vibration: Reconstruction of Acoustic Radiation from Vibrating Structures I

Sean F. Wu, Chair

*Department of Mechanical Engineering, Wayne State University, 505 Anthony Wayne Drive, Detroit, Michigan 48202*

Chair's Introduction—8:00

### Invited Papers

8:05

**4aSA1. Patch nearfield acoustical holography.** Earl G. Williams (Code 7137, Naval Res. Lab., Washington, DC 20375, williams@pa.nrl.navy.mil)

Nearfield acoustical holography (NAH) requires the measurement of the pressure field over a complete surface in order to recover the normal velocity on a nearby vibrating concentric surface. Patch NAH provides a major advancement by eliminating the need for complete surface pressure scans—only a small area needs to be scanned to determine the normal velocity on the corresponding (small area) concentric patch on the vibrator. The theory of patch NAH is based on: (1) an analytic continuation of the patch pressure which provides a spatially tapered aperture extension of the field; and (2) a decomposition of the transfer function (pressure to velocity and/or pressure to pressure) between the two surfaces using the singular value decomposition for general shapes and the fast Fourier transform for planar surfaces. Inversion of the transfer function is stabilized using Tikhonov regularization and the Morozov discrepancy principle. Experimental results show that the normal velocity reconstruction errors (rms) for a point-driven vibrator over 200–2700 Hz average less than 20% for two surfaces 0.4 cm apart. Reconstruction of the active normal acoustic intensity was also achieved, with less than 30% rms error over the frequency band. [Work supported by ONR.]



**4aSA2. Reconstructing acoustic radiation using eigenfunction expansions.** J. D. Maynard (Phys. Dept., Penn State Univ., University Park, PA 16802)

A powerful tool for reconstructing the sound radiation from a vibrating structure is near-field acoustic holography. In this method, the sound pressure is measured at points near the vibrating structure, and these data are analyzed by computer to determine any desired feature of the entire sound field, including acoustic energy sources at the vibrating structure. Some versions of this method have involved first solving the “forward problem,” which consists of finding the radiated sound field, assuming that the surface velocity of the vibrating structure is known. Usually such problems are solved using acoustic boundary elements. However, we have been studying a method for solving the reconstruction problem and by-passing the forward problem by using eigenfunction expansions. Furthermore, Sean Wu has developed a significant extension of the technique for solving the forward problem, and this method may challenge the boundary element method. In this talk, the fundamentals of the eigenfunction expansion method will be reviewed, and results of some practical tests will be presented.

9:05

**4aSA3. Nearfield acoustic holography and its evolution.** Sean Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202)

This paper reviews the evolution of nearfield acoustic holography (NAH) over the past two decades. The original NAH (Williams and Maynard, 1980) could only reconstruct acoustic radiation from a surface containing a level of constant coordinate in an exterior region. Soon a generalized NAH based on the Helmholtz integral theory and boundary element method (BEM) was developed to reconstruct acoustic radiation from arbitrarily shaped surfaces. This BEM-based NAH is advantageous in many aspects, but has several inherent shortcomings. An alternative is to expand the acoustic field in terms of the spherical wave functions, the most prominent one being the Helmholtz equation-least squares (HELs) method. This HELs method offers a great flexibility and versatility in reconstructing the acoustic fields in both exterior and interior regions, and allows reconstruction of transient acoustic radiation from impulsively accelerated objects. Meanwhile, we have seen the extension of NAH to characterize acoustic sources of a moving object and variations of NAH based on measurements of acoustic intensity rather than pressures. Many techniques are developed along the way to tackle the ill-posedness difficulties inherent in this inverse acoustic problem. Progression of able researchers is transforming NAH into an ever more powerful diagnostic tool. [Work supported by NSF.]

### Contributed Papers

9:35

**4aSA4. On the validity and convergence of the Helmholtz equation least squares solutions.** Tatiana Semenova and Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI 48202)

The problem of reconstruction of acoustic radiation from arbitrary shaped vibrating bodies is often encountered in engineering applications. The Helmholtz equation least-squares (HELs) method proposed by Wu (2000) is an assumed-form solution method in which the reconstructed acoustic pressure is found as a superposition of spherical waves. The unknown coefficients are determined by matching the assumed-form solution to the measured acoustic pressures at a number of points around the source and their errors are minimized by the least-squares method. To further minimize the errors of reconstruction, constraints are used on the surface of the object. In this paper a two-dimensional, noncircular object is considered and the two-dimensional Helmholtz equation is solved. The locations of singularities in the analytic continuation of the solution across the source surface are found using the Schwarz function of the source surface. It will be shown that the HELs solution remains bounded and close to the true pressures inside the minimum circle enclosing the object as well as outside. Other types of modifications of the HELs method for reconstruction of acoustic radiation in an exterior two-dimensional region will be considered. [Work supported by NSF.]

9:50

**4aSA5. Acoustic imaging of harmonic near-field sources from surface pressure measurements on a body using singular value decomposition.** Peter R. Stepanishen (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882) and Irsan Brodjonegoro (Bandung Inst. of Technol., Indonesia)

An inverse or backward projection method based on a combined Greens function and singular value decomposition method is developed to locate and to determine the strength of near field harmonic sources from the acoustic field on the surface of a nearby rigid body. A resolution matrix, which is based on the free space Greens function, the geometry of

the measurement surface of interest and the field locations of interest, is introduced to determine the resolution and accuracy of the backward projection method. Point source distributions located above a rigid planar surface are addressed both analytically and numerically. In addition, line source distributions located outside of infinite rigid circular and elliptical cylinders are also addressed. It is demonstrated that the method is able to identify the location and to determine the strength of harmonic near-field sources which are separated by less than 0.01 of a wavelength.

10:05–10:20 Break

10:20

**4aSA6. The reconstruction of the acoustical field over nonconformal surfaces.** Nicolas Valdivia and Earl G. Williams (Naval Res. Lab., Code 7130, 4555 Overlook Ave., SW, Washington, DC 20375-5320, valdivia@pa.nrl.navy.mil)

Consider the inverse problem of reconstructing the acoustical field over a surface  $S_v$  by measuring the acoustical pressure over an interior surface  $S_p$ . Ideally the measurement surface  $S_p$  is conformal to the reconstruction surface  $S_v$ , and close to  $S_v$ . In this work the reconstruction surface  $S_v$  will be a box, and the interior surface  $S_p$  will be a small sphere inside the box. We study the solution to this inverse problem with two different methods: Near-field Acoustical Holography (NAH) and Helmholtz Equation Least Squares (HELs). We will compare the reconstruction of the two methods using two test solutions. The first test solution will be a point source, and the second test solution will be a plate vibrator in one side of the box. In addition we study the inclusion of white noise to the test solution and regularization methods for its recovery. [Work supported by ONR.]

10:35

**4aSA7. Health monitoring using Lamb waves. Application to the characterization of low impacts in aerospace structures.** Sebastien Grondel, Jamal Assaad, and Christophe Delebarre (IEMN, DOAE, UMR CNRS 9929, Universite de Valenciennes et du Hainaut Cambresis, Le Mont Houy, 59313 Valenciennes Cedex 9, France)

This work was devoted to the development of a health monitoring system assigned to aerospace applications. The application concerned the detection of low damaging impacts on composite structures due to the extreme sensitivity of this material to this kind of solicitation. The chosen health monitoring was based first on the excitation and reception of Lamb waves along the structure by using thin piezoelectric transducers (active monitoring) and second on a continuous monitoring with the same transducers (passive monitoring). Preliminary tests were performed on a composite plate and the sensitivity of the A0 mode to the damage was demonstrated. Concerning the passive monitoring, the study showed the ability to use the acoustic signature of the impact to detect a possible damage. Further tests were performed on a wingbox composite structure. This wingbox structure consisted of composite skins with variable thickness mounted onto a metallic substructure. Moreover, four stringers were bonded on these skins. The aim was to demonstrate the ability of the system to perform rapid inspections of complex structures. After the application of a serial of impacts at different locations of the composite wingbox, the feasibility of the health monitoring was demonstrated.

10:50

**4aSA8. Reconstruction of transient behavior of a nonuniformly vibrating transducer.** Oleg A. Sapozhnikov (Dept. of Acoust., Faculty of Phys., M. V. Lomonosov Moscow State Univ., Leninskie Gory, Moscow 119992, Russia)

An accurate prediction of a transient acoustic field radiated from piezoceramic transducers is important for many applications. The field can be calculated if the normal velocity distribution along the transducer surface is known. However, up to now, there are no reliable direct methods of surface vibration measurement in liquids. The well-developed laser vibro-

meter method can be employed only when the transducer is in contact with vacuum or gas because of strong acousto-optic interaction in condense media. In this talk, a holographic method is proposed to reconstruct the radiator vibration. The method includes a measurement of the temporal waveform in different points of a plane grid perpendicular to the acoustic axis at some distance from the source, theoretical time reversal of the waveform in each grid-point, and back propagation of the field to the source using the transient form of the Rayleigh integral. The reconstruction procedure is described and accuracy of the method is studied depending on grid-step, distance to the source, and dimensions of the grid. [Work supported by CRDF RP2-2384-MO-02 and RFBR 02-02-17029.]

11:05

**4aSA9. Ecological prognosis near intensive acoustic sources.** Stanislav A. Kostarev (Lab. of Acoust. and Vib. Tunnel Assoc., 21 Sadovo-Spasskaya Str., Moscow 107217, Russia), Sergey A. Makhortykh (Russian Acad. of Sci., Pushchino, Moscow Region 142290, Russia), and Samuil A. Rybak (N. N. Andreev Acoust. Inst., Moscow 117036, Russia)

The problem of a wave-field excitation in a ground from a quasiperiodic source, placed on the ground surface or on some depth in soil is investigated. The ecological situation in this case in many respects is determined by quality of the raised vibrations and noise forecast. In the present work the distributed source is modeled by the set of statistically linked compact sources on the surface or in the ground. Changes of parameters of the media along an axis and horizontal heterogeneity of environment are taken into account. Both analytical and numerical approaches are developed. The latter are included in the software package *VibraCalc*, allowing to calculate distribution of the elastic waves field in a ground from quasilinear sources. Accurate evaluation of vibration levels in buildings from high-intensity underground sources is fulfilled by modeling of the wave propagation in dissipative inhomogeneous elastic media. The model takes into account both bulk (longitudinal and shear) and surface Rayleigh waves. For the verification of the used approach a series of measurements was carried out near the experimental part of monorail road designed in Moscow. Both calculation and measurement results are presented in the paper.

THURSDAY MORNING, 5 DECEMBER 2002

GRAND CORAL 3, 8:00 TO 11:30 A.M.

### Session 4aSC

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

Catherine L. Rogers, Cochair

*Communication Science and Disorders, University of South Florida, 4202 East Fowler Avenue, Tampa, Florida 33620-8150*

Paolo Escudero, Cochair

*School of Communication Science and Disorders, McGill University, 1266 Pine Avenue, West, Montreal, Quebec H3G 1A4, Canada*

### Contributed Papers

All posters will be on display from 8:00 a.m. to 11:30 a.m. 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 9:45 a.m. and contributors of even-numbered papers will be at their posters from 9:45 a.m. to 11:30 a.m.

**4aSC1. Consonant contrasts in a multiple, simultaneous word identification task.** Mark Ericson (Air Force Res. Lab., 2610 Seventh St., Wright-Patterson AFB, OH 45433) and Pamela Mishler (Dept. of Veteran Affairs, Dayton, OH)

People often have difficulty communicating in the presence of concurrent conversations. This well-known "cocktail party effect" can be described as a combination of energetic and informational masking. The purpose of this study was to measure and model the effects of energetic

and informational masking that occur when two people speak at the same time. The word identification test used in the experiments was the Modified Rhyme Test (MRT), which afforded a multitude of initial and final consonant pairs when phrases were played simultaneously. The MRT was used as both the stimulus and the masker, which were presented monaurally at 75 dB SPL over Sennheiser HD-520 headphones to four normal hearing listeners. The independent variables included 30 pairs of MRT word lists, and twelve pairs of same-sex talkers. The dependent variable

was speech (consonant) intelligibility as measured with the MRT. Results will be discussed in the context of place of articulation, manner of articulation, signal-to-noise-ratio of the phoneme pairs, and informational masking model predictions.

**4aSC2. Realistic listening improved by adding fine structure.** Ginger S. Stickney, Kaibao Nie, and Fan-Gang Zeng (Dept. of Otolaryngol., 364 Med. Surg. II, Irvine, CA 92697)

Speech communication often takes place in noisy and reverberant environments. Unfortunately, cochlear implant (CI) users perform significantly poorer than normal hearing listeners under adverse conditions. We hypothesize that the noise susceptibility of CI users is largely due to the lack of encoding the fine structure cue. Most CI speech processing algorithms extract and encode the slowly varying amplitude modulation in speech (the envelope), while the rapidly varying fine structure is replaced with a fixed rate carrier. We have developed a new speech processing algorithm, called Frequency-Amplitude-Modulation-Encoding (FAME), which codes both amplitude and frequency modulations. Normal hearing listeners were presented with IEEE sentences in the presence of an interfering voice as a function of the number of noise bands (information channels) and signal-to-noise level. Sentence recognition with a competing talker was generally improved with the addition of fine structure, and by as much as 60% with 4 frequency bands (i.e., from 20%–80% at a 20 dB signal-to-noise level). This result suggests that the fine structure is important for speech recognition under realistic situations and should be encoded in future cochlear implants.

**4aSC3. Fricative spectral moments and the perception of anticipatory coarticulation.** William F. Katz, Sneha V. Bharadwaj, Monica P. Stettler, and Peter F. Assmann (School of Human Development and the Callier Ctr. for Commun. Disord., Univ. of Texas, Dallas, P.O. Box 830688, Richardson, TX 75083, wkatz@utdallas.edu)

Spectral moments (mean, variance, skewness, kurtosis) have been used to classify fricative place of articulation in English and other languages. For example, studies have shown the first spectral moment (centroid) is consistently lower in /ʃ/ than in /s/. It has also been hypothesized that fricative spectral moments contribute to listeners' judgments of vowel identity in fricative-vowel (FV) syllables, providing acoustic cues for the perception of anticipatory coarticulation. To study the relationship between fricative spectral moments and traditional fricative spectral attributes as cues to the perception of coarticulation, we obtained a database of FV syllables (/si/, /su/, /ʃi/, /ʃu/) produced by adults and children (ages 5 and 7). Acoustic analyses examined spectral and temporal properties of the signal, while a perceptual experiment determined how adult listeners use gated "slices" ( $\frac{1}{2}$  fricative,  $\frac{3}{4}$  fricative, full fricative, fricative plus  $\frac{1}{2}$  vowel) to identify the fricative and upcoming vowel in the FV syllables. The identification of full syllables showed effects of gate length and talker group, while the vowel identification data revealed little difference as a function of either variable. Logistic regression is used to test the independence of fricative and vowel cues in syllable perception, and to model developmental trends.

**4aSC4. English-learning infants use juncture cues to segment speech.** Elizabeth K. Johnson (Dept. of Psych., Johns Hopkins Univ., Baltimore, MD 21218)

Seven-month-olds perceive recurring sequences of strong–weak (SW) syllables as words, presumably due to their heavy reliance on the metrical segmentation strategy for English. For example, 7-month-olds will segment "taris" from a passage containing repeated tokens of "guitar is." However, by 10 months, infants no longer make this mistake [Jusczyk *et*

*al.*, Cognit. Psychol. **39**, 630–645 (1999)]. There are many possible explanations for this result. Some have argued that 10-month-olds are sensitive to the subtle juncture cues suggesting a word boundary between syllables such as "tar" and "is." However, others have suggested that 10-month-olds have adopted a lexical segmentation strategy. In other words, infants know common function and auxiliaries such as "is." Therefore, they know that "taris" cannot be a word. The head-turn preference procedure was used to investigate whether English-learning infants would segment "toga" from "toe galore" just as readily as they would segment "toga" from "toga lore." Seven-month-olds segmented "toga" equally well in both cases, whereas 12-month-olds failed to segment "toga" from "toe galore." These results suggest that knowledge of frequently occurring function and auxiliaries cannot explain Jusczyk *et al.*'s results. Future work will investigate the information infants might use to distinguish "toe galore" from "toga lore."

**4aSC5. Perception of different speech registers in noise.** Jean Andruski and Eileen Bessega (Wayne State Univ., 581 Manoogian Hall, Detroit, MI 48202)

Infant-directed speech and adult-directed clear speech both exhibit acoustic characteristics (other than amplitude differences) which should make them easier to perceive against a background of noise than adult-directed conversational speech. For example, in adult-directed clear speech, vowels are typically acoustically more distinct. Infant-directed speech shows similar differences in vowel quality, and in addition typically shows larger pitch contours. This experiment examines listeners' perception of these three speech registers (adult-directed conversational speech, adult-directed clear speech, and infant-directed speech) in noise. Effect of the different speech registers will be compared for easy words (frequent words with few frequent lexical neighbors) and hard words (infrequent words with numerous, frequent neighbors).

**4aSC6. Auditory and visual clear speech effects measured during a simulated conversational interaction.** Jean-Pierre Gagne, Monique Charest, Ariane Laplante-Levesque, and Olivia Guilbert (Orthophonie et Audiologie, Univ. of Montreal, C.P. 6128, Succursale Centreville, Montreal, QC H3C 3J7, Canada)

Iterations of sentences were recorded audio-visually from talkers while they participated in a speech-tracking task. Six female talkers produced iterations of conversational and clear speech under two different experimental conditions: (a) while the talker was informed that only her visual–speech cues would be transmitted to the interlocutor and (b) while she was informed that only her auditory–speech cues would be transmitted to the interlocutor. In reality, both her auditory– and her visual–speech cues were recorded under each experimental condition. Target sentences were extracted for the recordings, edited, and presented in a random order to a group of 48 subjects. The subjects completed a speech-recognition task under two perceptual modalities: auditory-only and visual-only. The subjects' mean speech-recognition scores were used to determine the speech intelligibility scores of individual talkers for each experimental condition. The results failed to reveal any differences between the speech intelligibility scores obtained while a talker intended to produce iterations of visual-clear speech and those obtained while she intended to produce iterations of auditory-clear speech. Hence, the findings failed to demonstrate that talkers modify their articulation patterns in order to compensate for the perceptual modality under which the interlocutor receives the speech information. [Work supported by a NSERC grant awarded to J-PG.]

**4aSC7. Visual enhancement for consonants, words, and sentences in normal-hearing young and older adults.** Mitchell S. Sommers (Dept. of Psych., Campus Box 1125, Washington Univ., St. Louis, MO 63130), Nancy Tye-Murray, and Brent Spehar (Central Inst. for the Deaf, St. Louis, MO 63110)

Visual enhancement in speech perception refers to the benefit obtained from seeing and hearing a talker, compared with listening alone. Previous studies comparing visual enhancement in younger (under age 25) and older (over age 65) adults have been confounded by age-related hearing loss. Therefore, the present study was designed to compare visual enhancement for consonants, words, and sentences in normal-hearing older and younger adults. All participants had clinically normal audiograms for octave frequencies from 250 Hz to 4 kHz. Consonant identification was tested using the Iowa Consonant Confusion Test. Word identification was measured using the Children's Auditory-Visual Enhancement Test and sentence identification was measured using the Iowa Sentence Test. All testing was conducted in the presence of a multitalker background babble and baseline differences in auditory-only performance were minimized by setting signal-to-babble ratios individually to produce approximately 50% correct identification in the auditory-only conditions. Visual enhancement, computed as the relative benefit obtained from adding visual speech information to the auditory signal, was significantly higher for younger than for older adults for consonants and words, but not for sentences. The results are discussed as supporting an age-related decline in intermodal sensory integration.

**4aSC8. Examination of the relevance of cochlear dead regions to hearing aid fittings.** Paul D. Dybala and Linda M. Thibodeau (Adv. Hearing Res. Ctr., Univ. of Texas at Dallas, 1966 Inwood Rd., Dallas, TX 75235, dybala@utdallas.edu)

Vickers *et al.* [J. Acoust. Soc. Am. **110**, 1164–1175 (2001)] suggested that identification of Cochlear Dead Regions (CoDR) may provide information relative to hearing aid fittings. The relationship between CoDR and hearing aid fittings was explored in the current study two ways: (1) by reviewing the hearing loss characteristics of Vickers *et al.* (2001) subjects with CoDR using the Articulation Index, and (2) examining the frequency of CoDR in a clinical population. Using amplified, low-pass filtered nonsense syllables Vickers *et al.* (2001) found a relationship between the edge of the identified high-frequency CoDR and the filter setting that provided optimal syllable recognition. Because of the severity of the high-frequency hearing loss for the CoDR group the Articulation Index results from the present study suggested that these listeners may have not benefited from high-frequency amplification because of inaudibility. The frequency of CoDR was determined in a clinical population using the Threshold Equalization Noise (TEN) test described in Moore *et al.* [Br. J. Audiol. **34**, 205–224 (2000)]. Results suggested that CoDR are not encountered as frequently as implied in the literature. These findings have important implications on how CoDR testing relates to hearing aid fittings.

**4aSC9. Application of Plomp's model to recognition of the Hearing in Noise Test sentences.** Linda M. Thibodeau and Jack M. Scott III (Adv. Hearing Res. Ctr., Univ. of Texas at Dallas, 1966 Inwood Rd., Dallas, TX 75235, thib@utdallas.edu)

Sensitive measures of speech recognition are needed for adequate characterization of the effects of various types and degrees of hearing loss. The Hearing in Noise Test (HINT), by Nilsson *et al.* [J. Acoust. Soc. Am. **95**, 1085–1099 (1994)], was developed to quantify the benefits of binaural amplification by measuring a reception threshold for sentences (RTS) in the soundfield in quiet and various noise arrangements. The purpose of the present study was to expand the use of the HINT as a tool to characterize

effects of hearing loss in monotic listening conditions. Modeled after Plomp's [J. Acoust. Soc. Am. **63**, 533–549 (1978)] adaptive, speech-recognition paradigm with Dutch sentences, the HINT sentences were presented in quiet and in increasing noise levels via insert receivers to listeners with normal and mild-to-severe hearing impairments. Following Plomp's analysis, two variables were derived for each listener with hearing loss from the function relating RTS to noise level: *A* and *D*, which represent attenuation and distortion effects, respectively. As expected, the pure-tone average was more strongly correlated with *A* than with *D*. Derivation of these two variables was also explored using an abbreviated procedure of only two RTS values, one in quiet and one in noise.

**4aSC10. Effects on performance of partial misalignments of spectral information in acoustic simulations of cochlear implants.** Anthony Spahr, Michael Dorman (Arizona Biomed. Inst., Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ 85287-0102), and Philipos Loizou (Univ. of Texas, Dallas, Richardson, TX 75083-0688)

Objective: To assess the effects on speech understanding of frequency misalignments in channel outputs for an acoustic model of a cochlear implant. Design: Consonants, vowels, and sentences were processed through a four-channel sine-wave simulation of a cochlear implant. In the five experimental conditions, channels 1 and 3 were always output at the correct frequency while channels 2 and 4 were output at frequencies varying from the correct frequency to frequencies 25%, 50%, and 75% lower than appropriate. In a fifth condition (2-of-4 condition), channels 2 and 4 were turned off. Results: Consonant recognition was reduced significantly with a 50% shift in channels 2 and 4. Vowel and sentence recognition were reduced significantly with a 75% shift in channels 2 and 4. For all material performance in the 75% shift condition was better than in the 2-of-4 condition. Conclusions: The perceiving system that underlies speech recognition is relatively tolerant of misplaced frequency information if other information is presented to the correct frequency location. This flexibility may account for some of the success of cochlear implants. [Research supported by a grant from NIDCD (No. RO1 00654-12) to the second author.]

**4aSC11. The effects of completely-in-the-canal hearing aids, active noise reduction headsets, and ambient noise on speech intelligibility.** John Hall, Mark Ericson (Air Force Res. Lab., Wright-Patterson AFB, OH 45433), and Pamela Mishler (Dept. of Veterans Affairs, Dayton, OH)

Many people with hearing loss need to communicate and work in high noise environments. Hearing aids and headsets are often worn to protect the ear from further damage and facilitate communication. The purpose of this study was to determine the effects of the active noise reduction (ANR) headsets and Completely-in-the-Canal (CIC) hearing aids on speech intelligibility in ambient noise. Six male listeners with noise-induced hearing loss were each fitted with two digital CIC hearing aids. The listeners wore the hearing aids successfully for one year. The speech stimuli were presented over the ANR headphones in quiet and in 95 dB and 105 dB SPL of ambient pink noise. The independent variables were the active noise reduction, hearing aids, and noise level. All MRT stimuli were presented at the most comfortable loudness level. The dependent variable was speech intelligibility as measured with the Modified Rhyme Test (MRT). A main effect was discovered for noise, however no effect was found for active noise reduction or hearing aids. Although no overall effect was found with ANR or hearing aids on speech intelligibility there were differences across phoneme identifications and initial and final consonants.



**4aSC12. Perception of Mandarin Chinese tones by Mandarin versus French listeners.** Pierre A. Hallé (CNRS-Paris V, 71 Av. Edouard Vaillant, 92774 Boulogne-Billancourt, France, halle@psycho.univ-paris5.fr), Catherine T. Best (Wesleyan Univ., Middletown, CT 06459 and Haskins Labs., New Haven, CT 06511), and Yueh-Chin Chang (Tsing-Hua Univ., Taiwan)

Few reports exist on perception of lexical tones by native listeners (e.g., Mandarin Chinese) versus non-native listeners whose language lacks tone contrasts. In those few, the non-native language was English, in which pitch differences signal stress alternations. No studies have examined perception of minimal tone contrasts, of central concern to theories of nonnative speech perception [e.g., PAM: Best (1995)]. We examined categorical perception of Mandarin tone contrasts, in order to detect fine-grained performance differences [see Hall *et al.* (1999)] between listeners of Mandarin and of French, a nonstress language. Three tone continua were derived from natural Mandarin target utterances within carrier sentences. We interpolated on both *F0* and intensity contours. First, Mandarin listeners identified the tone of target syllables within carrier sentences. Next, both French and Mandarin listeners completed A×B identification and discrimination tests on isolated syllables. Mandarin, but not French listener perceived tones quasicategorically. French listeners showed substantial sensitivity to tone contour differences, although to a lesser extent than Mandarin listeners. Thus, despite the lack of linguistically relevant pitch contrasts in their language, French listeners are not “deaf” to tonal variations. They simply fail to perceive tones along the lines of a well-defined and finite set of linguistic categories.

**4aSC13. Native-language influence on phonetic perception in Dutch–English bilinguals.** Michele Mondini (Northeastern Univ., Boston, MA 02115), Petra M. van Alphen (Max Planck Inst. for Psycholinguist., Nijmegen, The Netherlands), and Joanne L. Miller (Northeastern Univ., Boston, MA 02115)

We examined how native-language experience influences processing a second language, focusing on how native Dutch listeners who learned English as a second language perceive the English voiceless consonant /p/. Previous research [J. E. Flege and W. Eefting, *Speech Commun.* **6**, 185–202 (1987)] shows that the voiced–voiceless boundary for an (English-based) voice-onset-time (VOT) series is located at a shorter VOT for such bilingual listeners than for native English listeners, consistent with the fact that voiceless stops are produced with shorter VOTs in Dutch than in English. We asked whether such bilinguals also differ from native English listeners in which stimuli throughout the series are perceived as reasonable exemplars of /p/. Native English listeners and native Dutch listeners were tested on a three-choice identification task with an (English-based) extended VOT series that ranged from /ba/ to /pa/ to an “unnatural” exaggerated /pa/, labeled \*/pa/. Both the /b-/p/ and /p-\*/p/ boundaries were located at shorter VOTs for the native Dutch than the native English listeners, indicating that Dutch native-language experience influenced the entire range of VOTs perceived as reasonable exemplars of the /p/ category. Thus native-language experience has a comprehensive influence on the mapping from acoustic signal to phonetic category. [Work supported by NIH/NIDCD.]

**4aSC14. An evaluation of the semantic categorization task in Japanese word recognition.** Kiyoko Yoneyama (Nat. Inst. for Japanese Lang., 3-9-14, Nishigaoka, Kita-ku, Tokyo 115-8620, Japan, yoneyama@kokken.go.jp)

In this paper an evaluation of the semantic categorization task in Japanese word recognition is reported. Data used in this experiment were taken from Yoneyama (2002) in which fillers were the words for gyesch responses for semantic categories. In this experiment, participants had to

deal with 28 multiple semantic categories. Two findings were concluded. First, although participants generally showed a good performance, the degree of task difficulty among 28 semantic categories was not equal. The average percent correct for gyesch filler-word responses by participants was 95%. The mean values of the words responded to correctly within each category and the number of words correctly classified by at least 93% of the listeners both indicate such tendencies. Second, although the degree of task difficulty among semantic categories within the experiment varied, the selection of semantic categories by itself seemed to be reasonable. The lowest number of words correctly classified by at least 93% of the participants was 17 out of 25 filler words for 4 categories, suggesting that at least 17 of the filler words were considered by general undergraduate students to be good instances of these semantic categories. A generalized methodology to select words in semantic categories will be also discussed. [Work supported by NIH (PI: Keith Johnson).]

**4aSC15. Identification of English /r/ and /l/ in white noise by native and non-native listeners.** Kazuo Ueda, Noriko Yamasaki (Unit of Auditory and Visual Commun. Sci., Kyushu Inst. of Design, 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, ueda@kyushu-id.ac.jp), and Ryo Komaki (ATR Human Information Sci. Labs., Keihanna Science City, Kyoto 619-0288, Japan)

Fifty-three English word pairs, minimally contrasting in /r-/l/, were identified by 14 native American English (AE) listeners and 42 native Japanese (J) listeners, under systematically varied signal-to-noise ratios (SNR) which ranged from no-noise to 21 dB. Word identification for filler contrasts (FL: /d-/k/, /m-/n/, etc.) was also tested under the same SNR conditions. Results show that: (a) identification accuracy by the AE listeners was nearly 100% in the no-noise condition compared to about 70% with a SNR of 21 dB; (b) the J listeners showed similar identification accuracy to the AE listeners for the FL words, whereas they performed much poorer on the /r-/l/ contrasted words (70% accuracy without noise and 55% accuracy with a SNR of 21 dB); and (c) /r-/l/ identification accuracy was affected by an interaction between the consonant position in a word and the listeners’ native languages. The role of perceptual assimilation in the interaction will be discussed. [Work supported by TAO and JSPS, Japan.]

**4aSC16. Influence which the sound as a noise has on maintenance of phoneme information.** Takashi Yagyu (Dept. of Psych., Univ. of Tokyo, 7-3-1 Hongo, Bukyo-ku, Tokyo 113-0033, Japan, yagyu@l.u-tokyo.ac.jp)

This research is investigated from the viewpoint of cross-language processing about the influence which the noise generated by reading aloud has to the maintenance results of phoneme information. If a sentence is read aloud simultaneously when holding phoneme information, the quantity of the phoneme information which can be held and set will decrease. This phenomenon can check the rehearsal of the phoneme information which the noise produced by reading a sentence aloud holds, and it is also possible that maintenance results fall as a result. Then, in this research, the cognitive psychological experiment was conducted using the dual-task method of reading aloud of a sentence, and memory of a word (phoneme information). Two conditions of a native language and a foreign language were set up as a noise generated by reading aloud, and it examined to what noise memory of phoneme information would decline from the point of the individual difference of phonological short-term memory (phonological STM) capacity. Consequently, the more phonological STM capacity was large, the more it was suggested that the stoutness of the phoneme memory to a similar noise increases.



**4aSC17. Auditory-visual integration: A comparison of neuropathy and developmental dyslexia.** Joshua C. Ramirez and Virginia Mann (Dept. of Cognit. Sci., Univ. of California, Irvine, Irvine, CA 92612, ramirezj@uci.edu)

Auditory neuropathy and developmental dyslexia have been previously associated with inferior perception of acoustic speech in noise. To probe the underlying mechanisms of these deficits, an audio-visual integration experiment was conducted to compare the speech perception abilities of reading-impaired subjects and auditory neuropathy patients (relative to controls). Natural acoustic speech stimuli that were white-noise masked at various intensities and accompanied by a video of the talker were presented to subjects. Neuropathy patients were expected to benefit from the pairing of visually presented articulatory cues and auditory CV stimuli, owing to the fact that their speech perception impairment reflects a peripheral auditory disorder. To the extent that developmental dyslexia reflects a general impairment of speech processing rather than a peripheral disorder of audition, it was hypothesized that reading-impaired subjects would not benefit from an introduction of visual cues. Current experimental data yielded a significant correlation between reading ability and dependence upon visual cues. Furthermore, under conditions where articulatory cues are matched with heavily masked CV stimuli, neuropathy patients and controls made more use of the visual cues than reading-impaired subjects. This may suggest that some forms of reading impairment stem from deficits of general speech processing.

**4aSC18. Audio-visual integration of speech with time-varying sine wave speech replicas.** Jyrki Tuomainen, Tobias Andersen, Kaisa Tiippana, and Mikko Sams (Cognit. Sci. and Technol., Lab. of Computational Eng., P.O. Box 9203, Helsinki Univ. of Technol., Finland)

We tested whether listener's knowledge about the nature of the auditory stimuli had an effect on audio-visual (AV) integration of speech. First, subjects were taught to categorize two sine-wave (sw) replicas of the real speech tokens /omso/ and /onso/ into two arbitrary nonspeech categories without knowledge of the speech-like nature of the sounds. A test with congruent and incongruent AV-stimulus condition (together with auditory-only presentations of the sw stimuli) demonstrated no AV integration, but instead close to perfect categorization of stimuli in the two arbitrary categories according to the auditory presentation channel. Then, the same subjects (of which most were still under the impression that the sw-stimuli were nonspeech sounds) were taught to categorize the sw stimuli as /omso/ and /onso/, and again tested with the same AV stimuli as used in the nonspeech sw condition. This time, subjects showed highly reliable AV integration similar to integration obtained with real speech stimuli in a separate test. We suggest that AV integration only occurs when subject are in a so-called "speech mode."

**4aSC19. Effects of 3-D projection on audiovisual speech perception.** D. H. Whalen (Haskins Labs, 270 Crown St., New Haven, CT 06511), Richard Gans (L.I.P.S., Woodbridge, CT 06525), Carol A. Fowler, and Julie M. Brown (Haskins Labs, New Haven, CT 06511)

Visual information about speech influences speech perception, leading to better perception in noise and to illusions such as the McGurk effect. Here, the question was addressed of whether visual influences would be greater with a three-dimensional visual speaker [the patented Life Imaging Projection System (L.I.P.S.)] than the two-dimensional one. Perception in noise was tested with 54 monosyllabic English words of moderate frequency. The McGurk effect was tested with acoustic words having bilabial onset consonants and visual nonwords with alveolar onsets. Visual information improved perception in noise, as found previously; the 3-D version (a videotape projected onto a life cast of the speaker's head) elicited better

performance than the 2-D. However, the 3-D version elicited fewer McGurk responses. Since the presence of noise increases visual fixation on the lips [Vatikiotis-Bateson *et al.*, *Percept. Psychophys.* **60**, 926–940 (1998)], it is possible that more direct information about mouth shape was available in that condition. Further, the 3-D version may have been more likely than the 2-D to engage a direct eye gaze, perhaps reducing the McGurk effect. A test is planned in which noise is added to McGurk stimuli; this may restore the advantage for the 3-D information. [Work supported by NIH grant HD-01994.]

**4aSC20. Multiple talker effects in auditory-visual speech perception among older adults.** Kathleen Cienkowski (Dept. of COMS, Univ. of Connecticut, 850 Bolton Rd., Storrs, CT 06269)

Talker variability effects have been shown for speech stimuli presented auditorily to young adult and older adult participants. Recent investigations suggest that similar variability is seen among young adults for auditory-visual congruent and discrepant stimuli. Older adults are reported to be at a disadvantage for processing information spoken by multiple talkers. Some investigators have suggested that perceptually demanding conditions, such as listening in noise may be more difficult for older listeners. In the current investigation, talker variability effects for CV syllables were examined in multiple modalities visual-only, auditory-only, and auditory-visual for older listeners. Auditory-visual presentations were of two types: congruent [auditory and visual were of the same token—/visual bi and auditory bi/] or disparate [/visual gi and auditory bi/]. The disparate condition has elicited the McGurk effect, in which listeners may report a fused response that is neither the auditory or the visual component of the stimulus. The talkers were 11 individuals with known equivalent auditory intelligibility. Consistent with previous work, individual older listeners varied considerably in their overall perceptions, from complete fusion for all talkers to almost no fusion for any talker. Further, older listeners demonstrated poorer performance in comparison to younger adults in the multitalker condition.

**4aSC21. Audio-visual integration in listeners with normal hearing and hearing aid users.** Nicole L. Marrone (Dept. of Commun. Disord., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215), Ellen S. Stockmann, Frank H. Guenther, Jennell C. Vick, Joseph S. Perkell, and Harlan Lane (MIT, Cambridge, MA)

In listeners with normal hearing, the sight of a speaker's face articulating a syllable can influence the auditory percept, most observably when the auditory and visual stimuli are different from one another. This study investigates differences in audio-visual (AV) integration ("the McGurk effect") between adults with hearing loss who wear hearing aids (HA) and their normal-hearing (NH) counterparts. The following hypothesis is being tested: HA users will rely more on visual input and thus be biased more toward the visual stimulus in the mismatch condition. Audio-visual stimuli from three speakers are presented, pairing the consonants /b/, /d/, and /g/ with the vowels /a/, /i/, and /u/, in three conditions (auditory-only, visual-only, and AV) to the two subject groups, NH and HA. Participants label each stimulus according to the consonant perceived. Responses are coded into four categories: fusion, combination, auditory, or visual. Data analysis examines the relative strength of visual influences in the two groups. Pilot data show fusion and visual bias in an HA user. Further results will be presented. [Work supported by NIH.]

**4aSC22. Intelligibility testing of dysarthria in native Spanish speaking adults.** Michael Fraas (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, P.O. Box 670394, Cincinnati, OH 45267-0394)

Intelligibility testing in dysarthria has attempted to assess, not only the severity of the motor speech deficit but also the nature of the disorder. Kent, Weismer, Kent, & Rosenbek (1989) developed a single word test of intelligibility consisting of acoustic/phonetic contrasts sensitive to both dysarthric impairment and its contribution to speech intelligibility. Although effective for assessing dysarthria in neurodegenerative diagnoses such as Parkinson's disease (PD) and amyotrophic lateral sclerosis (ALS), it was constructed to serve an English speaking population. The current investigation will use the Kent *et al.* (1989) study as a model to develop a single word test of intelligibility in native Spanish speakers. Here 30 subjects, 15 with dysarthria due to neurodegenerative disease and 15 normal controls will be used. Subjects will be recorded reading a list of Spanish minimal pair contrasts. In the study we will attempt to describe the speech characteristics associated with dysarthria in Spanish speakers using acoustic-instrumental analysis and comparatively evaluate these characteristics against those of Spanish-speaking normals. Recordings of the responses will be examined to determine the effectiveness of the Spanish paired word test of intelligibility, and its sensitivity to error patterns of Spanish-speaking dysarthrics.

**4aSC23. Representational specificity of lexical form in the perception of spoken words.** Conor T. McLennan, Paul A. Luce (Language Percept. Lab., Univ. at Buffalo, 245 Park Hall, Buffalo, NY 14260, mclennan@acsu.buffalo.edu), and Jan Charles-Luce (Univ. at Buffalo, Buffalo, NY 14214)

In previous research using the long-term repetition-priming paradigm, we demonstrated that flapped intervocalic alveolar stops in American English may be mapped onto underlying representations of /t/ and /d/ during spoken word processing. More specifically, we found that casually articulated, flapped stimuli primed their carefully articulated counterparts (e.g., casually articulated atom primed carefully articulated atom and Adam) and *vice versa*. We argued that form-based or phonological ambiguity (present in our flapped stimuli) is one of the conditions leading to the activation of underlying representations. However, our flapped items were also lexically ambiguous (e.g., our casually articulated atom was ambiguous between Adam and atom). Is the long-term priming effect we observed for flapped and carefully articulated words lexically or sublexically mediated? In the present research we attempt to answer this question by examining form-based ambiguity in the absence of lexical ambiguity. Long-term repetition priming experiments were conducted with nonword stimuli that preserve form-based ambiguity (e.g., oytem, oydem) and flapped stimuli without lexical pairs (e.g., pretty). The results provide further insights into the conditions under which underlying representations are activated during spoken word processing.

**4aSC24. Nasal consonant speech errors: Implications for "similarity" and nasal harmony at a distance.** Rachel Walker, Narine Hacopian, and Mariko Taki (Dept. of Linguist., USC, GFS 301, Los Angeles, CA 90089-1693, rwalker@usc.edu)

It has been observed that nasals interact with "similar" consonants in phonological long distance nasal harmony [Walker, BLS 26, 321–332 (2000)]. In such patterns, nasals cause other consonants in a word to become nasal. Voiced stops and liquids are preferentially affected; in some cases obligatorily homorganic with the nasal. In this study we ask whether a parallel occurs in consonants with greater likelihood to participate in speech errors with nasals; specifically, is preferred interaction of voiced and homorganic stops with nasals also evident in the error pattern of a

language without nasal harmony? Experiments were conducted on English speakers using the SLIPS technique [Baars and Motley, Catalog of Selected Documents in Psychology, 1974]. CVC word pairs were presented for reading. Critical pairs, primed to encourage initial consonant errors, were cued to be spoken. Experiment 1 (35 subjects) investigated whether more errors involve nasals and voiced stops than nasals and voiceless stops (partially replicates Stemberger's 1991 study). Experiment 2 (37 subjects) examined whether more errors involve nasals and homorganic voiced stops than nasals and heterorganic voiced stops. Results indicate that consonants participating in more errors with nasals parallel those preferentially affected in long distance nasal harmony. [Work supported by a Zumberge grant.]

**4aSC25. Accessing coarticulatory information.** Suzanne Curtin (Dept. of Linguist., 2806 Cathedral of Learning, Univ. of Pittsburgh, Pittsburgh, PA 15260, scurtin@pitt.edu), Neelam Ladhar, and Janet Werker (Univ. of British Columbia, Vancouver, BC V6T 1Z4, Canada)

The speech signal is comprised of coarticulatory cues. Here, we explore whether adults' access to coarticulatory information depends upon the particular language task. Specifically, we tested adults' sensitivity to coarticulation when segmenting new sequences versus remembering presegmented items. Forty undergraduates were familiarized to either a continuous string of appropriately coarticulated nonsense CV syllables where every third syllable was stressed to facilitate segmentation, or to the same CV syllables presegmented into trisyllabic units without any stress contour. During the test phase, adults were presented with familiar and novel sequences appropriately coarticulated, and to both sequence types with inappropriate coarticulation. Subjects rated the familiarity of items using a seven point scale. Both groups rated familiar test sequences as more familiar than novel sequences. However, only the group presented with presegmented items demonstrated sensitivity to coarticulation. These subjects, compared to the segmentation group, significantly preferred the coarticulated items ( $p < 0.05$ ). These results suggest that access to information depends on the task. In the case of word segmentation, adults pay attention to the most useful properties (in this case, stress). Whereas adults in the presegmented group were able to pick up all the details of the sequences since their task only required word recognition.

**4aSC26. The perception of gated Dutch diphones.** James M. McQueen, Roel Smits, Anne Cutler (Max Planck Inst. for Psycholinguist., Postbus 310, 6500 AH Nijmegen, The Netherlands, james.mcqueen@mpi.nl), and Natasha Warner (Univ. of Arizona, Tucson, AZ 85721-0028)

The results of a large-scale speech perception study are reported. Eighteen Dutch listeners identified gated fragments of 1179 Dutch diphones. Diphones were presented six times (for some diphones beginning with voiceless stops, four times) to each listener, in fragments ranging from the shortest gate (the first sixth of the diphone), through to the longest gate (the complete diphone). Order of presentation of all gates of all diphones was fully randomized. Listeners were asked to identify the complete diphone on every trial. The results showed that listeners based their decisions on the acoustic information available in the stimuli, not on higher-order factors such as phoneme occurrence frequencies or transitional probabilities. Perceptual confusions reflected the temporal evolution of phoneme similarities (for example, in their shorter gates, long vowels were confused with their short counterparts). This database provides detailed information about the perceptual hypotheses which listeners entertain about phoneme sequences, as those sequences unfold over time. It can also be used to generate predictions about the temporal pattern of activation of spoken words. Such data are necessary for the evaluation and further

development of current models of spoken word perception, in which fine-grained acoustic information is mapped continuously onto the mental lexicon.

**4aSC27. Temporal integration of acoustic cues in fricative perception.** Santiago Fernández and Sergio Feijóo (Dept. de Física Aplicada, Fac. de Física, Univ. de Santiago, 15782 Santiago de Compostela, Spain)

An important issue in speech perception is to determine how the components of a syllable interact to enhance perception of both consonant and vowel. To date the mechanism underlying that integration has not yet been discovered. Different approaches to the temporal integration between fricative and vowel in a set of natural syllables were explored. Two hypotheses were considered. (a) The two segments are evaluated separately and then combined into a single percept; (b) both cues are evaluated jointly. To test those hypotheses several computational models were considered. If the F and V segments are evaluated separately, two statistical functions are available: An "OR" function corresponding to the perceptual hypothesis predicting that only one of the segments determines the identity of the fricative; an "AND" function corresponding to the perceptual hypothesis predicting the use of both cues. The hypothesis of the joint evaluation of both cues was tested using the whole FV segment. Their performances were compared with the perceptual performance of a group of listeners in a fricative identification task. Although the joint evaluation model was superior to the other models, it was unable to extract the same benefits as listeners from the fricative-vowel interaction. [Work supported by Xunta de Galicia.]

**4aSC28. Learning-induced neural plasticity associated with acquisition of a difficult second-language phonetic contrast.** Daniel Callan, Rieko Kubo (Human Information Sci. Labs., ATR Intl., Japan), Akiko Callan, and Shinobu Masaki (Brain Activity Imaging Ctr., ATR Intl., Japan)

Adult native Japanese speakers have difficulty perceiving the English /r/ phonetic contrast even after years of exposure. However, after extensive perceptual identification training long lasting improvement in identification performance that generalizes to novel stimuli can be attained. In this fMRI study we investigate localized changes in brain activity associated with one-month of extensive feedback-based perceptual identification training by native Japanese speakers learning the English /r/ phonetic contrast. Before and after training separate functional brain imaging sessions were conducted for identification of the English /r/ contrast (difficult for Japanese speakers), /bg/ contrast (easy), and /bv/ contrast (difficult). Neural plasticity, denoted by exclusive enhancement in brain activity for the /r/ contrast (not present for the /bv/ and /bg/ conditions), does not only occur in brain regions involved with acoustic-phonetic processing (superior temporal areas, supramarginal gyrus, planum temporale) but also in additional bilateral cortical (Broca's area, premotor cortex, orosensory cortex) and subcortical regions (cerebellum, basal ganglia, substantia nigra) involved with speech production as well as with formation of perceptual-motor mappings. The results support the hypothesis that learning of an auditory-articulatory mapping improves identification performance by allowing a perception to be made in reference to potential action. [Work supported by TAO, Japan.]

THURSDAY MORNING, 5 DECEMBER 2002

GRAND CORAL 1, 7:50 A.M. TO 12:00 NOON

### Session 4aUW

## Underwater Acoustics and Acoustical Oceanography: Littoral Environmental Variability and Its Acoustic Effects II

Peter H. Dahl, Cochair

*Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, Washington 98105-6698*

James F. Lynch, Cochair

*Woods Hole Oceanographic Institution, 203 Binglew Building, Woods Hole, Massachusetts 02543*

Chair's Introduction—7:50

### Invited Papers

7:55

**4aUW1. Overview of ASIAEX field experiments in the South and East China Seas.** James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543) and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

In the spring of 2001 the Asian Seas International Acoustics Experiment (ASIAEX) was performed in the South and East China Seas. The ASIAEX program originated from the Office of Naval Research's initiative to develop a Sino-American cooperation in the field of ocean acoustics, and expanded to involve Taiwan, Korea, Singapore, and Japan. In the South China Sea, the emphasis was on lower frequency (50–600 Hz) acoustic propagation through complex oceanography along and across the shelfbreak. The oceanographic variability was driven at the mesoscale by the monsoonal wind stress, buoyancy fluxes from the Chinese coast, and by occasional Kuroshio intrusions through the Luzon Strait. Large, nonlinear internal waves also significantly affected the acoustics. In the East China Sea the emphasis was on low- to mid-frequency acoustic interaction with, and reverberation from, the sea floor and sea surface. The experimental site was chosen for flatness, to minimize the influence of bathymetric trends, and emphasize bottom

roughness and subbottom structures, in the measurements. This talk will describe the scope of the ASIAEX experimental program, including the ocean acoustic and environmental characterization of the seafloor, sea surface, and water column. Results from various measurement programs will be described in separate papers.

8:15

**4aUW2. Long range sediment tomography.** James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882)

The development of long range sediment tomography using shot data and its application in various field experiments is presented. Whereas traditional ocean acoustic tomography has mainly relied on linear inverse theory using the travel time of rays in deep water, our technique relies on nonlinear optimization techniques using the frequency-dependent travel time of normal modes in shallow water. Our experience with broadband sediment tomography using explosive sources started with the Barents Sea Polar Front Experiment where sediment properties were extracted using ground wave analysis. The techniques were further refined during the Shelf Break Primer Experiment in the New England Bight where a nonlinear inversion scheme using the Genetic Algorithm was successfully applied. The inversion scheme was based on modal travel time dispersion characteristics, which in shallow water conditions, is highly sensitive to the bottom properties. Advanced signal processing tools for the space-time-frequency processing and hybrid optimization techniques for global search were developed and implemented in parallel. In addition to the individual modal arrival times, emphasis has now been placed on the arrival corresponding to the Airy phase. These inversion techniques are currently being applied to the data from ASIAEX-East China Sea Experiment. Characteristics of this tomography approach such as spatial coverage, resolution, error bars, robustness, and simple instrumentation requirements will be presented. The usefulness of this inversion approach as a rapid environmental assessment tool will be highlighted. [Work supported by ONR.]

8:35

**4aUW3. The effects of the ocean environmental variability on sound propagation and reverberation from ASIAEX (ECS).** Fenghua Li, Renhe Zhang, Zhenglin Li, and Jianjun Liu (Natl. Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing 100080, PROC, lfh@farad.ioa.ac.cn)

During the duration of the Asian Seas International Acoustics East China Sea Experiment, the ocean environment (e.g., wind speed, sea state, internal wave) varied significantly. Propagation and reverberation data from explosive charges recorded by a vertical line array and horizontal line array at different times during the experiment show that the acoustic data have a strong fluctuation. In this paper, the recorded data are analyzed to estimate the relationship between the acoustic signal fluctuation and the environmental variability. A PE code is used to simulate the effect of the recorded ocean environmental variability on the sound signal fluctuation. The experimental data and numerical results indicate the strong dependence between the acoustic fluctuation and ocean environmental variability. A coupled normal-mode-based model is also used to explain some of the acoustic fluctuation. [Work supported by the National Natural Science Foundation of China and ONR.]

8:55

**4aUW4. Environmental measurements and mid-frequency backscatter in shallow water.** Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

To understand bottom backscattering mechanism at the mid-frequency range (nominally 3.5 kHz), direct measurements of bottom roughness and sub-bottom heterogeneity were made along with concurrent backscatter measurements in a shallow water site with 105 m water depth. The backscatter was recorded on a vertical line array with 31 elements with a 3.5 kHz source attached at the bottom of the array. Bottom roughness and sub-bottom heterogeneity were measured using an *in situ* conductivity probe. The roughness measurements cover a one-dimensional profile of approximately 4 m in length with vertical resolution of 4 mm and horizontal resolution of 1.5 to 2.5 cm. Heterogeneity measurements cover a depth of 5 to 10 cm. Ambient noise received by the vertical line array was used to estimate the sound speed, density, and attenuation coefficient of the surficial sediments. A Monte Carlo modeling capability was developed to extensively simulate the backscatter with different environmental inputs. The results of this effort covering all three areas of acoustics, environments, and modeling will be reported.

### Contributed Papers

9:15

**4aUW5. Analysis of time series data in the East China Sea generated from explosive sources.** D. P. Knobles, Thomas W. Yudichak (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029), Peter Cable, Y. Dorfman (BBN Technologies), Peter H. Dahl (Univ. of Washington, Seattle, WA), James H. Miller, Gopu R. Potty (Univ. of Rhode Island, Providence, RI 02803), Renhe Zhang, Zhaohui Peng, Fenghua Li, and Zhenglin Li (Inst. of Ocean Acoust., Beijing, PROC)

Time series data collected on the APL-UW/URI VLA in the East China Sea as part of the Asian Sea International Acoustics Experiment are analyzed for the information they contain on the characteristics of the seabed. Sound generated by explosive sources deployed by the IOA propa-

gates in a shallow water wave guide under downward refracting conditions, making the received field at the VLA sensitive to the structure of the seabed. A broadband normal mode approach is used to model the measured time series in the 10–500 Hz band. The complex multipath arrival pattern as a function of source–receiver range and source depth allows one to infer certain characteristics of the seabed without the aid of an inversion approach. A finer specification of the seabed, including the determination of the statistics of the geoacoustic parameters is achieved by a simulated annealing inversion methodology. The sensitivity of the acoustic propagation to the elastic properties of the seabed is also investigated along with the nature of the attenuation. The estimated properties of the seabed are compared with independent geophysical measurements and those that can be obtained from reverberation data. [Work supported by ONR.]



9:30

**4aUW6. Analysis of sound propagation data taken in the East China Sea.** Zhaohui Peng, Ji-xun Zhou (Inst. of Acoust., Chinese Acad. of Sci., Beijing 100080, PROC and School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, zpeng@sununo.me.gatech.edu), Renhe Zhang (Inst. of Acoust., Chinese Acad. of Sci., Beijing 100080, PROC), and Peter H. Dahl (Univ. of Washington, Seattle, WA 98105)

As a part of the Asian Seas International Acoustic Experiment (ASIAEX2001), sound propagation data from wideband (explosive) sources were recorded in the East China Sea by using a 32-element suspended array. The propagation was measured as a function of range in two perpendicular tracks (one up to 60 km) and as a function of azimuth for a fixed range of 30 km. Supporting environmental data, obtained from a 17-element thermister chain, XBT, and CTD, showed very complex variation in the water column. In this paper, transmission loss (TL) as a function of range, frequency and azimuth is briefly introduced. Seabottom acoustic parameters such as density, velocity, and attenuation are inverted from the sound propagation data. Then, these parameters plus internal wave data are used as inputs to PE and normal-mode codes qualitatively to explain observed strong fluctuations in sound propagation. The inversion techniques used for estimating seabottom parameters, including spatial mode filtering and dispersion analysis, are also discussed. [Work supported by the National Natural Science Foundation of China and ONR.]

9:45

**4aUW7. A comparison of East China Sea low frequency bottom scattering strength determinations.** P. G. Cable, R. Gibson, (BBN Technologies, 1300 N. 17th St., Arlington, VA 22209), Y. Dorfman (BBN Technologies, Cambridge, MA 02138) D. P. Knobles, T. W. Yudichak Univ. of Texas, Austin, TX 78713-8029), R. Zhang, Z. Peng, F. Li, Z. Li (Chinese Acad. of Sci., Beijing, PROC), P. H. Dahl (Univ of Washington, Seattle, WA 98105) J. H. Miller, and G. R. Potty (Univ. of Rhode Island, Kingston, RI 02881)

Low frequency shallow water bottom scattering strength determinations conventionally involve accounting for the two-way transmission from source to scattering region to receiver and correcting for the size of the contributing bottom scattering area. Experiment uncertainties in transmission loss, bottom homogeneity and isotropy, contributions from volume, and ocean surface backscatter, can all contaminate the accuracy and robustness of such determinations. Bottom scattering strengths were determined in octave bands from 50–800 Hz reverberation data using Institute of Acoustics sources and the receiving array on Shi-Yan III and from transmission data obtained on the APL-UW/URI receiving array on R/V Melville during East China Sea ASIAEX in 2001. Seabed geoacoustic parameters were inferred from the forward data, which were then used to model transmission and extract scattering strength from the reverberation data. The ASIAEX scattering strengths were compared with other measurements of bottom scattering strength derived from previously obtained East China Sea broadband data sets. The separate bottom scattering strength determinations, though obtained at different times and East China Sea sites, and by different methods, are very close. The experimental sensitivity of these measurements will be described and implications regarding the robustness of the results to seasonal variability and geographic change discussed. [Work supported by ONR.]

10:00–10:15 Break

10:15

**4aUW8. Spatial coherence of signals forward scattered from the sea surface in the East China Sea.** Peter H. Dahl, Christian J. Eggen, and Russell D. Light (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

Measurements of sea surface forward scattering, wind speed, and directional wave spectra made in 100 m of water in the East China Sea are discussed. The experiment was part of the Asian Seas International Acoustics Experiment (ASIAEX) conducted in the spring of 2001. Signals were

received at ranges near 500 m on 2 vertical line arrays that were co-located but separated in depth by 25 m. Estimates of the vertical spatial coherence along these arrays as a function of frequency, path geometry, and sea surface environmental conditions are compared with a model for spatial coherence. The model is based on identifying the probability density function that describes vertical angular spread at the receiver position. An alternative approach utilizing the van Cittert–Zernike theorem from statistical optics is shown to give equivalent results. Both approaches require computation of the sea surface bistatic cross section, done here with the small slope approximation. Forward scattering from the sea surface represents an important channel through which sound energy is transmitted, and spatial coherence determines in part the performance of imaging and communication systems that utilize the sea surface bounce path.

10:30

**4aUW9. Time- and space-varying interference patterns of broadband acoustic field sampled by drifting buoys during ENVERSE 97 experiments.** Jean-Pierre Hermand (Royal Netherlands Naval College, Postbus 10000, 1780 CA Den Helder, The Netherlands, jhermand@ulb.ac.be), Serge Scevenels (Université Libre de Bruxelles, B-1050 Brussels, Belgium), and Frans G. J. Absil (Royal Netherlands Naval College, 1780 CA Den Helder, The Netherlands)

During the winter of 1997 SACLANTCEN deployed a fixed controlled sound source, a vertical receive array, and drifting hydrophone buoys in a complex coastal environment on the western Sicilian shelf (ENVERSE 97). The acoustic impulse response of the medium was measured in a broad frequency band as a function of range and azimuth from the source, using repeated, large time-bandwidth-product FM transmissions and DGPS positioning. This paper investigates the combined effects of water column and bottom variability upon the space-frequency distribution of the sound-field intensity. The time and space dependence of extracted features such as patterns of field extrema are analyzed and related to the observed environmental conditions. In particular, perturbations of acoustic field invariants are detected and shown to be well correlated with the range-dependent bottom properties and the time-varying ocean sound speed and current fields. Preliminary modeling (C-SNAP) results obtained from concurrent oceanographic and geophysical ground-truth data (ENVERSE 98) are compared to the acoustic measurements to determine their sensitivity to various environmental parameters. [Work supported by the Royal Netherlands Navy.]

10:45

**4aUW10. Temporal resolutions of time-reversal and passive-phase conjugation for underwater acoustic communications. I. Stationary source and receivers in a dynamic shallow-water environment.** T. C. Yang (Naval Res. Lab., Code 7120, Washington, DC 20375)

Multipath arrivals cause interference between transmitted symbols and hence, symbol errors in underwater acoustic communications. A time-reversal mirror uses the ocean to combine the multipath arrivals and can be used for underwater acoustic communications. Passive-phase conjugation uses the received data of a probe signal to deconvolve the channel transfer function and thereby removes the multipath effects. In this paper, we study the following temporal resolutions of a time-reversal or passive-phase conjugation process as applied to underwater acoustic communications: (1) the time resolution or the pulse width of a backpropagated, time-compressed pulse as compared with the original transmitted pulse; (2) the effectiveness of temporal focusing as measured by the peak-to-sidelobe ratio of the backpropagated or phase-conjugated pulse; both pulse elongation and sidelobe leakages are causes of intersymbol interference and bit errors for communications; (3) the duration of temporal focusing or the temporal coherence time of the underwater acoustic channel; and



(4) the stability of temporal focusing as measured by the phase fluctuations of successive pulses (symbols). BPSK signals collected at sea were used to extract the above four parameters. The bit error rates are modeled with simulated data. [Work supported by ONR.]

11:00

**4aUW11. Temporal resolutions of time-reversal and passive-phase conjugation for underwater acoustic communications. II. The effect of source motion.** T. C. Yang (Naval Res. Lab., Code 7120, Washington, DC 20375)

Multipath arrivals cause interference between transmitted symbols and hence, symbol errors in underwater acoustic communications. A time-reversal mirror uses the ocean to combine the multipath arrivals and can be used for underwater acoustic communications. Passive-phase conjugation uses the received data of a probe signal to deconvolve the channel transfer function and thereby removes the multipath effects. In this paper, we study the following temporal resolutions of a time-reversal or passive-phase conjugation process as applied to underwater acoustic communications: (1) the time resolution or the pulse width of a backpropagated, time-compressed pulse as compared with the original transmitted pulse; (2) the effectiveness of temporal focusing as measured by the peak-to-sidelobe ratio of the backpropagated or phase conjugated pulse; both pulse elongation and sidelobe leakages are causes of intersymbol interference and bit errors for communications; (3) the duration of temporal focusing or the temporal coherence time of the underwater acoustic channel; and (4) the stability of temporal focusing as measured by the phase fluctuations of successive pulses (symbols). We analyze BPSK signals collected at sea from a moving source, extract the above four parameters, and evaluate the bit error rate. [Work supported by ONR.]

11:15

**4aUW12. Direct path fluctuations due to shallow-water variability.** Stephen Karpi, Kevin B. Smith, Steven Ramp (Naval Postgrad. School, Monterey, CA 93943), and Peter H. Dahl (Univ. of Washington, Seattle, WA 98105)

The current interest in enhancing the forecasting capabilities of both active and passive sonar systems employed in littoral regions has greatly escalated. This requires improvements in the general understanding of the influence of shallow-water internal waves on acoustic propagation. This work will contribute to a more fundamental understanding of ocean acoustic propagation and fluctuations in shallow-water regions and examine the influence of shallow-water variability on the relatively short-range waterborne propagation paths. Specifically, internal wave fluctuations will be considered and the influence on the acoustic propagation will be quantified in terms of spatial (vertical) coherence functions. The data to be examined

will be generated numerically based on an acoustic propagation model employing environmental data taken from the East China Sea as part of the ONR-sponsored ASIAEX experiments. The results of this analysis will be compared with the measured data currently being analyzed at the Applied Physics Laboratory at the University of Washington.

11:30

**4aUW13. The measurement of acoustic dispersion in loosely consolidated, saturated sediments using a water-filled impedance tube.** Preston S. Wilson, Eun-Joo Park, Ronald A. Roy, and William M. Carey (Dept. of Aersp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

At frequencies of several kilohertz and below, the measurement of sound propagation in marine sediments is difficult due to the larger wavelengths. Laboratory experiments can be limited by the size of the facility required for propagation studies but are amenable to material property measurements. Impedance tube techniques can be used to measure the complex interfacial properties of small samples over a broad and continuous range of frequencies. From this, frequency-dependent sound speed and attenuation is obtained. Results from compressional wave speed and attenuation measurements made with a laboratory impedance tube using artificial and natural water-saturated sediments will be presented and compared to existing propagation models. Variation of attenuation with frequency will be discussed. [Work supported by the U.S. Navy Office of Naval Research and the Coastal Systems Station.]

11:45

**4aUW14. Study on ambient noise generated from breaking waves simulated by a wave maker in a tank.** Ruey-Chang Wei and Hsiang-Chih Chan (Inst. of Undersea Technol., Natl. Sun Yat-Sen Univ., Kaohsiung City, Taiwan)

This paper studies ambient noise in the surf zone that was simulated by a piston-type wave maker in a tank. The experiment analyzed the bubbles of a breaking wave by using a hydrophone to receive the acoustic signal, and the images of bubbles were recorded by a digital video camera to observe the distribution of the bubbles. The slope of the simulated seabed is 1:5, and the dimensions of the water tank are 35 m × 1 m × 1.2 m. The studied parameters of ambient noise generated by breaking wave bubbles were wave height, period, and water depth. Short-time Fourier transform was applied to obtain the acoustic spectrum of bubbles, MATLAB programs were used to calculate mean sound pressure level, and determine the number of bubbles. Bubbles with resonant frequency from 0.5 to 10 kHz were studied, counted from peaks in the spectrum. The number of bubbles generated by breaking waves could be estimated by the bubbles energy distributions. The sound pressure level of ambient noise was highly related to the wave height and period, with correlation coefficient 0.7.

4a THU. AM

## Meeting of the Standards Committee Plenary Group

to be held jointly with the

### ANSI-Accredited U.S. Technical Advisory Group (TAG) Meeting for: ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics

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

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

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

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

<b>S12</b>	9:45 a.m. to 12:00 noon
<b>S3</b>	1:45 p.m. to 3:15 p.m.
<b>S1</b>	3:30 p.m. to 5:00 p.m.

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees. The ANSI-Accredited U.S. Technical Advisory Group (TAGs) for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, whose membership consists of members of S1 and S3, and other persons not necessarily members of these Committees, will meet during the Standards Plenary meeting. *The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting.* There will be a report on the interface of S1 and S3 activities with those of ISO/TC 43 and IEC/TC 29 including plans for future meetings of ISO/TC 43 and IEC/TC 29.

Members of S2 Mechanical Vibration and Shock (and U.S. TAG for ISO/TC 108 and five of its Subcommittees, SC1, SC2, SC3, SC5, and SC6) are also encouraged to attend the Standards Committee Plenary Group meeting even though the S2 meeting will take place one day earlier, on Wednesday, 4 December 2002, at 9:00 a.m.

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

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

**Meeting of Accredited Standards Committee (ASC) S12 Noise**

to be held jointly with the

**ANSI-Accredited U.S. Technical Advisory Group (TAG) Meetings for: ISO/TC 43/SC 1, Noise**P. D. Schomer, Chair S12, and Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC 1, Noise  
*2117 Robert Drive, Champaign, Illinois 61821*R. D. Hellweg, Vice Chair S12  
*Compaq Computer Corporation, Acoustics Lab, Mechanical Engineering Group, MR01-3/03, 200 Forest Street,  
Marlborough, Massachusetts 01752*H. E. von Gierke, Vice Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC 1, Noise  
*1325 Meadow Lane, Yellow Springs, Ohio 45387*

**Accredited Standards Committee S12 on Noise.** Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. There will be a report on the interface of S12 activities with those of ISO/TC 43/SC 1 Noise, including plans for future meetings of ISO/TC 43/SC 1. The Technical Advisory Group for ISO/TC 43/SC 1 consists of members of S12 and other persons not necessarily members of the Committee. Open discussion of committee reports is encouraged.

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

THURSDAY AFTERNOON, 5 DECEMBER 2002

GRAND CORAL 2, 1:30 TO 2:30 P.M.

**Session 4pAAa****Architectural Acoustics, Musical Acoustics and Physical Acoustics: Ancient Acoustics III  
Sounding Instruments**James P. Cottingham, Cochair  
*Department of Physics, Coe College, Cedar Rapids, Iowa 52402*Roberto Velasquez, Cochair  
*Trujillo 726, Col. Lindavista, C.P. 07300, D. F. Mexico***Invited Papers****1:30****4pAAa1. The Asian free-reed mouth organs.** James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402, jcotting@coe.edu)

Mouth-blown instruments employing a free reed coupled to a pipe resonator have long been known and used throughout East and Southeast Asia. Details of the origin and development of these instruments are not known, but are closely connected with the history and prehistory of a multitude of ethnic groups. These instruments exemplify a pipe-resonator coupling significantly different from the standard wind instruments of European origin. The free reed used is approximately symmetric, often operating on both directions of air flow. In some cases the reed is at or near one end of a closed pipe, but in other examples the reed is mounted in the side of a resonator open at both ends. The instruments can be either multiple pipe instruments with one pipe per note, or a pipe with a single reed and tone holes. A number of experimental studies have been conducted on examples of Asian free-reed instruments, primarily the khaen, bawu, and sheng. These include studies of reed vibration, measurements of sound spectra, and impedance measurements of the pipes. Comparisons have been made between experimental results and theoretical work on the coupling of reed vibration with the pipe resonator.

1:50

**4pAAa2. Ancient celtic horns.** Murray Campbell (Univ. of Edinburgh, Edinburgh EH9 3JZ, UK)

There is considerable evidence from iconographic and documentary sources that musical lip-reed instruments were important in the early celtic communities of Scotland and Ireland. In recent years several studies have been undertaken with the aim of gaining a better understanding of the musical nature of these ancient horns, and of their place in the life and culture of the time. A valuable source of tangible evidence is to be found in the archaeological remains deposited across Scotland and the whole of Ireland. A project is now under way, under the auspices of the Kilmartin House Trust and the general direction of John Purser, which has brought together an international team of musicians, craftsmen, archaeologists, musicologists and physicists with the aim of analyzing ancient musical artifacts, reconstructing some of the original instruments, and analyzing the sounds they produce. This paper describes acoustical studies carried out on a number of recent reconstructions of wooden and bronze instruments, and discusses the role of acoustics in this type of investigation. [Work supported by Sciart and EPSRC.]

2:10

**4pAAa3. Characteristics and speculations on the uses of *Strombus* trumpets found at the ancient Peruvian center Chavín de Huántar.** John Rick (Dept. of Anthropological Sci., Stanford Univ., Bldg. 360, Stanford, CA 94305-2117) and David Lubman (David Lubman & Assoc., Westminster, CA 92683-4514)

A cache of 20 *Strombus* shell trumpets was excavated in 2001 from an underground gallery at Chavín de Huántar, the type site of the Peruvian Early Horizon period (ca. 1200 to 400 B.C.). *Strombus* shell usage stretches from antiquity to present day Peru, with the trumpet function showing remarkable continuity. Soon after their discovery, a dozen of the ancient shells were played to an enthusiastic audience at Lima's National Museum of Archaeology, Anthropology and History. Evidence suggests that *Strombus* trumpets are, and were, used as ritual instruments for the legitimization of political and religious power. Those uses may have developed from earlier practical uses for ritual and communication. This paper describes measured acoustical properties of a *Strombus* horn. *Strombus* blasts are rich in overtones. Source strengths as high as 111 dBA @ 1 M were observed. These high source levels support speculation that shell trumpets could have been used for signaling over great distances. The trumpets can produce strong combination tones, which, accompanied by likely ingestion of local hallucinogenic substances, the use of reflected light, and other sound manipulation, suggests that early leaders in Chavin were using a range of methods to help establish early religious and political authority.

THURSDAY AFTERNOON, 5 DECEMBER 2002

GRAND CORAL 2, 2:45 TO 5:05 P.M.

### Session 4pAAb

## Architectural Acoustics, Musical Acoustics and Physical Acoustics: Ancient Acoustics IV Pre-Columbian Sounding Instruments

Roberto Velasquez, Cochair

*Trujillo 726, Col. Lindavista, C.P. 07300, D. F. Mexico*

James P. Cottingham, Cochair

*Department of Physics, Coe College, Cedar Rapids, Iowa 52402*

### *Invited Papers*

2:45

**4pAAb1. Construction practices in pre-Hispanic flutes.** Susan Rawcliffe (P.O. Box 924, San Pedro, CA 90733)

The ancient cultures of the Americas were separated by thousands of miles and thousand of years. There was a long history of trade over the miles and the years with many shared cultural ideals and artifacts, as evidenced by their musical instruments. Flutes were the most prevalent instruments found throughout the ancient Americas. Some types are unique to the pre-Hispanic world. Although flutes were constructed from a variety of materials, including bone, cane, seed pods, skulls, it is primarily the ceramic ones that survived, and it is ceramic flutes which form the bulk of this writer's work and research. This paper includes musical demonstrations to show how ancient flutemakers could have manipulated timbre during construction. Clay's plasticity enabled the construction of some instruments, and limited the development of others. Pitch jump flutes, certain Veracruzano whistles, and chamberduet flutes and whistles all share the addition of clay flaps or chambers around the aperture, as do hooded pipes. Some instruments exhibit a seemingly cultural predilection for complex tones that are windy, raspy, or animalistic. Simple adjustments of the airduct promote these timbres. Also included will be samples of the original sounds of ancient flutes.

3:05

**4pAAb2. Zapotec whistles.** Gonzalo Sanchez (Virgen del Carmen 158, Col. Virgencitas, Nezahualcoyotl, Edo. Mexico, C.P. 57300, Mexico)

Sonorous artifacts or “pre-Hispanic musical instruments” have attracted the attention of ethnomusicologists, acousticians and archaeologists. The Mexican cultures that have had most studies on the matter are the Maya, Mexica, and Totonaca ones. In the case of the Oaxaca culture, until recently, investigations of this type had not been made. Thus, the task was to look for and to analyze material that belonged to the cultures that inhabited the present Oaxacan territory from a perspective acoustics-organologic to be able to understand what type of sonorities that were used. The first step was to consider the project of investigation within the field of ethnomusicology. In the explorations of the archaeological sites of the Oaxaca Valley it is common to find whistles, which are complete or in fragments, that show certain organologic characteristics that until now have only detected for the Zapotec culture. In this study the morphology, the chronological location, the contexts, the characteristics of the sounds, the symbolism, and the techniques of elaboration of Zapotec whistles are presented. (To be presented in Spanish.)

3:25

**4pAAb3. Yaxchilan’s whistles.** Roberto Velazquez (Virtual Res. Inst. Tlapitzcalzin, Trujillo 726, Col. Lindavista, Mexico D.F., C.P. 07300, Mexico, rvelaz.geo@yahoo.com)

Eight clay whistles found buried in Small Acropolis temples (650–800 BC) of Yaxchilan were analyzed. The study was centered on three dual whistles with body shapes resembling those of frogs. A method developed in previous studies and evidence from several disciplines were applied. Experimental replicas were made to find possible ancient ways of construction and to test hypotheses. The Helmholtz equation for globular resonators was calculated, sound signals were analyzed with spectrograms, and radiated acoustic power in different modes was estimated (0.0005–0.003W). The power level indicates that whistles’ sounds could not be heard well if they were played along with louder Mayan aerophones like those of Bonampak’s mural band nor in noisy celebrations in big plazas. Clay frogs were adequate to be used in the Small Acropolis, in the Labyrinth and by big groups, surely related to religion. The whistles can sing like natural frogs and produce beats, and they might have been used to produce a chorus in ceremonies to the god Chaac to call for rain, in H-men rites, or in the infraworld where they were discovered.

3:45

**4pAAb4. The songs of Tlaloc: Interference of ten ceramical duct flutes, Offering 89 of the Aztec Templo Mayor.** M. A. Adje Both (Inst. for Latin-American Studies, Ruedesheimer Str. 52-54, 14197 Berlin, Germany, adje@antamauna.de)

Among the many preserved sound artefacts deposited in the offerings of the Aztec Templo Mayor are a set of ten tubular duct flutes made from clay, dating Late Postclassic Mesoamerica, 1350–1521 AD. The aerophones are completely painted in blue, and characterized by: (1) a short mouthpiece; (2) a framed aperture; (3) a tube with four fingerholes; and (4) an applied mask with features of the Aztec rain god Tlaloc, basically three rings and a standardized relief structure of two clouds. While all measurements follow the same pattern, one particular organological distinction was made, as five flutes show an exit hole in the middle ring of the mask and five flutes are stopped. Thus, five instruments sound considerably higher, apart from the minimal pitch deviation of each specimen. Both the tonal capacity of each flute and the acoustics of several flutes played simultaneously were recorded and measured. A series of remarkable interference effects could be produced, which were strongly related to the ritual complex reflected in the offering. Taking in consideration the Aztec concept of music, it could be supposed that they were perceived as a principle of the song, or proper voice of Tlaloc.

### Contributed Papers

4:05

**4pAAb5. Acoustical differences between pre-Columbian Mexican aerophones.** J. Guillermo Contreras Arias (Dept. of Ethnomusicology, Natl. School of Music, UNAM, 126 Xicotencatl Coyoacan, D.F. 04100, Mexico and CENIDIM, INBA, Centro Nacional de las Artes)

This paper discusses Mesoamerican musical instruments that produce special sounds and noises. The types of musical instruments to be discussed and demonstrated include those that (a) produce animal sounds; (b) allow performers to exploit multiphonic and beat effects; and (c) allow performers to produce microtonal and polyphonic effects. Examples are

shown of simple flutes, vessel flutes, and whistles of Aztec, Mixtec, Mayan, Olmec, and other pre-Colombian cultures. (To be presented in Spanish.)

4:20

**4pAAb6. Musical instruments from the Mayan culture.** Isabel Elena Romero Rizo, Jose de Jesus Negrete Redondo, and Pablo Roberto Lizana Paulin (Acoust. Lab. ESIME, IPN, Mexico, plizana98@hotmail.com)

This research work presents different kinds of sound generation instruments from the Mayan culture together with the Pablo Castellanos and Arturo Chamorro classification. It has to be noted that most of these mu-



sical instruments, whistles and noise generators are lost and some of them can only be seen on codex and museums. (To be presented in Spanish.)

4:35

**4pAAb7. Ancient noise generators.** Sergio Beristain, Rolando Menchaca, and Roberto Velazquez (Acoust. Lab., ESIME, IPN, Mexico, sberista@hotmail.com)

There has been found a whole family of similar artifacts which produce singing noises. These noises sometimes resemble those sounds generated by some animals and/or naturally produced by strong winds passing through holes and edges. It means that these sounds have wide frequency spectrums and very often some clear tones are identified. The original purpose of these artifacts is unknown, but some researchers think that some were used in mortuary ceremonies and employed by H-men. The

Olmecan whistle previously presented belongs to this family, and now it is compared with a bone or wooden instrument shown in the Florentine codex.

4:50

**4pAAb8. The musical instruments in the codexes mixtecos.** Luis A. Gomez (Coord. de Documentacion, CENIDIM, CNA, Rio Churubusco y Tlalpan s/n, Country Club, Coyoacan, Mexico)

The objective of this work is to describe the inventory of prehispanic musical instruments in the Codexes Mixtec: Vindobonensis, Nuttall, Selden, Colombino, Becker I and II. For this, relative data are gathered as to their quantity, localization and classification; their function in the Meso-American colored writing; and the meaning of the image of which they are part. This inventory will allow academics of different disciplines to select and to locate the image of the musical instrument of interest quickly and to understand in a general way, its function in the image that contains it. (To be presented in Spanish.)

THURSDAY AFTERNOON, 5 DECEMBER 2002

CORAL KINGDOM 2 AND 3, 1:00 TO 2:05 P.M.

### Session 4pAB

## Animal Bioacoustics: History of Animal Bioacoustics

Mardi C. Hastings, Chair

*Biomedical Engineering Center, The Ohio State University, 270 Bevis Hall, 1080 Carmack Road, Columbus, Ohio 43210-1002*

Chair's Introduction—1:00

### *Invited Paper*

1:05

**4pAB1. History of animal bioacoustics.** Arthur N. Popper (Dept. of Biol. and Neurosci. & Cognit. Sci. Prog., Univ. of Maryland, College Park, MD 20742, apopper@umd.edu) and Robert J. Dooling (Univ. of Maryland, College Park, MD 20742)

The earliest studies on animal bioacoustics dealt largely with descriptions of sounds. Only later did they address issues of detection, discrimination, and categorization of complex communication sounds. This literature grew substantially over the last century. Using the *Journal of the Acoustical Society of America* as an example, the number of papers that fall broadly within the realm of animal sound production, communication, and hearing rose from two in the partial first decade of the journal in the 1930's, to 20 in the 1970's, to 92 in the first 2 years of this millennium. During this time there has been a great increase in the diversity of species studied, the sophistication of the methods used, and the complexity of the questions addressed. As an example, the first papers in JASA focused on a guinea pig and a bird. In contrast, since the year 2000 studies are often highly comparative and include fish, birds, dolphins, dogs, ants, crickets, and snapping shrimp. This paper on the history of animal bioacoustics will consider trends in work over the decades and discuss the formative work of a number of investigators who have spurred the field by making critical theoretical and experimental observations.

## Session 4pBB

## Biomedical Ultrasound/Bioresponse to Vibration: Ultrasound Applications

Ronald A. Roy, Cochair

*Department of Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215*

Salvador Echeverria, Cochair

*CENAM, Los Cues, Queretaro, Mexico*

## Contributed Papers

1:00

**4pBB1. Ultrasound detection and computer segmentation of high intensity focused ultrasound (HIFU) lesions.** Neil R. Owen, Michael R. Bailey (Ctr. for Indust. and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698), Yongmin Kim (Univ. of Washington, 406 EE/CSE, Seattle, WA 98195), and Lawrence A. Crum (Univ. of Washington, Seattle, WA 98105-6698)

High intensity focused ultrasound (HIFU) can necrose tumors or cauterize tissue bleeds at intensities on the order of  $1000 \text{ W/cm}^2$ . A synchronized HIFU and *B*-mode ultrasound system reveals a hyperechoic region at the treatment site, which grows with treatment duration and intensity. Our goal was to segment the hyperechoic region representing the lesion via image analysis and measure the ratio of its major and minor axes. Our algorithm uses the RF data as input, processes it, and outputs a binary image that represents the lesions cross-sectional profile. With depth settings from the clinical ultrasound imager, it is possible to calculate lesion dimensions from the binary image. The algorithm was tested with lesions made in a transparent polyacrylamide tissue phantom that became opaque in response to focal heating during HIFU exposure. Lesion size was recorded simultaneously with ultrasound and a CCD camera, and both measurements agreed well. Additionally, computer segmentation agreed well with segmentation by HIFU users blinded to the experimental conditions. The average difference of the determined ratio was 13% for lesions less than 0.5 cm in length. Thus, it is possible to localize precisely the treated tissue region. [Work supported by NSBRI, NSF, and NIH-SBIR.]

1:15

**4pBB2. A quantitative comparison of theory and experiments on high intensity focused ultrasound (HIFU) induced heating in a vascularized tissue phantom.** Ronald A. Roy, Jinlan Huang, and R. Glynn Holt (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, ronroy@bu.edu)

High intensity focused ultrasound (HIFU) can be used to help control bleeding, both from individual blood vessels as well as from gross damage within the capillary bed. The dominant mechanism is rapid localized heating, and vascularity limits ones ability to elevate the temperature of blood vessels owing to convective heat transport. To better understand HIFU-induced heating in tissues with vascular structure, we employ a numerical scheme that couples models for ultrasound propagation, acoustic streaming, ultrasound heating, and blood cooling in Newtonian viscous media. We coordinate the theoretical effort with a series of *in vitro* experiments employing nonuniform flow-through tissue phantoms and designed to provide a ground truth verification of the model predictions. Calculated and measured results are compared over a range of parameter values (insonation pressure, insonation time, and flow rate). We demonstrate excellent agreement between predictions and measurements, and the simulations are extended to a study of the efficacy of HIFU in producing temperature elevations in large and small blood vessels. [Work supported by DARPA and the U.S. Army.]

1:30

**4pBB3. High intensity focused ultrasound for treatment of bleeding and air leaks due to lung trauma.** Shahram Vaezy, Roy Martin, Larry Crum (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98195), Carol Cornejo, Jerry Jurkovich, Sam Sharar (Harborview Medical Ctr., Univ. of Washington, Seattle, WA 98195), and Savvas Nicolaou (Univ. of British Columbia)

In thoracic surgery, bleeding and air leaks from the lungs can be difficult to control. We have investigated the use of high intensity focused ultrasound (HIFU) for control of lung bleeding and air leaks in operative situations. An intraoperative HIFU device, equipped with a Titanium coupler, was used. The HIFU transducer was a PZT-8 concave element, with a focal length of 5 cm, and a diameter of 2.5 cm. The transducer was operated at 5.7 MHz and intensity of  $5000 \text{ W/cm}^2$ . The coupler length was 4 cm, placing the focal volume, defined by full-width half-maximum at approximately 1 cm from the tip of the coupler. A pig animal model was used. Incisions in the lung were made, having lengths of 2–5 cm, and depths of 3–10 mm which created both parenchymal hemorrhage and air leakage from the lung. HIFU was applied within 10 seconds of inducing the injury. The average hemostasis time was approximately 60 seconds. All incisions were completely sealed, and no blood or air leaked from the incisions. Intraoperative HIFU may provide an effective method in various pulmonary surgery indications, and hemostasis and control of air leaks from lacerations due to trauma. [Work supported by NIH and NSF.]

1:45

**4pBB4. A needle-size hydrophone and thermocouple for high-intensity focused ultrasound.** Shahram Vaezy (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98195), Mark Wyzgala, and George Keilman (Sonic Concepts, Inc., Woodenville, WA 98072)

Thermal and acoustic measurements at high acoustic intensities are desirable in studies involving high-intensity focused ultrasound (HIFU). We are developing a needle-size probe comprised of a hydrophone and a thermocouple, that can withstand high-intensity fields without damage. The probe can simultaneously map the acoustic pressure and temperature field in a test medium. The hydrophone will be nominally 25- $\mu\text{m}$  PVDF or PVDF-TrFE with thin-file electrodes, with a flat frequency response from 1 to 20 MHz, isolated from the metallic needle by an ultrasound-absorbing lossy polymer material. The thermocouple (accurate within 1C) signals will be rf-isolated from the hydrophone signals and electromagnetic interference originating from the HIFU source transducer, allowing rapid temperature rise measurements (250 readings per second) to be made with minimal noise and interference. A conformational coating will protect the hydrophone and the thermocouple from cavitation at high acoustic intensities of  $4000 \text{ W/cm}^2$ . The complete assembly will be integrated into a 13-gauge (0.24-mm o. d.) needle (150 mm long). The results of the manufacturing, characterization, and testing in novel, protein-based tissue-mimicking phantoms will be presented. [Work supported by NIH, NSF.]

2:00

**4pBB5. Thrombolytic effects of 120-kHz and 1-MHz ultrasound and tissue plasminogen activator on porcine whole blood clots.** Christy K. Holland, Sampada S. Vaidya, Constantin-C. Coussios, and George J. Shaw (Dept. of Biomed. Eng., Univ. of Cincinnati, 234 Goodman St., Cincinnati, OH 45267-0761)

Stroke is the third leading cause of death and the leading cause of disability in the United States. For patients with ischemic stroke, the thrombolytic drug tissue plasminogen activator (tPA) is the only FDA-approved treatment. To aid in the development of a stroke therapy, the synergistic thrombolytic effect of tissue plasminogen activator (t-PA) and ultrasound was assessed *in vitro* in a porcine clot model. Whole blood clots were prepared from fresh porcine blood by aliquoting 1.5 ml into 8-mm-i.d. glass tubes, immersing the tubes in a 37 °C water bath for 3 h and storing the clots at 5 °C for at least 3 days prior to use in comparative ultrasound and t-PA studies, which ensured complete clot retraction. The 120-kHz or 1-MHz ultrasound peak-to-peak pressure amplitude used for exposures was 0.35, 0.70, or 1.00 MPa. The range of duty cycles varied from 10% to 100% (continuous wave) and the pulse repetition frequency was 1.7 kHz. Clot mass loss was measured as a function of t-PA concentration (without ultrasound) from 0.25 to 8.5 times the human clinical dose. The mass loss increased monotonically as a function of [t-PA] and saturated at 4 times the clinical dose. The degree of the ultrasound enhancement of t-PA was also explored at 120 kHz and 1 MHz. With ultrasound exposure, clot mass loss increased by as much as 250% over sham (t-PA alone). A weak dependence of clot mass loss on duty cycle was noted. We conclude that 120-kHz and 1-MHz ultrasound enhances thrombolysis. [Work supported by Senmed Medical Ventures.]

2:15

**4pBB6. Influence of vasoactive drugs on the induction of microvascular leakage in rat hearts by contrast echocardiography.** Douglas Miller (Dept. of Radiol., 3315 Kresge III, Univ. of Michigan, 200 Zina Pitcher Pl., Ann Arbor, MI 48109-0553, douglm@umich.edu), Peng Li, and William Armstrong (Univ. of Michigan, Ann Arbor, MI)

Diagnostic ultrasound can induce microvascular leakage in rat hearts when a gas-body based contrast agent is present in the circulation, a response which might be partly physiological in origin. Anesthetized rats with tail vein cannulas were mounted in a 37 °C water bath. Rats were pretreated with saline, or with isoproterenol, propranolol or dobutamine, drugs which can modulate microvascular leakage. Evans blue dye and a 0.5 ml/kg bolus of Optison (Mallinckrodt) were injected before 5 min exposures. A diagnostic ultrasound scanner (GE Vingmed System Five) operated at 1.7 MHz and a measured Mechanical Index of 1.5 was used to obtain cardiac images with end-systole triggering each fourth beat. For shams, only a few petechiae were occasionally observed on the excised hearts. For exposure (five rats in each treatment), an average of 208 (86 standard deviation, SD), 200 (72 SD), 236 (52 SD), and 159 (57 SD) petechiae were counted for saline, isoproterenol, propranolol, and dobutamine, respectively (no significant differences). Evans blue dye leakage also appeared to be approximately the same for all treatment conditions. These results suggest that the microvascular leakage induced by contrast echocardiography in rats is primarily a mechanical effect of the ultrasound-induced gas body activity.

2:30

**4pBB7. Effect of absorption on nonlinear propagation of short ultrasound pulses generated by rectangular transducers.** Vera A. Khokhlova, Anatoly E. Ponomaryov (Dept. of Acoust., Faculty of Phys., M. V. Lomonosov Moscow State Univ., Leninskie Gory, Moscow 119992, Russia), Michalakis A. Averkiou (Philips Ultrasound, P.O. Box 3003, Bothell, WA 98041), and Lawrence A. Crum (Univ. of Washington, Seattle, WA 98105)

A numerical solution of the KZK-type parabolic nonlinear evolution equation is presented for finite-amplitude sound beams radiated by rectangular sources. The initial acoustic waveform is a short tone burst, similar to those used in diagnostic ultrasound. The generation of higher harmonic

components and their spatial structure are investigated for media similar to tissue with various frequency dependent absorption properties. Nonlinear propagation in a thermoviscous fluid with a quadratic frequency law of absorption is compared to that in tissue with a nearly linear frequency law of absorption. The algorithm is based on that originally developed by Lee and Hamilton [J. Acoust. Soc. Am. **97**, 906–917 (1995)] to model circular sources. The algorithm is generalized for two-dimensional sources without axial symmetry. The diffraction integral is adapted in the time-domain for two dimensions with the implicit backward finite difference (IBFD) scheme in the nearfield and with the alternate direction implicit (ADI) method at longer distances. Arbitrary frequency dependence of absorption is included in this model and solved in the frequency-domain using the FFT technique. The results of simulation may be used to better understand the nonlinear beam structure for tissue harmonic imaging in modern medical diagnostic scanners. [Work supported by CRDF and RFBR.]

2:45–3:00 Break

3:00

**4pBB8. Contrast enhancement in harmonic imaging using time reversal processing.** Mickael Tanter, Thomas Gauthier, Christophe Barriere, and Mathias Fink (Laboratoire Ondes et Acoustique, E.S.P.C.I., C.N.R.S. UMR 7587, Universit Paris VII, 75005 Paris, France)

Harmonic imaging coupled to the use of contrast agents has become a topic of wide interest in ultrasonic medical imaging. Instead of achieving images at the fundamental frequency of the emission signals, images are built at twice the excitation frequency that corresponds to the resonance frequency of the bubbles embedded in the medium. Consequently, the contrast becomes important between areas of low and high concentration of bubbles. However, at a high mechanical index, the harmonic components of backscattered echoes depends on the intrinsic nonlinear properties of tissues as well as the bubble's resonance. The focused beam is generating harmonics during its nonlinear propagation. It results in a degradation of the harmonic image contrast. Time reversal is an elegant way to find the emission codes allowing us to cancel the harmonic components due to nonlinear propagation of the ultrasonic beam and so to recover an optimal image contrast. Indeed, the wave equation in a nonlinear regime remains time reversal invariant: in the absence of bubbles, if the backscattered echoes are time-reversed and reemitted by the array, the harmonic components of the resulting wavefront are transferred back to the fundamental frequency during propagation. Experiments conducted with 1-D-linear arrays illustrate this cancellation technique.

3:15

**4pBB9. Optical monitoring of ultrasound-induced damage in glass catfish after injection with contrast agents.** Subha Maruvada and Kullervo Hynynen (Harvard Med. School, Cambridge, MA 02139 and Brigham and Women's Hospital)

This work is an investigation of the ultrasound interaction with contrast agent bubbles in the blood vessels of fish *in vivo*. Glass catfish were used for these experiments because they are optically transparent. Anesthetized fish were injected with the contrast agent, Optison, using micro-injection techniques. Shortly after injection, the fish were insonified with one of two single-element focused transducers (1.091 and 0.747 MHz). An inverted microscope combined with a digital camera was used to optically monitor ultrasound interaction with the blood vessels in the tail of the fish at 200× magnification. The entire interaction was videotaped and digitized. The fish were insonified at power levels between 1–80 W, which translated to acoustic pressures from 0.45–3.55 MPa. Sonifications were pulsed with burst lengths of 1 ms, 10 ms, 100 ms, and 100 s, and a

repetition frequency of 1 Hz. The entire length of one sonification at a specific power was 20 s. The effects of the sonification were analyzed at each power level. Ultrasound-induced damage to the fish included structural damage to the cartilage rods in the tail, formation of large gas bubbles within the blood vessels, and hemorrhaging.

3:30

**4pBB10. Application of pulse laser induced vapor bubbles and shock waves to neurosurgery.** Atsuhiko Nakagawa, Takayuki Hirano (Dept. of Neurosurgery, Tohoku Univ. Grad. School of Medicine, 1-1, Seiryu-machi, Aoba, Sendai 980-8574, Japan), Kazuyoshi Takayama, and Tsutomu Saitoh (Tohoku Univ., 2-1-1, Katahira, Aoba, Sendai 980-8577, Japan)

As a part of the basic research of tissue damages which occurred during ESWL treatments, shock/bubble interactions are found to be responsible to tissue damage. Having this background, we have been applying pulse laser induced vapor bubbles and shock waves to various neurosurgical therapies. Pulsed HO:YAG laser beams were transmitted via 0.6 mm diam optical fiber, whose edge was formed in a convex lens shape so as to enhance the laser beam focusing. Laser beams were focused inside a 2 mm diam water filled tube. Then, vapor bubbles and shock waves were generated, which successively drove a microwater jet. The mechanism of the initiation of bubbles and shock waves and consequential jet formation has been intensively investigated. It was found in *in vitro* experiments that microjets wonderfully pierced cerebral thrombi. Combining this system with a very small amount of fibrinolytic agents, cerebral thrombi were removed four times as efficiently as simple irradiation of pulsed laser beams.

3:45

**4pBB11. Mie scattering off coated microbubbles.** Radboud Nelissen, Elmer Koene, Sascha Hilgenfeldt, and Michel Versluis (Appl. Phys., Univ. of Twente, P.O. Box 217, 7500 AE Enschede, The Netherlands, m.versluis@tn.utwente.nl)

The acoustic behavior of coated microbubbles depends on parameters of the shell coating, which are in turn dependent on bubble size. More intimate knowledge of this size dependence is required for an improved modeling of a distribution of coated microbubbles such as found in ultrasound contrast agents (UCA). Here a setup is designed to simultaneously measure the optical and acoustic response of an ultrasound-driven single bubble contained in a capillary or levitated by the pressure field of a focused transducer. Optical detection is done by Mie scattering through an inverted microscope. Acoustical detection of the single bubble by a receiving transducer is made possible because of the large working distance of the microscope. For Mie scattering investigation of excited bubbles, two regimes can be distinguished, which require different detection techniques: Conventional wide-angle detection through the microscope objective is sufficient for bubbles of radius exceeding 10  $\mu\text{m}$ . For smaller bubbles, two narrow-aperture detectors are used to reconstruct the bubble dynamics from the complex angle-dependence of the scattered light.

4:00

**4pBB12. High-speed optical imaging of microbubble dynamics at 40 ns time scale.** Michel Versluis (Phys. of Fluids, Appl. Phys., Univ. of Twente, P.O. Box 217, 7500 AE Enschede, The Netherlands, m.versluis@tn.utwente.nl), Chien Ting Chin, and Nico de Jong (Erasmus Univ., Rotterdam, The Netherlands)

The interactions between ultrasound and microbubble contrast agents, such as nonlinear oscillations, acoustic disruptions, and destruction of microbubbles, are exploited in diagnostic methods such as harmonic imaging and triggered imaging, as well as emerging therapeutic techniques such as controlled drug delivery. Optical visualization at the microsecond to nanosecond time scale has the potential to elucidate such interactions. However, these recordings are limited by image resolution, frame rate, total number of frames, and light sensitivity. We have constructed a digital ultra-high-speed camera system, named Brandaris 128, by combining the flexibility of highly sensitive CCDs with the high number of frames available in rotating mirror cameras. In a standard run our camera records 6 full sequences of 128 digital images at a frame rate of 25 million frames per second. The high number of frames allows for the monitoring of acoustic disruptions and nonlinear oscillations of a single contrast agent microbubble under consecutive diagnostic ultrasound pulses at various amplitudes applied to the same bubble. In this way the contribution of each individual bubble to the acoustic response can be measured, which is essential for further optimization of contrast agents and ultrasound detection methods.

4:15

**4pBB13. Enhancement of the biocidal efficacy of a mild disinfectant through enhanced transient cavitation.** Kenneth A. Cunefare (School of Mech. Eng., Georgia Tech., Atlanta, GA 30332-0405), Stephen D. Carter (Snellville, GA 30078), and Don Ahern (Georgia State Univ., Atlanta, GA 30302)

Cavitation is known to have biocidal effect upon micro-organisms. The work presented here considered the impact of transient cavitation at elevated hydrostatic pressures as a means to improve the biocidal efficacy of otherwise mild disinfectant solutions. Cultures of *Bacillus stearothermophilus* and *Bacillus subtilis* were exposed to a mild isopropyl alcohol-based disinfectant for 15 to 18 minutes, which resulted in negligible cell reduction. Spores from the same cultures were exposed to the disinfectant for 15 to 18 minutes while being cavitated for one minute at a frequency of 24 kHz, and at ambient pressure, with an estimated energy density of 70 watts per liter. This achieved cell reduction of approximately 2%. Spores from the same cultures were exposed to the same cavitation field but at a hydrostatic pressure of approximately 210 kPa. The pressure was selected to exploit "the anomalous depth effect" for enhanced transient cavitation. These conditions achieved cell reduction of approximately 90% after 1 minute of cavitation exposure. The results are suggestive of more efficient means of sterilizing the burgeoning numbers of heat sensitive surgical instruments.

4p THU. PM



**Session 4pEA****Engineering Acoustics: Noise Identification and System Elements**

Federico Miyara, Cochair

*Acoustics and Electroacoustics Laboratory, National University of Rosario, Riobamba 245 BIS, Santa Fe 2000 Rosario, Argentina*

Brandon Tinianov, Cochair

*Acoustical Laboratory, Johns Manville, 10100 West Ute Avenue, P.O. Box 625005, Littleton, Colorado 80162***Chair's Introduction—1:30****Contributed Papers****1:35****4pEA1. Noise source identification: A comparison of seven different techniques.** Marc Marroquin (34465 Hainesville Rd., Round Lake, IL 60073)

One of the most powerful tools that can be used in modern acoustics labs today is noise source identification. By identifying where the noise source(s) may be located, it becomes much easier then to try and reduce/remove/modify the overall dB level of a product. This is typically the end result of a noise reduction program, which is the typical goal of all NVH programs. Over the years many different techniques have become available as technology moves forward. In the beginning it was simple sound-pressure mapping techniques, then sound intensity added a new dimension to the art and science of NSI (noise source identification); last, multichannel applications such as acoustic holography and beamforming have given us tools that far exceed our capabilities of even 10 years ago. As with any advance in technology, each technique has strengths and weaknesses which must be understood to ensure proper measurement and analysis. This paper will offer an overview of seven commonly used NSI techniques along with a comparison of the strengths and weaknesses associated with each technique.

**1:50****4pEA2. The effect of envelope pattern on the impression of sound quality.** Sonoko Kuwano (Osaka Univ., 1-2 Yamadaoka, Suita, Osaka 565-0871, Japan) and Seiichiro Namba (Takarazuka Univ. of Art and Design, Hanayashiki, Takarazuka, Hyogo 665-0803, Japan)

Since the 1970's the present authors have conducted a series of experiments on the effect of envelope patterns on a subjective impression of sounds and a model of dynamic hearing characteristics was proposed on the basis of the results of these experiments. The present investigation was designed to examine the effect of envelope pattern on the impression of sound quality. Three kinds of envelope pattern were used; they were sounds with short rise time and long decay, sounds with long rise time and short decay, and steady state sounds. The duration of each sound was 125, 250, and 500 ms. In total, nine kinds of sound were prepared. The sound was repeated with silent intervals and the total duration was 5 sec. The sound quality was judged using a semantic differential and a paired comparison. It was found that the envelope pattern has a significant effect on the impression of pleasantness and the duration of the sound on the impression of sharpness and articulation.

**2:05****4pEA3. Evaluation of the noise emitted by the condenser of a household refrigerator.** Juan Llado Paris and Beatriz Sanchez Tabuenca (Dept. of Mech. Eng., Zaragoza Univ., 50018 Zaragoza, Spain, juan.llado@posta.unizar.es)

It has been determined that the noise emitted by a household refrigerator is the addition of two main sources: the direct noise emitted by both compressor shells, and the structural noise caused by the condenser vibration, that depends on the coolant excitation, on its own structural rigidity, and on the rigidity of the joints that fix it to the back panel of the refrigerator and to the compressors. The decrease of the sound power due to the condenser implies the modification of their mechanical and geometric characteristics: shape, thickness, type of material, redesign of the joints to the back panel of the refrigerator, and optimization of their number and location, etc., to achieve a less rigid structure. From these proposals, a new more flexible design of the joints was tested because it was the faster one to implement due to the refrigerator efficiency is not altered. A sound power reduction of about 2 dBA, was obtained. Finally, the location and number of the joints were optimized, getting the same sound power level with three than with four joints. (To be presented in Spanish.)

**2:20****4pEA4. Characterization of the noise emitted by a washing-machine due to the pump.** Beatriz Sanchez Tabuenca and Juan Llado Paris (Dept. of Mech. Eng., Zaragoza Univ., 50018 Zaragoza, Spain, bstb@posta.unizar.es)

The noise sources during the pump operation of a washing machine are: the electric motor, the water-air impulsion, and the structural radiation of the pump-housing, hose, and cabinet. The determination of the sound power level under different working conditions let us identify that the noise emitted by the cabinet vibration was the more noticeable, being irrelevant the noise emitted by the other elements. In order to know which part of the cabinet is the noisiest, the standard ISO-TR 7849 was applied to calculate the contribution of the noise emitted by the structural radiation of each part of the cabinet to the total sound power level. This experimental procedure relates the noise radiated by a structure with its vibration velocity, and it was found that the kick plate was the more relevant element. Once the noise transmission paths have been characterized, the proposals to reduce noise are focused on a new design of the kick plate to make it more flexible and the modification of the join zone of the pump to the kick plate to reduce the force transmitted between both elements. (To be presented in Spanish.)



**4pEA5. A study of standing pressure waves within open and closed acoustic resonators.** C. Daniels (OAI, 22800 Cedar Point Rd., Cleveland, OH 44142, Christopher.Daniels@grc.nasa.gov), B. Steinetz (NASA Glenn Res. Ctr., 21000 Brookpark Rd., MS 23-3, Cleveland, OH 44135), J. Finkbeiner, G. Raman, and X. Li (Illinois Inst. of Technol., 10 W. 32nd St., E1, Chicago, IL 60616)

The first section of the results presented herein was conducted on an axisymmetric resonator configured with open ventilation ports on either end of the resonator, but otherwise closed and free from obstruction. In the remaining section we present the results of a similar resonator shape that was closed, but contained an axisymmetric blockage centrally located through the axis of the resonator. Ambient air was used as the working fluid. In each of the studies, the resonator was oscillated at the resonant frequency of the fluid contained within the cavity while the dynamic pressure, static pressure, and temperature of the fluid were recorded at both ends of the resonator. The baseline results showed a marked reduction in the amplitude of the dynamic pressure waveforms over previous studies due to the use of air instead of refrigerant as the working fluid. A sharp reduction in the amplitude of the acoustic pressure waves was expected and recorded when the configuration of the resonators was modified from closed to open. A change in the resonant frequency was recorded when blockages of differing geometries were used in the closed resonator, while acoustic pressure amplitudes varied little from baseline measurements.

## 2:50

**4pEA6. Acoustic characteristics of a three-chamber hybrid silencer.** Iljae Lee, Ahmet Selamet (Ctr. for Automotive Res., The Ohio State Univ., 930 Kinnear Rd., Columbus, OH 43212, selamet.1@osu.edu), and Norman T. Huff (Owens Corning, Inc., Novi, MI 48377)

The acoustic characteristics of a hybrid silencer consisting of two identical single-pass, concentric, perforated dissipative chambers combined with a reactive Helmholtz resonator in between are investigated computationally and experimentally. Transmission loss predictions from a three-dimensional boundary element method are compared with experimental results obtained from an impedance tube setup in the absence of mean flow. In addition to the overall design, the effects of filling material density in dissipative chambers and the neck geometry of the reactive resonator are examined. The dissipative chambers are found to be effective at high frequencies, while the reactive resonator is shown, in general, to improve the acoustic attenuation at low frequencies typical of airborne noise in internal combustion engines running at low- to mid-speed range. The acoustic behavior near the resonance frequency is determined, however, to be sensitive to the duct length connecting dissipative and reactive chambers. Potential merits of the hybrid concept are assessed.

## 3:05

**4pEA7. Frequency-domain methods for modeling a nonlinear acoustic orifice.** David P. Egolf<sup>1)</sup> (Dept. of Elec. and Computer Eng., Univ. of Idaho, Moscow, ID 83844, degolf@ece.uidaho.edu), William J. Murphy, John R. Franks (National Inst. for Occupational Safety and Health, Cincinnati, OH 45226), and R. Lynn Kirlin (Univ. of Victoria, BC, Canada)

This presentation describes frequency-domain methods for simulating transmission loss across a single orifice mounted in an acoustic waveguide. The work was a preamble to research involving earplugs containing one or more orifices. Simulation methods included direct Fourier transformation, linearization about an operating point, and Volterra series. They were applied to an electric-circuit analog of the acoustic system containing the orifice. The orifice itself was characterized by an empirical expression for nonlinear impedance obtained by fitting curves to experimental resistance and reactance data reported by other authors. Their data-collection procedures required the impedance expression presented herein to be properly labeled as a describing function, a quantity well known in the nonlinear control systems literature. Results of the computer simulations were compared to experimental transmission-loss data. For a single-tone input sound pressure, the computer code accurately predicted the output

fundamental (i.e., without harmonics). For a broadband input, the simulated output was less accurate, but acceptable. Levels of the sound-pressure input ranged from 60 to 160 dB. [Work supported by the National Institute for Occupational Safety and Health, Cincinnati, OH, through a research associateship granted the first author by the National Research Council.] <sup>a)</sup>Currently on leave at National Institute for Occupational Safety and Health, Cincinnati, OH.

## 3:20

**4pEA8. Experimental investigation of sound absorbers based on microperforated panels.** Nilda S. Vecchiatti, Antonio M. Mendez (Laboratorio de Acustica y Luminotecnica, C.I.C. Pcia. de Buenos Aires, Cno. Centenario y 506, (1897) M. Gonnet, Argentina, ciclal@gba.gov.ar), and Juan C. Gimenez de Paz (Decibel Sudamericana, Las Bases 165 (1706), Haedo, Argentina)

Microperforated panels have been studied as a good, interesting absorbing element. In previous papers the properties of high absorption obtained in a wide frequency band was demonstrated, based on the impedance of the very small perforations. The perforation area ratio, the diameter of the holes, the thickness of the panel, and the density and viscosity of the air are the terms that define the sound absorption provided by the element. The microperforated panels have a simple structure and it is possible to build single or double resonators, in order to obtain a wide band response. An orifice may be considered as a short tube. Many years ago, Rayleigh and Crandall studied the propagation of sound in small tubes, of a very short length compared to wavelength. They found a high acoustics resistance and a very small reactance. So, the microperforated panel can be used as a dissipative element. An experimental investigation was carried out on different samples of microperforated panels, in order to obtain their sound absorption coefficient and so verify the validity of the mathematical models. Microperforated panels have been developed to cover a welding cabin internally, where classical absorbents are useless. (To be presented in Spanish.)

## 3:35

**4pEA9. Interaction of a slot-tone with a pipe.** J. Alexis Billon, Vincent Valeau, and Anas Sakout (L.E.P.T.A.B., Univ. La Rochelle, Av. M. Crepeau, 17042 La Rochelle Cedex 1, France)

The self-sustained tone generated by a low Mach number free flow impinging on a slotted plate, usually referred to slot-tone, is experimentally studied. In this work we focus on its coupling with the pipe from which the jet flows out. The jet nozzle and the beveled slot are aligned and have the same large aspect ratio. The tones generated have a single well defined frequency (around 1 kHz), a high amplitude (typically, 105 to 115 dB) and exhibit typical characteristics of self-sustained tones. Their frequencies are about the same order of magnitude than the natural frequency of the shear layer calculated from the linear stability theory. For some Reynolds numbers and some plate distances, a standing wave pattern is observed in the pipe. The outer and inner acoustic fields show high coherence and the slot-tone locks on a pipe eigenmode, which indicates an acoustic feedback. However, especially for lower Reynolds numbers, this coherence vanishes but the self-sustained tones are still present. It suggests that the hydrodynamic feedback becomes preponderant.

## 3:50

**4pEA10. Turbulent airflow noise production and propagation patterns of a subsonic jet impinging on a flat plate.** Stephen Martin (Wyle Labs., 128 Maryland St., El Segundo, CA 90245) and William Meecham (UCLA, Los Angeles, CA 90024)

Turbulent airflow noise production is examined utilizing a four-inch diameter jet impinging on a flat plate. The analysis utilizes cross-correlation techniques to investigate the far field sound pressure relationship to turbulent sources produced near the plate surface. The variables examined include the effects of jet speed, the angle of the plate with respect to the jet axis, the location of the turbulent source from the plate center, the location of the far-field microphone, and the plate-jet distance.

The cross-correlation techniques allow discrimination of the far-field sound pressure related to the turbulent noise source located near the surface of the plate. By rotating the far-field microphones, directivity patterns are obtained for a source located a particular distance from the plate center. The resulting cloverleaf shaped directivity patterns indicate a quadrupole source, verifying Powells developments on Lighthills theory on jet noise. The variations of the parameters of the experiment allow a comparison of the directivity patterns for each condition. [Work was performed through the UCLA School of Engineering with the experimental data collected at the NASA-Ames research facility.]

4:05

**4pEA11. The development of polymer layers for vibration damping steel sheet.** T. S. Oh and J. H. Ryou (Dept. of Metal and Coating, RIST (Res. Inst. of Industrial Sci. and Technol.), Pohang, P.O. Box 135, 790-600, South Korea, ohts@rist.re.kr)

The characterization of damping resins, and the forming characteristics of vibration damping steel sheets were carried out in order to evaluate polymer layers for vibration damping steel sheets. The bonding strength and damping properties depend on the chemical nature of individual polymers on the vibration damping steel sheets. Polyester-based thermoplastic polymer shows the best performance. Damping characteristics of vibration damping steel sheet can be well predicted from the dynamic loss factor of resins via the WLF equation. The vibration damping properties of polymers can be changed by a plasticizer. It was found that the service temperature of polymer generally decreased using an increasing amount of

plasticizer. It is concluded that the vibration damping steel sheet made by the loss factor greater than 0.1 in the range of normal service temperature as well as the tensile shear strength greater than 80 kg<sub>f</sub>/cm<sup>2</sup> at room temperature and the addition of Ni powder have been found to improve weldability the same an ordinary steel sheet. [Work supported by POSCO.]

4:20

**4pEA12. Development of beam volume displacement sensors using genetic algorithm.** Saiprasad Jangiti, Marcellin Zahui, and George Bibel (Dept. of Mech. Eng., Univ. of North Dakota, Box 8359, Grand Forks, ND 58202-8359, Marcellin.Zahui@und.edu)

Further development of total (local) volume displacement sensors is presented. This development supports the implementation of noise control techniques that are based on the minimization of local volume displacement of vibrating structures. In this work, a genetic algorithm is used for the design of volume displacement sensors for vibrating beams using PolyVinylidene Fluoride (PVDF). First the sensor is assumed to cover the entire beam surface. Then, the covered area is discretized into small patches. The algorithm selects the required patches to yield the PVDF shape necessary to measure the beam volume displacement. The sensor is numerically verified for various beam boundary conditions. The results show close agreement between the calculated beam volume displacement of the beam and the simulated output charge of the sensor. The extension of the genetic algorithm methodology to volume displacement sensors for 2-D vibrating structures is briefly discussed.

THURSDAY AFTERNOON, 5 DECEMBER 2002

CORAL GALLERY FOYER, 1:00 TO 2:30 P.M.

### Session 4pED

## Education in Acoustics: Take Fives: Sharing Ideas for Teaching Acoustics

Uwe J. Hansen, Chair

*Department of Physics, Indiana State University, Terre Haute, Indiana 47809*

### Chair's Introduction—1:00

Do you have a novel demonstration, a new laboratory experiment, a favorite video, a recorded sound example, or a new idea for teaching acoustics which you are willing to share with your colleagues? At this session a sign-up board will be provided for scheduling presentations. No abstracts are printed. Presenters are encouraged to have handouts to distribute. Multiple presentations are acceptable (not consecutively). Presentations are limited to 5 minutes. Keep them short! Keep them fun!

**Session 4pNS****Noise and Engineering Acoustics: Noise in Urban Communities**

Martha Orozco, Chair

*Div. de Ciencias Biologicas y Ambientales, CUCBA, Universidad de Guadalajara Km 5 1/2, Carr. a Nogales, Las Agujas Zapopan, Jalisco, Mexico***Chair's Introduction—1:00*****Invited Papers*****1:05****4pNS1. Characteristics of successful local noise enforcement programs.** Eric Zwerling (Rutgers Univ. Noise Tech. Assistance Ctr., 14 College Farm Rd., New Brunswick, NJ 08901)

A well crafted local noise ordinance, customized to the needs of the jurisdiction, enforced by motivated and well-trained officers, can resolve long-standing problems, and prevent new ones from arising. The ordinance must be simple enough for all parties to understand: enforcement, prosecution, adjudication, residents and importantly, the regulated community. Uncomplicated performance standards lead to reliably frequent enforcement, the deterrence from which engenders self-policing by source facilities, after initial unsuccessful court challenges to the new ordinance (e.g., St. Augustine, FL). New facilities will be designed for compliance with a well established ordinance (e.g., Denville, NJ). Innovative approaches, such as an interior C-scale standard, successfully curb amplified low-frequency rhythmic emissions from bars without the need for 1/1 or 1/3 octave band analysis (e.g., Anchorage, AK). A “plainly audible” standard for vehicular sound systems is effective, easily enforced (e.g., Lafayette, LA), court tested, and enhanced penalties such as towing can reduce complaints and recidivism to negligible levels (e.g., Rochester, NY). Objective, content-neutral standards successfully control amplified speech (e.g., Long Beach, NY) and street music (New York City). Local communities are increasingly developing noise enforcement programs, many of which are highly effective.

**1:25****4pNS2. Urban noise pollution in the city of Curitiba, Brazil.** Paulo H. T. Zannin and Fabiano B. Diniz (Environ. Acoust. Lab., Federal Univ. of Parana, Parana, Brazil)

This paper presents the results obtained from a study on environmental noise pollution in the city of Curitiba, Brazil. The equivalent sound-level values  $LA_{eq}$ , 2 h were measured and tabulated for 1000 locations spread over the urban zones of the city of Curitiba in several urban zones: residential, mixed, services, downtown, and industrial. It was found that 93.3% out of the surveyed locations display during the day equivalent sound levels over 65 dB(A), and 40.3% out of the total number of locations measured during the day display extremely high values of equivalent sound levels: over 75 dB(A). The average noise levels per zone has ranged from 73.4 to 78.1 dB(A). (To be presented in Portuguese.)

**1:45****4pNS3. Interior noise profile of buses in Curitiba.** Paulo H. T. Zannin, Clifton R. Giovanini, Fabiano B. Diniz, and Jose C. Ferreira (Environ. Acoust. Lab., Federal Univ. of Parana, Parana, Brazil)

Evaluating the noise levels to which the bus drivers of Curitiba are exposed to during their working days is the main scope of this study. The city is served by an internationally known public transportation system featuring 1902 buses, which attend 1.9 million people per day. Two measurements have been taken inside each one of the 60 buses surveyed, one close to the driver and another one at the back of the bus. The results have showed that the dose levels the drivers are exposed to were below 50% in 92% out of the buses, but the normalized exposure levels were over 65 dB(A) in all cases. This level is considered as the threshold of comfort according to the Brazilian legislation on occupancy health NR-17—Ergonomics. The surveyed buses have been divided into three categories, according to their characteristics: feeder, rapid, and bi-articulated. A total of 20 buses within each category have been surveyed. Among the different categories, it has been found that the feeders have presented the highest noise levels. (To be presented in Portuguese.)

2:05

**4pNS4. Analysis of the levels of ambient noise present in the Colonia Auditorium (October 2001).** Zapopan Jalisco, Lourdes Palafox, Martha Orozco, and Erika Rodriguez (Div. de Ciencias Biologicas y Ambientales, CUCBA, Universidad de Guadalajara Km 15 1/2 Carr. a Nogales, Las Agujas Zapopan, Jalisco, Mexico)

The study area is located in the city of Guadalajara, defined by: delimited in the north by the Gonzalez Gallo street; in the south by the Anillo Perifrico street; in the east by Avenida Mezquitán; and in the west by Avenida Alcalde. There were 38 measurement points outside the auditorium and 25 inside. A poll was conducted among the neighbors and visitors in both areas, getting 70 replies. Noise levels were between 41–101 dB(A) on the outside, average Leq on the outside was 64.5 dB(A), and inside was 80.15 dB(A). The conclusion is that the whole area is affected by traffic noise, as well by the huge number of visitors, the electric games, and the electric generators that supply energy to them. On the outside 21 points were above 65 dB(A), which means that 55% were above the average; on the inside, 100% were above 65 dB(A). Thus the neighbors are exposed to high noise levels due to the festivities during October. The biggest problem, however, is the noise exposure of the workers inside the auditorium, due to music, games, people, and electric generators.

2:20

**4pNS5. Noise levels at critical points in the municipality of Guadalajara, Jalisco, Mexico.** Arturo Figueroa, Jesus Garcia, Jorge Macias (Depto. de Estudios y Proyectos Amb. Direccin General de Medio Ambiente y Ecología, H. Ayuntamiento de Guadalajara, Marsella 49, 5to. piso, Mexico), Martha Orozco, Javier Garcia (Las Agujas, Zapopan Jalisco, Mexico), and Alan Delgadillo (Secretaria de Educacion, Guadalajara, Jalisco, Mexico)

Studies of acoustic conditions are planning tools on which we can diagnose the problem of noise pollution in the cities. The first study on noise pollution made in the city was made by the University of Guadalajara in 1995 and updated in 1998 covering with measuring points the city center. This paper discusses the problem of noise pollution by motor vehicles at critical points and covers a total of 105 points. The study also analyzes the problem of noise pollution base on the community annoyance from which a regulation policy should derive. Results of the study show that the most critical points are located within zone I (center) where Leq levels within the range of 70–85 dB were found. Such levels exceed by far the international standard of 65 dB as recommended for ambient noise by the World Health Organization.

2:35

**4pNS6. Acoustic performance of roadside barriers in urban environments.** Kai Ming Li and Siu Hong Tang (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong)

Extensive research has been conducted for assessing the acoustic performance of a barrier in a sub-urban area where the receivers are normally situated at a considerable distance from the barrier. However, in a typical urban community, a noise barrier is frequently erected in front of a tall building. It provides a shadow zone to listeners on the other side of the barrier. The size of this shadow zone depends on the height and position of the barrier away from the building facade. Sound waves are diffracted at the top edge of the barrier before they reach the receivers either directly, or on reflection from the ground and/or from the facade surface. Diffraction and multiple scattering play important roles in determining the sound pressure levels between the facade and barrier. Furthermore, the possibility of a direct wave between the source and the receivers higher up the building requires further evaluations when considering the total sound fields. In this paper, the theory behind the facade-barrier system is outlined. Indoor scale model experiments are conducted to validate theoretical predictions. [Work supported by Research Grants Council of the Hong Kong Special Administrative Region and The Hong Kong Polytechnic University.]

2:50

**4pNS7. Noise barrier for the Temuco ByPass Highway.** Christian Gerard, Aldo Campos (Control Acustico Ltda., Rogelio Ugarte 1817, Santiago, Chile), and Sergio Belfiore (Tubosider America Latina Las Condes, Santiago, Chile)

Increases in traffic and the emergent Highway Concession Program created by the Public Works Ministry of the Chilean government, have led to the necessity of providing acoustic protection in inhabited areas in which the level of noise exceeds that allowed by the standard. Around the world, the installation of acoustic barriers has been one of the main measures of the controlling traffic noise. This is the case with the Temuco ByPass Highway, an extension of 21 km, located in a rural area typical of the environment and landscape of the South of Chile. A total of 6000 lineal meters of transparent acoustic barrier (Methacrylate) was installed. This barrier is the first one of its type to be installed in South America, and theoretical calculations and practical results are analyzed in this work. (To be presented in Spanish.)

3:05–3:20 Break

3:20

**4pNS8. Measurement of urban traffic noise due to heavy cargo transport services.** Francisco Ruffa, Daniel S. Gavinowich, Erica A. Barkasz, and Pablo R. Ciccarella (LACEAC, School of Eng., Paseo Colon 850, 1piso Dpto. Electr., 1063 Buenos Aires, Argentina)

The Acoustic and Electroacoustic Laboratory of the Buenos Aires University is working on a Traffic Noise Recommendation to be used in Argentina. To that end, measurements of traffic noise due to heavy cargo transport services were taken in a Buenos Aires neighbourhood, especially asked for by a neighborhood association. The goal of this measurement was to try to define a correct standard set for the annoyance level in order to pass that information to the city government and work together to improve the living standard of the inhabitants. The measured indexes were Leq, L90 and L10, as well as octave measurement focused on frequencies under 250 Hz. This paper will show the results of the measurements and the conclusions that were made.

3:35

**4pNS9. Absorption characteristics of porous media: A model for traffic-noise reducing asphalt.** Hasson M. Tavossi (Dept. of Physical and Environ. Sci., School of Natural Sci. & Mathematics, Mesa State College, Grand Junction, CO 81501)

This paper presents preliminary investigation into the feasibility of the use of granular materials, with special grain size distribution as the asphalt constituents, for selective absorption of traffic-noise—generated by the engine and tires-asphalt-interactions—of the roads and highways passing near residential areas. The near-term goal of this research is laboratory data collection and modeling on the sound absorption properties of asphalt model-media, selecting media with required absorption properties, and analysis of their traffic-noise absorption characteristics. Long-term efforts are aimed at the general reduction of traffic-noise reflected from the surfaces of these highways by selective absorption of noise of different frequency bands in the granular constituents of asphalt. Reducing noise from the road surfaces by selective sound absorption at the surface results in lowering of the noise level pollution of the streets in the residential areas and the communities situated close to highways.



**4pNS10. Analysis of a pollution problem of ambient noise and bases for a municipality inspection in the center of Tlaquepaque, Jalisco, Mexico, 2001.** Erika Rodriguez, Martha Orozco, and Lourdes Palafox (Div. de Ciencias Biologicas y Ambientales, CUCBA, Universidad de Guadalajara Km 15 1/2 Carr. a Nogales, Las Agujas Zapopan, Jalisco, Mexico)

The city of Tlaquepaque is delimited at the north by Guadalajara city, at the west side by Zapopan, at the south side by Tlajomulco and El Salto, and in its east side by Tonalá. Our purpose in this study is to create the first noise map of Tlaquepaque, being the study area between the streets: Avenida Niños Heroes, Revolucion, Florida, and Marcos Montero. We take 53 points of which 77.3% are above 65 dBA, the Leq average of the points was 69.7 dBA, having as a minimum 44.8 dBA and as a maximum 108.6 dBA. Nowadays Tlaquepaque does not have any noise map, so we are trying to identify its more complex areas. The high dB levels found in the study show us how high the noise in the street is, and this is reflected by several citizen complaints about noise received by the municipality. Because of this it is necessary to have an ambient noise diagnostic program, and do the necessary modifications to the current inspection and monitoring system. The preliminary results of this first noise map show a really serious problem with ambient noise caused by vehicular traffic, bad urbanization, and the existence of commercial centers that raise the ambient noise.

## 4:05

**4pNS11. Sound pollution in schools nearby highways crossing Buenos Aires City.** Daniel S. Gavinowich, Francisco Ruffa, Erica A. Barkasz, and Pablo R. Ciccarella (Acoust. Lab., Electron. Dept., School of Eng., Buenos Aires Univ., Paseo Colon 850, 1 Piso, 1063 Buenos Aires, Argentina)

According to an agreement between the concessionary company of the highways crossing Buenos Aires City (AUSA) and the School of Engineering of The Buenos Aires University, measurements of the sound level were made in the neighborhood of primary schools and high schools near the roads. The purpose of this work was to supply registered values to be able to obtain a general knowledge of sound levels "in front of" and "inside" schools, as reliably as possible. Relying on the previous measurements of some points, the registered values can be compared with the previous ones and possible changes of the sound level produced in the last times can also be estimated. Finally, the work was completed with the study of possibility of acoustic attenuation using noise barriers in critical areas because of high levels of pollution.

## 4:20

**4pNS12. Balanced noise control design: A case study for co-generation power plant.** Yong Ma and Salem Hertil (ATCO Noise Management, 1243 McKnight Blvd. NE, Calgary, AB T2E 5T1, Canada)

Power generation plant generally requires noise mitigation treatment to achieve the specified noise regulations. In this paper, a case study of the noise control design for a cogeneration power plant was presented. Major noise sources included two GE gas combustion turbines, two generators, two heat recovery steam generators (HRSGs), one steam turbine and generator, one 12-cell cooling tower, and other accessory equipment. The acoustic modeling software Cadna/A was used to predict the noise contributions from sources. During the acoustic modeling, alternative noise mitigation measures underwent two specific investigations before they were chosen as a noise solution recommendation. The first was to determine the technical feasibility of attenuating the source equipment. The second was to perform a cost benefit analysis, necessary to find the most cost-effective solution. For example, several acoustic wall and roof assemblies were entered into the acoustic model and the acoustic performance of the ventilation system was varied until we were able to arrive at the most economical acoustic solution. This is the premise on which so called balanced design is based.

**4pNS13. Declaration and verification of noise emission values of machinery and equipment in Russia.** Ilya Evseevich Tsukernikov (SC "NIIPolugrammash," 57 Profsoyusnaya St., 117420 Moscow, Russia, odpm@rmail.ru) and Igor Alekseevich Nekrasov (JSC "Algorithm-Acoust.," 107370 Moscow, Russia)

Original positions of the interstate standard GOST 30691-2001 are considered, in which the requirements are established to declaration and verification of noise emission values of machinery and equipment, being manufactured in Russia and imported there from different countries. The comparison of the standard with the International standard ISO 4871:1996 is conducted. The main differences are marked. The suggestions for standards revision are made.

## 4:50

**4pNS14. Study on the correlation between field experiments and laboratory experiments for audible environmental noise.** Takeshi Tokashiki (Faculty of Eng., Univ. of the Ryukyus, 1 Senbaru Nishihara cho Nakagami gun Okinawa Pref., Japan), Yasuhiro Yamashita, and Naoki Takagi (Shinshu Univ., 500 Wakasato Nagano shi, 380-0922, Japan)

Our objective in this study is to clarify, by some experiments, that there would be a different response and that subjective appraisal differs when audible sound types change even under the same sound level. So far various methods that we, appraise sound environment by our subjectivity, have been proposed and utilized. But we have seen a few study cases about the correlation between sound types, subjective appraisal, and, moreover, sound level. In this research, we have surveyed environmental noise in terms of data obtained by physical quantity (LAeq), and subjective appraisal and audible noise types, then examined the correlation between them. Consequently we could confirm their audible sound types might influence noise level and subjective appraisal. Moreover, as a result of the surveillance of noise environment along boulevards, commercial, and residential areas, the sound heard in each area constitutes the same ratios. The data obtained by subjective appraisal on each area is almost the same. This fact indicates we might infer an annoying response through its noise types.

## 5:05

**4pNS15. Analysis of Spanish adjectives to develop a Likert scale useful for studies of urban noise annoyance.** Rosendo Vilchez-Gómez, Juan Miguel Barrigón-Morillas, Valentín Gómez-Escobar, Juan Antonio Méndez-Sierra (Universidad de Extremadura, Escuela Politécnica, Avda. de la Universidad, s/n-10071 Cáceres, Spain, vilchez@unex.es), and José Manuel Vaquero-Martínez (Universidad de Extremadura, Escuela Politécnica, 10071 Cáceres, Spain)

The present work describes the development and analysis of a Spanish annoyance scale for use in community noise assessments. The use of different descriptors and intervals for different authors prevents comparing studies of assessment of noise reactions. It is not clear how many semantic distinctions can be made to describe subjective reactions to noise. There is also great variability in language use reflecting educational, social class, regional, and subcultural differences. Finally, scaling annoyance responses, when the considered intervals are not equally distributed, may seriously distort results when analyzed with parametric statistics. Using three different questions, almost 400 subjects were asked to rate 25 different Spanish descriptors of annoyance. Our attempt to produce a Spanish standardized annoyance scale with descriptors marking clear semantic distinctions, roughly equidistant from each other, and having wide acceptability is presented here. Taking into account that Spanish is the official language in 21 countries and it is spoken by more than 300 million people, one of our goals in this meeting is to discuss our method and try to generalize it in order to obtain a real, universal Spanish annoyance scale.



**Session 4pPA****Physical Acoustics: Global Infrasound Monitoring II**

Milton A. Garces, Chair

*Infrasound Laboratory, University of Hawaii, Manoa, 73 4460 Queen Kaahumanu Highway, #119,  
Kailua Kona, Hawaii 96740-2632***Invited Papers****1:15**

**4pPA1. Wind noise reduction research: Possible lessons for infrasound?** Richard Raspet (Dept. of Phys. and Astron., Univ. of Mississippi, P.O. Box 1848, University, MS 38677) and Michael A. H. Hedlin (Inst. of Geophys. and Planetary Phys., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0225)

Morgan [Ph.D. dissertation, University of Mississippi, 1992] extended the theory of wind noise reduction by spherical screens proposed by Phelps [RCA Rev. **3**, 203–212 (1938)] to modern porous foam windscreens and achieved good predictions of the large (> 12 dB) low frequency reduction afforded by spherical screens even when the turbulence scale is much greater than the wind screen size. In this paper the findings of Morgan are reviewed as background for recent measurements of the infrasonic wind noise reduction of a wind barrier design suggested by Ludwik Liska. The data suggest that significant reductions in infrasonic wind noise can be achieved with devices smaller than a turbulence scale size.

**1:35**

**4pPA2. Large meteoroid detection using the global International Monitoring System infrasound system.** Douglas O. ReVelle (Los Alamos Natl. Lab., P.O. Box 1663, MS J577, Los Alamos, NM 87545, revelle@lanl.gov)

We will review the subject of infrasound from large bolides (large meteor-fireballs) entering the atmosphere at hypersonic speeds and their expected rate of detection by the 60 infrasonic arrays of the global IMS network (International Monitoring System). This will include the details of the generation of a quasiline source blast wave and its subsequent decay for near-continuum flow conditions. We will also discuss new highly refined models of bolide ablation and fragmentation and of known compositional types and their effect on sound and light production. In addition, we will consider the effects of refraction of the waves by the middle atmospheric and tropospheric thermodynamic sound speed and horizontal wind profiles in a range-independent atmosphere so that the characteristic velocity and wave normal directions radiated at the source are conserved during the propagation. Next, we will discuss the detection of the signals and their interpretation in terms of plane wave arrivals regarding the 3-D source location (latitude, longitude, height), the source energy level, etc. Finally, we will use the infrasound data from bolides to estimate the expected steady-state global influx rate, including formal errors, as a function of their observed source energy. Infrasound from recent large events will also be examined.

**1:55**

**4pPA3. Global infrasound monitoring—Research issues.** Henry E. Bass, Kenneth Gilbert (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, P.O. Box 1848, University, MS 38677), Milton Garces, Claus Hetzer (Univ. of Hawaii, Kailua-Kona, HI 96740), Gene Herrin, Paul Golden (Southern Methodist Univ., Dallas, TX 75275), John Olson, Daniel Osborne, Curt A. L. Szuberla, Charles Wilson (Univ. of Alaska, Fairbanks, AK 99775), Jon Berger, Michael Hedlin (Scripps Inst. of Oceanogr., La Jolla, CA 92093), Rod Whitaker, Doug Revelle (Los Alamos Natl. Lab., Los Alamos, NM 87545), Bob Woodward, Bob North (Ctr. for Monitoring Res., Arlington, VA 22209), and Richard Kromer (Sandia Natl. Lab., Albuquerque, NM 87185)

The International Monitoring System being installed to support monitoring compliance with the Comprehensive Nuclear Test Ban Treaty provides scientists with a unique opportunity for research. There are still a number of problems which limit the full exploitation of the system. These include limitations on signal-to-noise imposed by wind noise and the absence of well defined, internationally accepted calibration standards for sensors. But perhaps the major research challenges lie in the area of source characterization and definition. Most of the signals recorded at the few sites now operating come from unidentified sources. There has been some effort devoted to identifying local and regional sources but the unidentified category still exceeds 50% of all distinct events. There are a number of infrasound sources that occur naturally. These include volcanic eruptions, bolides, microbaroms, mountaintop/wind interactions, severe storms, and earthquakes. Manmade sources include most any energetic activity. After sorting out all these sources, there may remain more exotic sources of infrasound not yet identified. Therein lies a major source of excitement.

2:15

**4pPA4. Evaluation of rosette infrasonic noise-reducing spatial filters.**

Michael A. H. Hedlin (Inst. of Geophys. and Planetary Phys., Univ. of California, San Diego, San Diego, CA), Benoit Alcoverro (Commissariat a l'Energie Atomique, Departement Analyse et Surveillance de l'Environnement), and Gerald D'Spain (Scripps Inst. of Oceanogr., Univ. of California)

The spatial noise filter currently preferred for use at the new International Monitoring System (IMS) infrasound array sites consists of an array of low-impedance inlets connected by solid tubes to a microbarometer. We present results from recent tests of "rosette" infrasonic noise-reducing spatial filters at the Pinon Flat Infrasound test-bed in southern California. At wind speeds up to 5.5 m/s, the 96 inlet 18-m rosette filter reduces wind noise levels above 0.2 Hz by 15–20 dB. Under the same conditions, the 144 inlet 70-m rosette filter provides noise reduction of up to 15–20 dB between 0.02 and 0.7 Hz. Standing wave resonance inside the 70-m filter degrades the reception of acoustic signals above 0.7 Hz. Synthetics accurately reproduce the noise reduction and resonance observed in the 70-m filter at all wind speeds above 1.25 m/s. Experiments with impedance matching capillaries indicate that internal resonance in the rosette filters can be removed. Rosette filters are tuned to vertically incident energy. Attenuation of signals by the 70-m rosette filter at frequencies above 3.5 Hz arriving at grazing angles of <15 deg from the horizontal are predicted to range upward from 10 dB to total cancellation at 5 Hz.

2:30

**4pPA5. An optical fiber infrasound sensor.** Mark A. Zumberge, Jonathan Berger, and Michael A. H. Hedlin (Univ. of California, San Diego, La Jolla, CA 92093-0225)

A new sensor for detecting infrasonic signals from distant sources has been designed, tested, and deployed. The instrument consists of a long (of order 100 m), compliant, tubular diaphragm tightly wrapped with optical fibers. The air-filled diaphragm is sealed from the ambient atmosphere and therefore expands and contracts radially along its length as the pressure changes. An optical system monitors minute variations in the optical path length of the fibers, which change as the tube expands and contracts. The new instrument responds to the spatially integrated pressure average along its length. As it relies on optical interferometry, it does not suffer from the propagation delays inherent to mechanical noise filters and as a result can be made very long. Above 1 Hz, the optical fiber infrasound sensor (OFIS) is less noisy than sensors relying on mechanical filters. The records collected from an 89 m long sensor indicate a new low noise limit in the band from 1 Hz to 10 Hz. Current research focuses on the relationship between sensor length and wind-induced noise, on the utility of the sensor as a directional infrasound antenna, and on the sensor's performance at audio frequencies.

2:45–3:00 Break

3:00

**4pPA6. A study of microbarom wave packets.** John V. Olson, Curt A. L. Szuberla, and Charles R. Wilson (Geophysical Inst., Univ. of Alaska, Fairbanks, AK 99775)

Microbaroms are a class of atmospheric infrasound that is characterized by narrow-band, nearly sinusoidal waveforms with periods near 5 s. These waves are generated by marine storms and are observed at infrasound sites across the globe. We report here on a study of microbarom

wave packets. The data analyzed were obtained using the University of Alaska infrasound array comprised of 4 microphones located in central Alaska. Exploiting the narrow-band feature of the microbarom signals we are able to apply the Hilbert transform as a method for finding phase breaks in the signal. The phase breaks are interpreted to be the demarcations of the boundaries of individual wave packets. When applied to long sequences of microbaroms we find a broad distribution of packet lengths that diminishes monotonically with length, with a mean near 10 cycles and a variance nearly as large. Once the packets are identified, high-resolution estimators have been used to obtain the distribution of arrival directions at the array site. The ensemble of arrival directions is a reflection of the possibility of multiple sources and/or multiple propagation paths between the sources and the array.

3:15

**4pPA7. Observations of infrasonic signals from natural and manmade sources.**

John M. Noble, Stephen M. Tenney (Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, jnoble@arl.army.mil), R. W. Whitaker, and D. O. Revelle (Earth and Environ. Sci. Div., Los Alamos Natl. Lab., Los Alamos, NM 87545)

We report on data recorded from small aperture (20-m) infrasonic microphone arrays. These data will include the well-observed Pennsylvania bolide of 23 July 2001, space shuttle launches, sounding rocket launches, and local severe storm passages. Data were obtained from two infrasonic arrays in southern Maryland. Analysis results from two independent packages will be presented, including bearing estimates and detector performance. Bolide detection results will be correlated with eyewitness reports of the flight trajectory. Estimates will be made of the bolide source energy from the application of the historical explosion scaling law relating the period at maximum acoustic amplitude. Detection of sounding rockets from over 150 km will be discussed. Finally, data from the passage of several severe storms will be presented showing the detection and tracking of a low-frequency acoustic signature of the storm.

3:30

**4pPA8. Detection and interpretation of infrasonic signals observed in Hawaii.**

Milton Garces and Claus Hetzer (Infrasound Lab., Univ. of Hawaii, Manoa, 73-4460 Queen Kaahumanu Hwy. #119, Kailua-Kona, HI 96740-2632)

Various array processing and signal detection algorithms have been tested and evaluated at International Monitoring System infrasound array I59US, also known as KONA, located on the western side of the Big Island of Hawaii. The array consists of four elements deployed as a triangle with a central element, with a baseline of 2 km. Initial tests were made by defining multiple beams and running STA/LTA, coherence, and *F*-statistic detectors to estimate the apparent horizontal phase velocity and incidence azimuth of arrivals in frequency–wavenumber space. Such methods generally proved effective only for infrasonic signals with signal-to-noise ratios (S/N) greater than ~4, and their performance varied with signal frequency. The Progressive Multi-Channel Correlation (PMCC) detector is a time–domain detector that uses the correlation between various groupings of three sensors to obtain an estimate of the consistency of specified closure relations. If the consistency is below a certain threshold, a detection is registered. This detector has performed well for S/N ratios that are close to unity and for all frequencies. Examples of infrasonic detections in Hawaii for small (Leonids 2001) and large bolides (23 April 2001), blasting activity, surf, thunder, severe storms, and low-frequency signals of unknown origin are presented.

**4pPA9. Co-located atmospheric infrasonic and seismic recordings of rocket launches.** Gerald L. D'Spain, Michael A. H. Hedlin, John A. Orcutt, William A. Kuperman, Catherine de Groot-Hedlin, Galina L. Rovner, and Lewis P. Berger (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 93940-0704)

Atmospheric infrasound data and co-located, three-component seismic data have been collected by the eight microbarometers of the International Monitoring System (IMS) station and the IRIS seismic station at Pinon Flat (PFO) plus five additional microbarometer/space filter systems at five Anza seismic stations located within 40-km range of PFO in Southern California. Characteristics of the infrasound and seismic recordings from this large-horizontal-aperture array of signals from 400-km-distant rocket launches at Vandenberg Air Force Base are analyzed using waveguide invariant theory. The Navy standard Gaussian Ray Bundle (GRAB) underwater acoustic propagation code (with slight modifications), along with launch trajectory information and atmospheric data collected at the time of the launches, is used to examine the predictability of the signal arrival structure. The predictions take into account the signal-distorting effects caused by phase delays across the spatial aperture of the space filters, which cause each infrasound array element to be directional over the frequency band of interest. [Work supported by the Defense Threat Reduction Agency.]

4:00

**4pPA10. Propagation variability and localization accuracy of infrasonic networks.** David E. Norris and Robert G. Gibson (BBN Technologies, 1300 N. 17th St., #1200, Arlington, VA 22209)

Prediction of propagation variability induced by the environment is used to evaluate the localization performance of infrasonic networks. The dominant source of variability affecting infrasonic propagation is believed to result from gravity waves. A gravity wave spectral model based on scale-independent diffusive filtering is used to generate multiple wind perturbation realizations. A Monte Carlo simulation is executed where rays are traced through the perturbed environmental fields, and uncertainty in ray travel time and azimuthal deviation is calculated. The propagation uncertainties, along with uncertainty in infrasonic measurements, are then used to compute 90 percent confidence bounds of multi-station event localizations. Infrasonic data from the April 2001 Pacific bolide event are used to compute the performance of a five station network, and the network localization is compared to that found from satellites. [Work sponsored by Defense Threat Reduction Agency, Contract No. DTRA01-00-C-0063.]

**4pPA11. Long-range infrasound propagation modeling using updated atmospheric characterizations.** Robert G. Gibson and David E. Norris (BBN Technologies, 1300 N. 17th St., #1200, Arlington, VA 22209, rgibson@bbn.com)

Infrasonic waves can propagate thousands of kilometers in range and sample regions of the atmosphere from the ground up to and including the thermosphere. Conventional infrasound propagation modeling techniques rely on climatological models of mean temperatures and winds to characterize the environment. However, temperature and wind vary over temporal and spatial scales that are not captured by climatological models. Recent work addresses the integration of infrasound propagation models, such as three-dimensional ray tracing, with numerical weather prediction models, such as the Navy Operational Global Atmospheric Prediction System (NOGAPS). Propagation results are computed using both climatological and updated atmospheric characterizations, and comparisons are presented. Implications for global infrasound monitoring are discussed. [Work supported by the Defense Threat Reduction Agency.]

4:30

**4pPA12. Determination of uncertainty in the estimation of velocity and direction-of-arrival for atmospheric infrasound signals.** Curt A. L. Szuberla and John V. Olson (Geophysical Inst., Univ. of Alaska, 903 Koyukuk Dr., Fairbanks, AK 99775-7320)

Upon the detection of an atmospheric infrasound signal, the problem of precisely estimating the signal's velocity ( $v$ ) and direction-of-arrival ( $\theta$ ) arises. Multiple sources, multipath, medium anisotropies, and other propagation effects can all degrade precision; however, uncertainty in the estimates of  $v$  and  $\theta$  is fundamentally governed by array geometry and the estimation of time delays across the array. Typically, as in the Comprehensive Test Ban Treaty Organization Provisional Technical Secretariat (CTBTO/PTS) specification for data from infrasound stations, the Cramér–Rao lower bound is invoked to ascertain the uncertainties associated with  $v$  and  $\theta$ . As this theoretical lower limit is often overly conservative, a more general, and useful, approach to calculate these uncertainties is developed. Examples of this uncertainty determination are presented for typical impulsive and continuous atmospheric infrasound signals received at arrays in Windless Bight, Antarctica and Fairbanks, Alaska. Since the determination of  $v$  and  $\theta$  serves as primary input to any propagation model, it is critical that uncertainties in these estimates be addressed. As an extension of this work, an interactive graphical tool is constructed to assist in the analysis of performance bounds for arbitrary array geometries and signal characteristics.

## Session 4pSAa

## Structural Acoustics and Vibration: Reconstruction of Acoustic Radiation from Vibrating Structures II

Earl G. Williams, Chair

Naval Research Laboratory, Code 7137, Washington, D.C. 20375

Chair's Introduction—1:00

## Invited Papers

1:05

**4pSAa1. A stopping rule for the conjugate gradient regularization method for ill-posed problems.** Thomas DeLillo and Tomasz Hrycak (Dept. of Mathematics and Statistics, Wichita State Univ., 1845 N. Fairmount, Wichita, KS 67260, hrycak@math.twsu.edu)

We present a novel parameter choice strategy for the conjugate gradient regularization algorithm which does not assume *a priori* information about the magnitude of the measurement error. Our approach imitates the truncated singular value decomposition within the Krylov subspaces associated with the normal equations. Conjugate gradient is implemented using the Lanczos bidiagonalization process with reorthogonalization. We compare our method with one proposed by Hanke and Raus and illustrate its performance with numerical experiments, including an inverse problem of acoustic source detection.

1:35

**4pSAa2. Multipole error estimation of an acoustical boundary-element solver combined with a dual-layer CHIEF approach.** Martin Ochmann (TFH-Berlin, Univ. of Appl. Sci., Luxemburger Strasse 10, D-13353 Berlin, Germany, ochmann@tfh-berlin.de) and Alexander Osetrov (St. Petersburg State Electrotechnical Univ., 197376 St. Petersburg, Russia)

The sound radiation from vibrating structures is studied using an iterative boundary-element method. The starting point is a self-adjoint formulation of the Helmholtz integral equation. The generalized minimum residual method (GMRES) is used for solving the corresponding system of equations. Three aspects of the method are investigated. First, the accuracy of the iterative solver is checked by performing multipole error tests: The body is forced to vibrate with the corresponding surface normal velocity of a multipole. Hence, the radiated sound field is analytically known and can be used for determining the exact error caused by the boundary-element solver. Second, instabilities at irregular frequencies are suppressed by combining the iterative solver with a special variant of the CHIEF method, where the total boundary of the body is retracted to an auxiliary surface lying totally within the interior of the structure. CHIEF points can be placed on such an auxiliary surface. The effectiveness of this combined dual-layer GMRES–CHIEF approach and of the Burton–Miller method will be compared. Third, the radiation from a structure with mixed boundary conditions will be investigated. One part of the surface is coated with an absorbing impedance, another part is vibrating with a prescribed normal velocity.

2:05

**4pSAa3. Iterative solution techniques for boundary element method (BEM) equations.** Steffen Marburg, Stefan Schneider, and Has-Juergen Hardtke (Institut fuer Festkoerpermechanik, Technische Universitaet, D-01062 Dresden, Germany, marburg@ifkm.mw.tu-dresden.de)

Boundary-element discretization of the Kirchhoff-Helmholtz integral equation gives rise to a linear system of equations. This system may be solved directly or iteratively. Application of direct solvers is quite common but turns out to be inefficient for large scale problems with 10 000 unknowns and more. These systems can be solved on behalf of iterative methods. This paper is dedicated to testing performance of four iterative solvers being the Restarted Bi-Conjugate Gradient Stabilized algorithm (RBiCGStab), the Conjugate Gradient method applied to the normal equations (CGNR), the Generalized Minimal Residual (GMRes), and the Transpose Free Quasi Minimal Residual (TFQMR). For that, it is distinguished between internal and external problems. Performance of iterative solvers is investigated with respect to different parameters. It turns out that sophisticated preconditioning is required for the hyper-singular external Burton and Miller formulation of nonsmooth meshes.

## Contributed Paper

2:35

**4pSAa4. A review optimization technique in structural acoustics.** Steffen Marburg (Institut fuer Festkoerpermechanik, Technische Universitaet, D-01062 Dresden, Germany, marburg@ifkm.mw.tu-dresden.de)

In recent years, several activities on structural-acoustic optimization have been reported. The problem of structural-acoustic design optimiza-

tion is characterized by nonlinearities. For that reason, it can only be dealt with numerically. Furthermore, the calculation of the objective function is usually very time-consuming since a multifield boundary value problem has to be solved. This paper summarizes results of a longer review paper. It will briefly discuss fast analysis techniques like FEM and BEM, objective functions, sensitivity analysis and optimization methods. Applications and parameter choices are categorized.



## Session 4pSAb

## Structural Acoustics and Vibration: Measurement and Analysis Techniques

Gerard P. Carroll, Chair

Naval Surface Warfare Center, Carderock Division, Code 7250, 9500 MacArthur Boulevard, Bethesda, Maryland 20817-5000

## Contributed Papers

3:00

**4pSAb1. Identification of very closely spaced modes using an iterative algorithm.** Matt S. Allen, Jerry H. Ginsberg, and Aldo Ferri (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The algorithm of mode isolation is a frequency domain method for processing measured response data in order to identify the modal properties of a system. It has been shown [M. V. Drexel and J. H. Ginsberg, Proceedings of the 19th IMAC, Orlando, FL, 5–8 February 2001] to accurately evaluate a pair of damped modes whose natural frequencies differ by an amount that is commensurate with the bandwidth of either mode. Here, SIMO measurement of a two-degree-of-freedom system is simulated analytically by adding white noise to the computed response. This system is selected because the natural frequency difference can be made as small as desired by adjusting a parameter that has little effect on the modal damping ratios in the range of interest. Analytically, the two normal modes are unique and mutually orthogonal if the frequency difference is nonzero, while repeated frequencies lead to two arbitrary, and not necessarily mutually orthogonal, modes. The present work explores the degree to which AMI tracks the analytical behavior, specifically, its ability to detect and characterize the modes with decreasing frequency difference.

3:15

**4pSAb2. The effect of high spatial resolution on techniques for experimental determination of loss factor.** Gerard P. Carroll, Matthew A. Craun, and Daniel T. DiPerna (Naval Surface Warfare Ctr. Carderock Div., 9500 MacArthur Blvd., Bethesda, MD 20817-5700, carrollgp@nswccd.navy.mil)

Using a common set of mobility functions, various methods for determining loss factor are compared. A scanning laser Doppler vibrometer was used to measure these mobility functions with high spatial resolution for a plate and beam sample. The advantages of high spatial resolution are explored by estimating the loss factor using the complete set of mobility functions, and then comparing the results to those obtained using smaller subsets of the measured data. The plate and beam specimens are treated with a free layer damping. The loss factor determination methods include: synthesized time decay, classical modal analysis methods, the power input method, and complex wave number analysis. The merits/limitation of the damping measurement methods and the possible advantages of high spatial resolution are compared, along with results for both test specimens.

3:30

**4pSAb3. Autocorrelation of vibration data for flaw detection.** Sally J. Pardue and Brahmaji Vasantharao (Dept. of Mech. Eng., Tennessee Technol. Univ., Box 5014, Cookeville, TN 38505, spardue@tntech.edu)

The autocorrelation of a structural FRF (frequency response function) can be used to extract structural parameters, such as length from a transversely vibrating beam. Further work has shown that the technique can also detect flaws in the structure. The frequency spacing between resonant peaks is extracted from the autocorrelation process and relates to the length of the object. Indicators of flaws in the structure were observed in the autocorrelation values plotted against a length axis. The first test structure was a free-free nine feet long, one inch diameter aluminum rod. The

flaws examined were machined slots of various widths cut into the rod. The concept was further explored numerically in ANSYS models of the same aluminum bar using much smaller flaw sizes. Now, the concept of autocorrelation of the FRF is being applied to composite beams, 8 ply, 10 in. long, 0.052 in. thick, 1 in. wide. Indicators in the vibration data are sought to relate to porosity levels of the manufactured composite beams.

3:45

**4pSAb4. Phase determination of reverberant structural-acoustic systems using pole and zero distribution.** Juan F. Betts (The Aerosp. Corp., P.O. Box 92957-M4/907, Los Angeles, CA 90009, Juan.F.Betts@aero.org) and Christopher Fuller (Virginia Tech, Blacksburg, VA 24061)

This study investigates the phase behavior of a simple coupled structural acoustic system with high modal overlap. It proposes that phase is simply accumulated across the components making up the structure. Classical dynamic solutions breakdown with high modal overlap and complex geometries, and standard Statistical Energy Analysis (SEA) solutions work well for high modal overlap but give no phase information. Many applications such as active control predictions and time-domain signal analysis require phase information. Results of the investigation show that the amount of damping in the system is very important in the estimation of reverberant phase. The amount of energy lost due to coupling with adjacent subsystems was also demonstrated to be important for both acoustic and structural subsystems. The reverberant phase of an acoustic subsystem was found to be affected more by the inclusion of coupling loss to an adjacent structure than the reverberant phase of a structure attached to an acoustic subsystem. The phase of the overall system is calculated by simply accumulating the phase of each of the subsystems.

4:00

**4pSAb5. Communications/Navigation Outage Forecasting System (C/NOFS) statistical energy analysis.** Juan F. Betts (The Aerosp. Corp., P.O. Box 92957-M4/907, Los Angeles, CA 90009, Juan.F.Betts@aero.org)

The objective of this study was the assessment of the dynamics response of the Communications/Navigation Outage Forecasting System (C/NOFS) spacecraft due to incident acoustic environments. C/NOFS is space vehicle (SV) launched using a Pegasus launch vehicle (LV). A statistical energy analysis (SEA) model of the SV was developed from a NASTRAN finite-element model (FEM). The study compared the dynamic response of the structural subsystems (panels and structural elements) due to incident acoustic Pegasus protoflight environment [which is 3 dB above maximum expected flight environment (MEFL)] and compared it against the Mil-340 standard workmanship acoustic levels. A concern was raised as to whether the workmanship acoustics, required by Mil-340, imposed on C/NOFS may induce higher vibration responses than what C/NOFS was qualified for, and may result in unnecessary increases in component test levels. The two incident acoustic spectrums were introduced as reverberant loads on the model. The PSD results at the structural panels were compared to random vibration test levels performed on the components for C/NOFS.



4:15

**4pSAb6. Effects of circumferential air gaps on the measurement of the absorption coefficient of poroelastic materials.** Dominic Pilon and Raymond Panneton (GAUS, Dept. of Mech. Eng., Université de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada, dominic.pilon@hermes.usherb.ca)

The influence of mounting conditions on the measurement of the absorption coefficient is investigated. More specifically, the effects of circumferential air gaps on poroelastic materials inside the standing wave tube are studied. The objective is to identify the materials, in terms of a ratio based on their physical properties, for which it is possible to measure the theoretical absorption coefficient using the tube. The difference, or error, between the measured and theoretical absorption coefficient is evaluated for a wide range of materials with various sample sizes. These errors are then sorted in terms of the chosen ratio. It is shown that for certain values of the ratio, the theoretical absorption coefficient can be efficiently measured using the standing wave tube. Through the use of this acousto-visco-inertial criteria, an experimenter can determine which absorption is going to be measured: the theoretical absorption coefficient or one that will be influenced by either the circumferential air gaps or the size of the sample.

4:30

**4pSAb7. Experimental investigation of sound transmission through thin-walled structures due to nonuniform pressure-fluctuation fields. Part 2: Supersonic flow.** Boris M Efimtsov, Alexey Yu Golubev, Alexander Ya Zverev (TsAGI, Acoust. Div., 17 Radio St., 107005 Moscow, Russia), Anders O. Andersson, and Sergey V. Kravchenko (Boeing, Seattle, WA 98124)

Theoretical models of the sound-energy transmission through thin-walled structures from nonuniform, convecting pressure-fluctuation fields

were validated through experiments. The fields were created with separated flows at forward-facing and backward-facing steps and an airfoil as the shock generator. Thin plates and model fuselage panels were used and vibrations and acoustic radiation of the structures were investigated. The data permit a direct comparison with predicted results as well as an evaluation of the validity of the prediction-model assumptions. In particular, the elastic and inertial properties of the skin but not of the stiffeners need to be taken into account. The experiments support the validity of previously reported theoretical predictions.

4:45

**4pSAb8. Transmission loss across a rectangular partition or aperture.** Colin Fox and Hyuck Chung (Mathematics Dept., The Univ. of Auckland, PB 92019, Auckland, New Zealand)

We give closed-form solutions for sound propagation through a rectangular aperture, or plate, in a finite or infinite acoustically opaque barrier. This model is useful in describing airborne excitation and reradiation through window openings or through floor/wall partitions that can be modeled in terms of bending stiffness and mass. While mode-matching solutions are routinely available for these problems, we are not aware of any previous analytic solutions. The value of the closed-form solution, other than the obvious ease of calculation, is in establishing scaling laws for the various regimes of structure-borne and air-borne sound. We derive the analytic solution using an extension of the Wiener-Hopf technique as applied to sound propagation in ducts with a partial rigid barrier.

THURSDAY AFTERNOON, 5 DECEMBER 2002

GRAND CORAL 3, 1:00 TO 5:00 P.M.

### Session 4pSC

## Speech Communication: Spanish and English in Contact and Other Cross-Language Studies (Lecture/Poster Session)

Winifred Strange, Cochair

*PhD Program in Speech and Hearing, City University of New York Graduate School, 365 Fifth Avenue, New York, New York 10016*

Ann R. Bradlow, Cochair

*Department of Linguistics, Northwestern University, 2016 Sheridan Road, Evanston, Illinois 60208*

Chair's Introduction—1:00

### Invited Papers

1:05

**4pSC1. Sensitivity to voiceless closure in the perception of Spanish and English stop consonants.** Mary L. Zampini (Dept. of Spanish & Portuguese, Univ. of Arizona, P.O. Box 210067, Tucson, AZ 85721), Constance M. Clarke, and Linda W. Norrix (Univ. of Arizona, Tucson, AZ 85721)

The duration of voiceless closure that precedes the release of a stop consonant is a temporal cue that, like voice onset time (VOT), varies across languages. This talk will examine the interaction between VOT and voiceless closure in Spanish and English and will focus on monolingual and bilingual listeners sensitivity to changes in the duration of voiceless closure during perception. Experimental data will show that monolingual Spanish listeners mean VOT boundaries decrease as the duration of voiceless closure increases. This pattern is consistent with the finding that Spanish speakers produce voiceless stops with longer voiceless closure durations than voiced stops [K. P. Green, M. L. Zampini, and J. Magloire, *J. Acoust. Soc. Am.* **102**, 3136 (1997)]. It will also be shown that

monolingual Spanish listeners show greater sensitivity to voiceless closure than monolingual English listeners. Lastly, the impact that experience with both languages has on perception by late English–Spanish bilinguals will be discussed. It will be shown that the bilinguals under study are affected by their first language (English) while listening to tokens in isolation, but perform like monolingual Spanish listeners when listening to tokens in a Spanish sentence context.

1:35

**4pSC2. Solving conflicts between vowel systems.** Paola Escudero (Inst. of Linguist. UiL OTS, Utrecht Univ., 10 Trans, Utrecht 3512 JK, The Netherlands, Paola.Escudero@let.uu.nl)

Conflicts may arise when listeners encounter a language with a different number of vowels. For instance, Spanish has 5 vowels, English at least 9 and Spanish listeners have many difficulties perceiving the /i/–/I/ contrast in some (though not all) varieties of English. Spanish learners of English solve this few-to-many problem by incorporating a new auditory dimension (vowel duration) into their categorization system. Perhaps surprisingly, the reverse scenario, namely encountering a language with fewer vowels, is also problematic. For instance, Dutch listeners of Spanish (Dutch is comparable to English in its vowel inventory size) have a boundary mismatch when listening to the Spanish /a/–/o/ contrast, and they hear the extra category /I/ when listening to Spanish /i/–/e/. Dutch learners of Spanish shift their boundaries toward what would be appropriate for the target language and they gradually stop perceiving the extra category. It turns out that both in few-to-many and many-to-few situations learners are capable of changing their perception to become like or almost like the native speakers of the target language. This can be explained within a formal model of speech perception and its development.

2:05

**4pSC3. Development of the Hearing In Noise Test (HINT) in Spanish.** Sigfrid D. Soli, Andrew Vermiglio (House Ear Inst., 2100 W. 3rd St., Los Angeles, CA 90057, ssoli@HEI.org), Karen Wen (House Ear Clinic, 2100 W. 3rd St., Los Angeles, CA 90057), and Carolina Abdala Filesari (Cochlear Corp. Latin America, Caracas, Venezuela)

Assessment of functional hearing ability is relevant to clinical outcome assessments, occupational health evaluations, and forensic applications. Communication with spoken language is a primary aspect of functional hearing ability. Comparable speech tests in multiple languages are required to make these assessments in a multilingual population. This presentation will report on an ongoing international research project to develop a Latin American Spanish version of the Hearing In Noise Test (HINT) for this purpose. The methods of selecting speech materials, recording the materials, synthesizing appropriate masking noise, equating the difficulty of the materials, forming sentence lists, norming the lists, and determining reliability and sensitivity will be discussed. Samples of Spanish, as well as other languages (English, Japanese, Mandarin, Cantonese, and Canadian French) will be used to demonstrate the procedures by which standard dialects of each language were selected. Cross-language comparisons of the spectral and temporal characteristics of the speech materials will be presented. Norms, reliability coefficients, and measurement errors for each language will also be reported. Methods for comparing and/or equating functional hearing ability across languages will be described. Finally, procedures for measuring functional hearing ability will be discussed.

2:35–2:40 Break

### *Contributed Papers*

All posters will be on display from 2:40 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:40 p.m. to 3:40 p.m. and contributors of even-numbered papers will be at their posters from 3:40 p.m. to 4:40 p.m.

**4pSC4. Within- and across-language acoustic variability of vowels spoken in different phonetic and prosodic contexts: American English, North German, and Parisian French.** Winifred Strange (Speech and Hearing Sci., City Univ. of New York—Grad. Ctr., 365 Fifth Ave., New York, NY 10016-4309), Andrea Weber, Erika Levy, Valeriy Shafiro, and Kanae Nishi (City Univ. of New York—Grad. Ctr., New York, NY 10016-4309)

This project investigates the acoustic variability of vowels produced in multisyllabic nonsense words /cvC1VC2(v)/ in carrier sentences by speakers of American English (AE), Parisian French (PF), and North German (NG). Variables under examination are (1) immediate phonetic context (C1=b,d; C2=b,d,p,t), (2) sentence prominence (narrow focus versus postfocus), and speaking rate (normal vs rapid). Preliminary results show that for AE vowels, phonetic context produces large differences (2–3 bark) in midpoint *F*<sub>2</sub> values for mid-to-high back vowels, reflecting allophonic fronting of these vowels in alveolar context. Much smaller increases in *F*<sub>2</sub> (~1 bark) were found for both PF and NG mid/high back vowels in alveolar contexts. Most AE speakers showed little or no effect of sentence

focus or speaking rate on the degree of fronting of back vowels in alveolar context. PF speakers produced more fronted mid/high back vowels in postfocused context than did NG and AE speakers. NG and PF speakers showed relatively greater *F*<sub>1</sub> undershoot in low vowels in postfocus context than for AE speakers. These within- and across-language differences in the effect of prosodic and phonetic context on vowel acoustics reflect both language-universal constraints and language-specific rules for contextual warping of vowel spaces.

**4pSC5. Perception of word-final /t/, /d/, /s/, and /z/ by beginning Spanish-speaking learners of English.** Roberta Pikser (Ph.D. Prog. in Linguist., CUNY Grad. Ctr., 365 Fifth Ave., New York, NY 10016) and Winifred Strange (CUNY Grad. Ctr., 365 Fifth Ave., New York 10016)

Previous research has indicated that speakers of Spanish learning English have problems perceiving and producing the word-final /t/–/d/ contrast [Flege, 1992]. In this study we examined whether beginning learners

of English from Latin America could distinguish word-final voicing and manner contrasts among /d/, /t/, /s/, and /z/ when produced in three different vowel contexts by two speakers. As expected, subjects had difficulty correctly discriminating the voicing distinction in both stops and fricatives. On average, performance was poorer on /t/-/d/ than /s/-/z/. Subjects had no difficulty distinguishing voiced or voiceless consonants differing in manner. There was a clear vowel effect; performance was poorer on average for voicing contrast in the high front vowel context than for low central and high back contexts, although there were individual differences. There was also a strong speaker effect; subjects more easily identified the /d/-/t/ contrast of the male speaker, who had longer closure durations and stronger release bursts than those of the female speaker.

**4pSC6. Spanish listeners' use of vowel spectral properties as cues to post-vocalic consonant voicing in English.** Geoffrey Stewart Morrison (Dept. of Linguist., Univ. of Ottawa, 70 Laurier Ave. E., P.O. Box 450, Stn. A, Ottawa, ON K1N 6N5, Canada, geoff@japan.co.jp)

Mexican Spanish listeners who had just arrived in an Anglophone region of Canada were tested on an edited-natural-speech continuum for Canadian English /bit bit bid bid/ in which vowel duration and vowel spectral properties were varied, and in which consonant closures were silent. The Mexican Spanish listeners did not use vowel spectral properties to identify the vowel: Vowel identification was near chance level with a tendency for longer vowel stimuli to be identified as /i/. However, they did use vowel spectral properties to identify the consonant: Stimuli containing vowels with low *F1* were identified as having voiceless consonants, and stimuli containing vowels with high *F1* were identified as having voiced consonants. This presentation will consider possible explanations for this identification pattern for consonant voicing, and will also present the results of additional tests conducted with the same participants, namely: an identical perception test conducted 6 months after the participants' arrival in Canada, English production tests using the same words conducted at the same time as the two perception tests, and a Spanish production test using the words /bit bid bíti bídi bíti bídi/ conducted at the same time as the first set of English tests.

**4pSC7. Could lack of experience with a second language be modeled as a hearing loss?** Monica Padilla (Dept. of Biomed. Eng., University Park Campus OHE 500, Univ. of Southern California, Los Angeles, CA 90089) and Robert V. Shannon (House Ear Inst., Los Angeles, CA 90057)

Previous studies have shown that listeners who learned a second language later in life have poorer speech recognition compared to native listeners, particularly under difficult listening conditions. We are interested in quantifying this demonstrated deficit experienced by non-native listeners as a function of the length of the exposure to the second language. In particular, we are interested in modeling it in terms of Plomp's *D* factor. In Plomp's model [J. Speech Hear. Res. 29, 146–154 (1986)], *D* is defined as a hearing loss due to distortion. We tested listeners whose first language is Spanish with English phonemes, words and sentences in noise, and with reduced spectral information. Preliminary results showed that *D* increased with an increased loss of spectral resolution. *D* also increased with the age of immersion in the second language. These results suggest that a lack of experience with the second language may be modeled as a type of hearing loss. Other issues of interest to us in this study are: the effect of the conflicting vowel spaces of the two languages in vowel recognition and the use of context by non-native listeners compared to native English listeners. [Work supported by NIDCD.]

**4pSC8. Nonsense syllable perception by monolingual and bilingual English speakers.** Catherine L. Rogers (Dept. of Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620), Maria R. Brea-Spahn, Mei-Wa Tam, and Edna E. Nyang (Univ. of South Florida, Tampa, FL 33620)

Understanding speech in demanding environments is essential for daily communication. Previous research has shown that even highly proficient bilinguals may experience greater difficulty than monolinguals in understanding speech in noise. In the present study we further address this issue by examining the effects of varying task demand, fatigue, and practice on speech perception by bilinguals. One group of monolingual English listeners and three groups of Spanish–English bilinguals (bilingual since childhood, bilingual since teenager, and bilingual since adulthood) listened to nonsense syllables presented in noise and at increasing presentation rates. Listeners twice completed the two speech perception tasks on two days of testing. On one day the speech tasks were preceded by and on the other day followed by approximately 40 minutes of testing on non-speech auditory tasks. Monolingual and bilingual listeners' overall performance, their performance across the two days of testing (practice effect), and their performance across test orders (fatigue effect) will be compared. Preliminary results from two monolingual and two highly proficient bilinguals (bilingual since age 16 or earlier) show similar overall performance on both speech tasks and no effect of fatigue (test order), but suggest a greater practice effect for bilinguals' perception of syllables in noise.

**4pSC9. Other-perception, self-perception and production of English phonemes by adult speakers of Spanish.** Ellie Hanlon (Dept. of Educational Psych., CUNY, 365 Fifth Ave., New York, NY 10036, ehanlon1@gc.cuny.edu)

A causal model of second-language production was tested. It was hypothesized that correct production results from accurate perception of unfamiliar second language (L2) phonemes when spoken by others (other-perception) and the learner (self-perception). For the correct production to develop, the learner must form new phonetic categories for the novel L2 phonemes by listening to and observing native speakers. The categories are then used by the learner as an internal standard against which to judge production attempts. The participants' production was measured by recording a nonword syllable reading task. Self-perception was measured by asking participants to listen to their recordings and identify phonemes using a two-alternative forced choice format. Other-perception was measured the same way as self-perception except the stimuli were recorded by a native English speaker. The path analysis shows the squared multiple correlation coefficient for production when other-perception and self-perception are used as predictors is 0.83 (83% of the variance in production is accounted for by other- and self-perception). However, the relationship between other-perception and production is nonsignificant when controlling for the effects of self-perception. This suggests that learners are developing an internal standard through other-perception skill and using this standard to adjust their attempts at pronouncing novel L2 phonemes.

**4pSC10. Effects of native language experience in Spanish–English bilinguals' ability to discriminate between two similar cross-language phonetic categories.** Janielle Lugo-Marin and Cynthia M. Connine (Psych. Dept., SUNY-Binghamton, P.O. Box 6000, Binghamton, NY 13902-6000)

Two phoneme monitoring experiments examined bilingual listeners' (Spanish/English) ability to discriminate similar L2 (American English) phonemes ("sh" and "ch") when L1 (Spanish) contains only one of the phonemes ("ch"). Experience with the "ch" phoneme in L1 was predicted to function as a perceptual magnet for the English phoneme "sh," which is also a voiced palatal consonant. Both "sh" (Experiment 1) and "ch" (Experiment 2) targets were used. Target phonemes occurred in the initial or

final position of a carrier word. Response times for the “sh” targets in initial position were slower than for the “ch” targets. In word-final position, the “ch” targets were detected more slowly than “sh” targets. Accuracy rates were comparable for the “sh” and the “ch” targets in both positions. The results are discussed in terms of categorical generalization of L2 phonemes that are similar to L1 phonemes, knowledge of phonological rules about the likelihood of a voiced palatal consonant in word-final position in L1, and experiential factors with L2.

**4pSC11. Does difficulty perceiving American English /r/ and /l/ affect Japanese listeners’ lexical confusion of these phonemes?** Ryo Komaki (ATR Human Information Sci. Labs., 2-2-2 Hikaridai, Keihanna Science City, Kyoto 619-0288, Japan, komaki@atr.co.jp)

Phonetic environment has proven to be one of the most important factors affecting non-native English speakers’ perception of American English /r/ and /l/ [e.g., Lively *et al.*, *J. Acoust. Soc. Am.* **96**, 2076–2087 (1994)]. In the present study native Japanese speakers’ lexical confusion of English /r/ and /l/ are examined through a word translation experiment which employed pairs of English words minimally contrasting /r/ and /l/ in various word positions. Results indicate that English words in which the /r/ or /l/ is located in a perceptually difficult to identify position (e.g., initial consonant cluster: “branch” versus “blanch”) are more likely to be mistranslated than those in which the phoneme is located in an easy to identify position (e.g., final singleton: “compare” versus “compel”). This finding suggests that errors in phoneme perception and lexical confusion might be related. Pedagogical implications for second language learning will be discussed with respect to results from both learning and retention experiments. [Work supported by TAO, Japan.]

**4pSC12. Effect of age on perceptual learning of second-language phonetic contrasts.** Reiko Akahane-Yamada, Tomoko Takada, and Rieko Kubo (ATR Human Information Sci. Labs., Kyoto 619-0288, Japan, yamada@atr.co.jp)

This paper is a mid-term report on an examination of the effect of age on learning new second language (L2) phonetic categories. Approximately 1000 Japanese speakers of various ages, from children (10 years old) to elderly (70 years old), were trained to identify English words produced by native speakers of American English. The words used minimally contrasted in /r/-/l/ for all age groups (10–70), and words contrasting in /b/-/v/ and /s/-/th/ were additionally used only for the younger age groups (10–22). A pretest–post-test design was administered in which the effect of the training was evaluated by the amount of improvement in accuracy from the pretest to the post-test. Results obtained at this point suggest that: (1) in general, the amount of improvement decreases with age; (2) during the early phases of training, the /r/-/l/ distinction is an extremely difficult one for Japanese children; and (3) elderly subjects, even in their 60s, can improve significantly. Implications for the theories of acquisition of speech perception will be discussed. [Work supported by TAO, Japan.]

**4pSC13. Does semantic context help Japanese speakers identify English /r/ and /l/?** Amanda Rothwell and Reiko Akahane-Yamada (ATR Human Information Sci. Labs., Kyoto 619-0288, Japan, rothwell@atr.co.jp)

Second language learners experience difficulty distinguishing phonemic contrasts that do not occur in their native language. In this study we investigated the effect of providing Japanese speakers with a semantic context when identifying /r/s and /l/s in English words. Questions addressed were: (1) whether semantic context improves performance; and (2) if context helps, whether the written or spoken context is most helpful. Japanese speakers participated in a two alternative forced-choice identification task in which participants identified words they heard by choosing

one of two words contrasting in /r/ and /l/. Words were presented either: (a) in isolation; or (b) within contextual sentences in a variety of written and spoken forms. Preliminary results show that Japanese speakers’ identification accuracy of the /r/-/l/ contrast was significantly better when words containing the contrast were presented in contextual sentences than when they were presented in isolation. The amount of improvement in performance was almost equal, whether the context was provided in written or spoken form, or a combination of thereof. These results suggest that a semantic context, both written and spoken, helps second language learners’ identification of difficult phonemic contrasts. Implications of this work on second language learning will be discussed. [Work supported by TAO, Japan.]

**4pSC14. Perceptual assimilation patterns and the perception of English /l/ and /ɹ/ by Japanese speakers.** Katsura Aoyama (Div. of Speech & Hearing Sci., Univ. of Alabama at Birmingham, CH20, 1530 3rd Ave. S., Birmingham, AL 35294, aoyama@uab.edu) and James E. Flege (Univ. of Alabama at Birmingham, Birmingham, AL 35294)

This study examined the relationship between the perceptual assimilation patterns and the discrimination of English /l/ and /ɹ/ by Japanese speakers. The participants were 50 Japanese adults differing in their length of residence (LOR) in the U.S. In an identification task, they usually identified both English /l/ and /ɹ/ as Japanese /r/. In a rating task, English /l/ was judged to be significantly more similar to Japanese /r/ than English /ɹ/ was. The participants with a longer LOR gave lower similarity ratings for both English /l/ and /ɹ/ than those with a shorter LOR, suggesting that they became more aware of phonetic differences between Japanese /r/ and English /l/ and /ɹ/. An exploratory analysis on similarity ratings was conducted for a subset of participants ( $N=28$ ) based on their scores on /l/-/ɹ/ discrimination. Both relatively poor and relatively good discriminators judged /l/ to be more similar to Japanese /r/ than /ɹ/, but the degree of difference between /l/ and /ɹ/ in terms of similarity ratings as Japanese /r/ was smaller among poor discriminators than better discriminators. This suggests that the degree of “goodness-of-fit” as an L1 category may affect the ability to discriminate two L2 consonants. [Work supported by NIH.]

**4pSC15. Neural activation during the discrimination of native and non-native speech contrasts by American and Japanese adults.** Feng-Ming Tsao, Patricia K. Kuhl, and Huei-Mei Liu (CMBL 357988, Univ. of Washington, Seattle, WA 98195)

Language experience influences speech perception in adults. Adult speakers have been shown to experience difficulty when discriminating non-native consonants that are not phonemic in the native phonetic repertoire. The present study examined behavioral and neural differences in the processes underlying the perception of native and non-native speech sounds. American-English and Japanese native-speaking adults participated in a functional magnetic resonance imaging (fMRI) study. The study focused on three consonants: the American English /r-/l/ consonants and Japanese /r/. The investigation focused on neural correlates of both Kuhl’s perceptual magnet effect (PME) and the boundary effects typical of categorical perception (CP). Subjects performed in a behavioral task in which they identified provided goodness ratings, and discriminated stimuli varying in formants 2 and 3. Results will be discussed in relation to cortical activation patterns related to the PME and CP boundary effect for native and foreign language contrasts, and the effects of language experience on brain organization. [This work was supported by NIH grant (HD37954) and Talaris Institute.]



**4pSC16. Examining developmental patterns of neural activation during the discrimination of native and non-native speech contrasts by American–English adults and children.** Jessica C. Pruitt, Tobey L. Nelson (Ctr. for Mind, Brain and Learning & Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA 98105), John S. Pruitt (Microsoft Corp., Redmond, WA), and Patricia K. Kuhl (Univ. of Washington, Seattle, WA)

To better understand the effects of language experience on speech perception, the present study examined developmental differences in the neural processes underlying the perception of native and non-native speech contrasts. Native American-English speaking adults and pre- and post-pubescent children participated in a functional magnetic resonance imaging (fMRI) study in which they discriminated non-native consonants from Hindi, Spanish, and Mandarin. The non-native contrasts differed in place of articulation, voicing, and manner. For comparison with the neural data, behavioral data were collected outside the scanner for the non-native and a native contrast (AE *r/l*) using a child-friendly computer-based testing method that presented the same contrasts to all subject groups. Results are discussed with regard to developmental differences in the location and extent of activation patterns for the native and non-native contrasts and the level of difficulty for the individual contrasts. [Work supported by NICHD grant HD37943 to Patricia K. Kuhl and Talaris Research Institute.]

**4pSC17. Brain potentials in response to native and non-native speech contrasts in 11-month-old American infants.** M. Rivera-Gaxiola, J. Silva-Pereyra, A. Garcia-Sierra, L. Klarman, and P. K. Kuhl (Speech and Hearing Sci. Dept. and Ctr. for Mind, Brain and Learning, M.S. 357920, Univ. of Washington, Seattle, WA 98195)

Behavioral studies show that 6- to 8-month-old infants discriminate both native and non-native phonetic differences, but that by 10–12 months of age, infants' perceptual sensitivities decline to many non-native contrasts. Kuhl *et al.* (1997) and Tsao *et al.* (2000) have also shown an enhancement in sensitivity to native contrasts, which was also shown by Cheour *et al.* (1998) using electrophysiological methods. Using both behavior and ERPs, Rivera-Gaxiola *et al.* (2000) reported MMNs in native English speaking adults for a non-native contrast that the same adults did not overtly discriminate. We here report auditory ERPs to native and non-native VOT contrasts in 11-month-old monolingual American infants ( $N = 16$ ). Stimuli were presented as a double-oddball paradigm. Continuous EEG was recorded using electrocaps. ERPs were obtained off-line using standard methods. The results show that 11-month-olds display differential brain activity for native and non-native contrasts. A clear, widely distributed phonetic MMN can be observed for the native contrast, while the non-native acoustic differences pre-attentively detected are shown as differences in the first positive peak of the complex. A smaller centro-parietal negativity was also observed for the non-native contrast. [Work supported by NIH (HD 37954) and by the Talaris Research Institute.]

**4pSC18. Native and non-native perception of phonemic length contrasts in Japanese: A categorization study.** Hiroaki Kato and Keiichi Tajima (ATR Human Information Sci. Labs., Keihanna Science City, Kyoto 619-0288, Japan)

Segment length is distinctive in Japanese, for example, /kaze/ (“wind”) versus /kaze:/ (“taxation”). Such length contrasts are not necessarily categorical for non-native speakers. To study this property precisely, a series of perception experiments were conducted. Stimuli were based on a pair of nonsense Japanese words, /erete/ and /erete:/. A continuum gradually varying between these words was synthesized using STRAIGHT, a high-fidelity speech analysis, synthesis, and manipulation system [Kawahara *et al.*, *Speech Commun.* 27, 187–207 (1999)]. Three subject groups were tested: (1) English speakers with no Japanese experience; (2) English speakers who had spent 1–6 months in Japan; and (3) native Japanese speakers. All subjects participated in a single-stimulus

two-alternative forced-choice identification task. Subjects in the second group were tested a second time after five days of perceptual training on phonemic length contrasts during which they were exposed to 1800 pairs of real Japanese words that minimally differed with respect to vowel length. Results suggest that identification boundaries of English speakers are less sharp than those of native speakers, and that identification boundaries sharpen with both exposure to Japanese and with perceptual training. Implications of these findings for second-language learning will be discussed. [Work supported by TAO, Japan.]

**4pSC19. Native and non-native perception of phonemic length contrasts in Japanese: Effect of identification training and exposure.** Keiichi Tajima, Amanda Rothwell (ATR Human Information Sci. Labs., Kyoto 619-0288, Japan, ktajima@atr.co.jp), and K. G. Munhall (Dept. of Psych., Dept. of Otolaryngol., Queen's Univ., Kingston, ON K7L 3N6, Canada)

Japanese distinguishes between words by the presence or absence of several types of mora phonemes, often realized as a contrast in segment duration, e.g., /hato/ (pigeon) versus /hat:o/ (hat). Several studies suggest that such contrasts are difficult for English listeners to perceive [e.g., R. Oguma, in *Japanese-Language Education Around the Globe*, 2000, Vol. 10, pp. 43–55]. In this study we investigated the effect on the perceptual ability of both Japanese exposure and perceptual identification training. Three groups of subjects were tested: (1) English speakers with no Japanese experience; (2) English speakers who had spent 1–6 months in Japan; and (3) native Japanese speakers. Subjects participated in a forced-choice identification task in which they heard words and nonwords produced by Japanese speakers and identified which word they heard by choosing among items that minimally differed with respect to these contrasts. Additionally, subjects in the second group underwent five days of perceptual training during which they received immediate feedback, repeating trials until they responded correctly. Results suggest that although the overall performance was relatively high, identification accuracy improved with exposure to Japanese and with perceptual identification training. Implications of this work on theories of second-language learning will be discussed. [Work supported by TAO, Japan.]

**4pSC20. Learning stop-consonant voicing in English: The effect of perceived cross-language similarity.** Wendy Baker (Dept. of Linguist., Brigham Young Univ., Provo, UT 84602) and Pavel Trofimovich (Dept. of Educational Psych., Univ. of Illinois at Urbana–Champaign, Urbana, IL 61801)

Previous research has demonstrated that perceived similarity between native- (L1) and second-language (L2) sounds overall influences how accurately L2 sounds are produced. No study, however, has examined whether certain phonetic cues to an accurate production of L2 sounds are susceptible to the influence of perceived similarity between L1 and L2 sounds more than others. In English, vowel duration is the primary cue to distinguishing word-final voiced and voiceless stops (e.g., bad versus bat). A secondary cue to stop-consonant voicing is the duration of the stop-gap closure of word-final voiced and voiceless stops. In this study we examined how perceived similarity between L1 and L2 sounds influences adult L2 learners' production of these two cues to stop-consonant voicing. Thus specifically examined were vowel and stop-gap closure duration in beginning, intermediate, and advanced native Korean adult learners' productions of six English vowels in 18 CVt and CVd words. The results of these acoustic analyses indicated that: (1) these two cues to stop-consonant voicing are differentially susceptible to the influence of perceived similarity between L1 and L2 sounds; and (2) perceived similarity between L1 and L2 sounds influences how accurately and quickly these cues are learned.

**4pSC21. Short-term immersion can change the location of a phonetic category boundary.** Anders Damgren-Hojen (Dept. of English, Univ. of Aarhus, Jens Chr. Skous Vej 7, DK-8000 Aarhus, Denmark)

In this study the effect of immersion in a second-language (L2) speaking environment on the perception and production of English sibilants by native (L1) speakers of Danish is examined. Three groups of subjects participated. Fourteen Danish au pairs or exchange students were tested before and after a 6–12 month immersion period. Eleven Danish control subjects were tested twice without immersion. Five native English baseline subjects were tested once. Danes tend to assimilate English /s/–/sh/ to two different Danish sound categories. However, the Danish /sh/ is palatalized and has more high frequency energy than English /sh/. If the immersed subjects changed their mental specifications for /sh/ after massive input of the more low frequency English /sh/ during immersion, one might expect a perceptual category boundary shift for /s/–/sh/. This shift was indeed found, using an interpolated synthetic /s/–/sh/ continuum for the identification and discrimination in both an English and a Danish language set. Production was measured using spectral moments. However, there was no clear indication of a more authentic English pronunciation of /sh/ after immersion than before. The results pertain to questions regarding the re-organization of phonetic categories after L2 experience, interactions between L1 and L2 phonetic systems, and the relation between perception and production.

**4pSC22. Perception of American English glide consonants by Danish listeners.** Catherine T. Best (Dept. of Psych., Wesleyan Univ., Middletown, CT 06459, cbest@wesleyan.edu) and Ocke-Schwen Bohn (Aarhus Univ., DK-8000 Aarhus C, Denmark)

Previous research has found both phonological and phonetic influences from the native language on perception of American English (AE) /r–l/, /w–r/, /w–j/ by Japanese and French speakers. This study examined native language effects on perception of the same stimuli by Danish listeners with minimal English experience. Danish contrasts /r–l/, but realizes them differently than AE, as pharyngeal versus nonvelarized lateral approximants. It has /j/ but lacks /w/; however, /v/ is realized as an unrounded labiodental approximant similar to /w/. Thus, on purely phonological grounds, Danes should have greater difficulty with AE /w–j/ and /w–r/ than /r–l/. On phonetic grounds, however, all three contrasts should be assimilated to Danish contrasts, and therefore categorized and discriminated quite well. Forced-choice identification was highly categorical for all three contrasts. However, A×B tests revealed a native-like discrimination peak only for /w–r/; /r–l/ showed a lower and less well-defined peak, whereas /w–j/ was discriminated at ceiling except for the /j/ end of the continuum. These results are consistent with previous evidence that both phonological and phonetic properties of the native language affect non-native speech perception. The findings will be compared to prior cross-language research, and discussed in light of phonological differences among the listeners' languages.

**4pSC23. Does one plus one equal three in early simultaneous bilingual speech perception?** Megha Sundara (School of Commun. Sci. & Disord., McGill Univ., 1266 Pine Ave. W., Montreal, QC H3G 1A8, Canada, msunda@po-box.mcgill.ca)

Over half the world's population is bilingual, yet we know little about organization of phonetic abilities in early simultaneous bilinguals. In this study we tested bilingual French–English adults who had learned and used both languages simultaneously since birth. They were presented dental and alveolar stop consonants excised from real-word productions of French /t/ and English /d/, selected to insure that VOT values overlap. The alveolar-dental place distinction is not phonemic in either French or English. However, bilingual individuals are systematically exposed to this place distinction across their two native languages. Recent findings [Sundara and Polka, *J. Acoust. Soc. Am.* **110**, 2685 (2001)] indicate that simultaneous bilinguals clearly produce this place distinction. Perception was assessed using a categorical AXB task with tokens produced by multiple talkers in both French and English. Assimilation data were also obtained using a keyword identification task in both languages. Performance of bilinguals was compared and contrasted with those of respective monolingual groups and with native Malayalam listeners (dental-alveolar distinction is phonemic in Malayalam). The findings provide insights into perceptual organization in simultaneous bilingual adults by addressing whether an emergent contrast, not evident in either monolingual group is observed in both perception and production of bilinguals.

**4pSC24. Age of immersion as a predictor of foreign accent.** Miles Munro and Virginia Mann (3151 SSPA, Univ. of California, Irvine, CA 92697-5100, mmunro@uci.edu)

This study examined the relationship between Age of Immersion (AOI) and the Degree of Phonological Accent (DPA) that native-English-speaking judges perceived in the speech of Mandarin speakers who learned English as a second language. AOI and speech samples of variable length and linguistic context (single words, sentences, a short paragraph, and a self-generated picture narration) were collected from the target group (n=32, AOI=3–16) and from native speaker controls (n=4, AOI=0). These were judged by a native speaker panel (n=14) using a continuous scale, along with measurements of confidence. Judging was broken over 3 separate visits. A set of similar samples and speakers was repeated at the onset and termination of the main rating task of each visit to ensure reliability. No single AOI was found for a “Critical Period” (CP). Instead, DPA was found to increase in a highly linear manner with AOI. In general, ratings of DPA began to increase at about AOI=5, maxing out as early as AOI=7, although near-native ratings were given to some speakers whose AOI was greater than 5, especially in the case of female speakers. Thus DPA can be linearly predicted with AOI, though individual speakers are fairly variable after AOI=5.

4:40–5:10

Discussion

Winifred Strange, Discussion Leader

**Session 4pSP****Signal Processing in Acoustics and Physical Acoustics: Seismic Signal Processing: Detection, Estimation and Inversion Methods**

Alan W. Meyer, Cochair

*Lawrence Livermore National Laboratory, L-154, 7000 East Avenue, Livermore, California 94550*

Moyses Zindeluk, Cochair

*Mechanical Engineering Department, COPPE/University of Rio de Janeiro, Caixa Postal 68 503, Rio de Janeiro 21945-000, Brazil***Invited Papers****1:00****4pSP1. Deconvolution and signal extraction in geophysics and acoustics.** Leon H. Sibul, Michael J. Roan, and Josh Erling (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804-0030)

Deconvolution and signal extraction are fundamental signal processing techniques in geophysics and acoustics. An introductory overview of the standard second-order methods and minimum entropy deconvolution is presented. Limitations of the second-order methods are discussed and the need for more general methods is established. The minimum entropy deconvolution (MED), as proposed by Wiggins in 1977, is a technique for the deconvolution of seismic signals that overcomes limitations of the second-order method of deconvolution. The unifying conceptual framework MED, as presented in the Donoho's classical paper (1981) is discussed. The basic assumption of MED is that input signals to the forward filter are independent, identically distributed non-Gaussian random processes. A forward convolution filter "makes" the output of the forward filter more Gaussian which increases its entropy. The minimization of entropy restores the original non-Gaussian input. We also give an overview of recent developments in blind deconvolution (BDC), blind source separation (BSS), and blind signal extraction (BSE). The recent research in these areas uses information theoretic (IT) criteria (entropy, mutual information, K-L divergence, etc.) for optimization objective functions. Gradients of these objective functions are nonlinear functions, resulting in nonlinear algorithms. Some of the recursive algorithms for nonlinear optimization are reviewed.

**1:30****4pSP2. Earth imaging using seismic migration.** Michael Fehler, Lianjie Huang, Douglas Alde, and Hongchaun Sun (Los Alamos Natl. Lab., MS D443, Los Alamos, NM 87545)

Most subsurface imaging of the Earth is accomplished using seismic migration, which uses the entire wave forms recorded at the surface. The traditional method for performing migration was to use the Kirchhoff integral using a ray-based Green function to backproject the recorded data to positions of reflectors within the subsurface. Recent developments in Kirchhoff migration include the use of multivalued traveltimes tables calculated with methods like wave-front construction ray tracing. Additional advances include the use of ray amplitudes calculated using dynamic ray tracing. Recently, more attention has been given to the development of wave-equation-based methods in which the recorded wave field is backprojected using some approximation to the wave equation. Wave-equation-based methods have grown in acceptance because better implementation of the backprojection operators have been developed and computing capability has allowed the generally slower methods to become more feasible.

**1:50****4pSP3. 3-D tomographic imaging of the geologic structure in the Salton Sea Geothermal Field.** Alan W. Meyer, Lawrence J. Hutchings, and Paul W. Kasameyer (Lawrence Livermore Natl. Lab., P.O. Box 808, Livermore, CA 94551-0808)

A three-dimensional tomographic reconstruction of the Salton Sea Geothermal Field is presented. This reconstruction is developed from data gathered in the course of one year between 15 September 1987 and 30 September 1988 using a microearthquake network. This geothermal field is important not only due to understanding potential energy sources but also because it is the result of a tectonic spreading zone bounded between two transverse fault systems: The San Andres system to the North and the Brawley fracture zone (BFZ) to the South. Here magma has penetrated into the crust to a depth of at least 8 km. This magmatic source is responsible for the microearthquakes generated along the BFZ as well as providing the thermal source for the geothermal activity. Using techniques for both blind source estimation as well as blind deconvolution, a travel time tomographic algorithm is applied to these data. The objective is to characterize the subterranean geological structure and estimate the fracturing that supports the geothermal field. These studies are a necessary foundation for future research into the energy capacity of this field. [Work performed under the auspices of the Department of Energy by the Lawrence Livermore National Laboratory under Contract No. W-7405-Eng-48.]

**4pSP4. Identification of a reversed-phase compressional in dipole borehole logging. Physical principle, theoretical modeling, and field examples.** Lucio N. Tello (Computalog Research, Inc., 500 Winscott Rd., Fort Worth, TX 76126) and Marek Kozak (Magnetic Pulse, Inc., Fremont, CA 94538)

Dipole array sonic tools are particularly suited for estimating acoustic shear wave velocities in soft, unconsolidated, slow formations in boreholes. Dipole sources induce flexural waves, which are sustained in such formations. By contrast, monopole sources create pseudo-Rayleigh waves, related to shear wave velocity, which, in slow formations, refract away from the borehole and vanish before reaching the receivers. In addition, monopole sources create compressional and Stoneley waves. The typical order of these arrivals is compressional, shear (when present), and Stoneley. In an ideal situation dipole sources excite only borehole-flexural waves. The propagation velocity of the flexural wave is near that of the shear wave. Field experience backed by theoretical modeling indicates that in real situations dipole sources might also generate (reversed-phase) compressional and Stoneley waves. Theoretical simulations using well parameters taken from other logs render nearly identical wavetrains. Verification of the results with seismic measurements further validates the processing.

**4pSP5. Transmission mode acoustic time-reversal imaging for nondestructive evaluation.** Sean K. Lehman (L-154, Lawrence Livermore Natl. Lab., 7000 East Ave., Livermore, CA 94550) and Anthony J. Devaney (Northeastern Univ., Boston, MA 02115)

In previous ASA meetings and JASA papers, the extended and formalized theory of transmission mode time reversal in which the transceivers are noncoincident was presented. When combined with the subspace concepts of a generalized Multiple Signal Classification (MUSIC) algorithm, this theory is used to form super-resolution images of scatterers buried in a medium. These techniques are now applied to ultrasonic nondestructive evaluation (NDE) of parts, and shallow subsurface seismic imaging. Results are presented of NDE experiments on metal and epoxy blocks using data collected from an adaptive ultrasonic array, that is, a "time-reversal machine," at Lawrence Livermore National Laboratory. Also presented are the results of seismo-acoustic subsurface probing of buried hazardous waste pits at the Idaho National Engineering and Environmental Laboratory. [Work performed under the auspices of the U.S. Department of Energy by University of California Lawrence Livermore National Laboratory under Contract No. W-7405-Eng-48.] [Work supported in part by CenSSIS, the Center for Subsurface Sensing and Imaging Systems, under the Engineering Research Centers Program of the NSF (award number EEC-9986821) as well as from Air Force Contracts No. F41624-99-D6002 and No. F49620-99-C0013.]

**4pSP6. Shallow subsurface applications of high-resolution seismic reflection.** Don Steeples (Dept. of Geology, The Univ. of Kansas)

Shallow seismic reflection surveys have been applied to a wide variety of problems. For example, in many geologic settings, variations and discontinuities on the surface of bedrock can influence the transport and eventual fate of contaminants introduced at or near the ground surface. Using seismic methods to determine the nature and location of anomalous bedrock can be an essential component of hydrologic characterization. Shallow seismic surveys can also be used to detect earthquake faults and to image underground voids. During the early 1980s, the advent of digital engineering seismographs designed for shallow, high-resolution surveying spurred significant improvements in engineering and environmental reflection seismology. Commonly, shallow seismic reflection methods are used in conjunction with other geophysical and geological methods, supported by a well-planned drilling-verification effort. To the extent that seismic reflection, refraction, and surface-wave methods can constrain shallow stratigraphy, geologic structure, engineering properties, and relative permeability, these methods are useful in civil-engineering applications and in characterizing environmental sites. Case histories from Kansas, California, and Texas illustrate how seismic reflection can be used to map bedrock beneath alluvium at hazardous waste sites, detect abandoned coal mines, follow the top of the saturated zone during an alluvial aquifer pumping test, and map shallow faults that serve as contaminant flowpaths.

**4pSP7. Bayesian probability analysis for acoustic–seismic landmine detection.** Ning Xiang (Natl. Ctr. for Phys. Acoust. and Dept. of Elec. Eng., Univ. of Mississippi, University, MS 38677, nxiang@olemiss.edu), James M. Sabatier, and Paul M. Goggans (Univ. of Mississippi)

Landmines buried in the subsurface induce distinct changes in the seismic vibration of the ground surface when an acoustic source insonifies the ground. A scanning laser Doppler vibrometer (SLDV) senses the acoustically-induced seismic vibration of the ground surface in a noncontact, remote manner. The SLDV-based acoustic-to-seismic coupling technology exhibits significant advantages over conventional sensors due to its capability for detecting both metal and nonmetal mines and its stand-off distance. The seismic vibration data scanned from the SLDV are preprocessed to form images. The detection of landmines relies primarily on an analysis



of the target amplitude, size, shape, and frequency range. A parametric model has been established [Xiang and Sabatier, *J. Acoust. Soc. Am.* **110**, 2740 (2001)] to describe the amplified surface vibration velocity induced by buried landmines within an appropriate frequency range. This model incorporates vibrational amplitude, size, position of landmines, and the background amplitude into a model-based analysis process in which Bayesian target detection and parameter estimation have been applied. Based on recent field measurement results, the landmine detection procedure within a Bayesian framework will be discussed. [Work supported by the United States Army Communications–Electronics Command, Night Vision and Electronic Sensors Directorate.]

3:45

**4pSP8. Modeling heavy-tailed correlated clutter via wavelet packet bases with application to seismic event estimation.** Paul J. Gendron (Naval Res. Lab., Acoust. Div., 4555 Overlook Ave. SW, Washington, DC 20375) and Balgobin Nandram (Worcester Polytechnic Inst., Worcester, MA 01609)

A model for correlated and heavy-tailed background noise in time series data is introduced. The approach is based on a multi-resolution time adaptive wavelet packet expansion that assigns to each resolution and band a measure of spikiness associated with independent  $t$  distributed variates. The model is adaptive similarly in time, scale, and correlation structure within the dyadic structure of the wavelet packet framework. Parameter estimation is accomplished by recursions in time at each frequency band/scale. The model is applied to both ground motion data and shallow water ocean acoustic background noise. A comparison with the Gaussian model suggest its favorable advantage. Estimators for transient signals based on the new model are derived and show usefulness in extracting transients with unknown or weakly specified characteristics. The usefulness of the model and the estimator is demonstrated via a comparison with the fixed transform DWT on quarry blast seisms. The results show that the method outperforms DWT based estimation in MSE by a factor of 2 at a moderate SNR for these classes of transients. [Work supported by ONR.]

### Contributed Papers

4:05

**4pSP9. Use of the Bayesian Cramér–Rao bound in seismic exploration and research planning.** Max Deffenbaugh (ExxonMobil Upstream Res. Co., P.O. Box 2189, Houston, TX 77252, max.deffenbaugh@exxonmobil.com)

Modern exploration for oil is heavily dependent on seismic data, which is used to detect and estimate properties of subsurface reservoirs. Seismic data is processed by applying a sequence of algorithms to filter, estimate velocity fields, beamform, and invert for reservoir properties. There are multiple algorithms to choose from at each processing step—ranging from simple and inexpensive to complex and costly. The Bayesian Cramér–Rao bound, combined with a statistical model of the signal and noise in seismic data, can be used to predict the information content of a specific data set at each stage in a proposed processing flow. This allows the exploration manager to identify the least-cost processing flow to achieve a certain information goal and to identify when the goal is unachievable at any cost. It can also guide research efforts to those processing steps where the greatest amount of information is being lost.

4:20

**4pSP10. Correlation vibroacoustic imaging of subsurface objects.** Alexander Ekimov, Nikolay Sedunov, Michael Tsionskiy, and Dimitri Donskoy (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030)

Any subsurface object whose mechanical impedance is different from the surrounding media manifests itself on the surface by contributing to the surface mechanical impedance. Since most of the surfaces (except plates and membranes) can be characterized by the local impedance, such a manifestation can be interpreted as an object image projected on the surface. This image projection can be captured by measuring surfaces local impedance at a grid of points or pixels. One of the problems in implementing this imaging approach is that object impedance contribution can vary substantially depending on the frequency, depth, and medium properties. Therefore, multiple impedance images must be obtained at the different frequencies and, consequently, an image selection or more so-

plicated image processing should be implemented at the expense of additional time and processing resources. The developed correlation technique utilizes broadband signals and performs convolution of the broadcasted signal with a measured vibration signal at each pixel. The resulted image has a high contrast and takes a fraction of time and processing power to create as compared with the image/frequency selection approach. Examples of practical implementation of the developed method are presented and discussed.

4:35

**4pSP11. Field site evaluation for seismic mine detection.** James S. Martin, Gregg D. Larson, Peter H. Rogers (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405), Waymond R. Scott, Jr. (School of Elec. and Computer Eng., Georgia Inst. of Technol., Atlanta, GA 30332), and George S. McCallIII (Georgia Inst. of Technol., Atlanta, GA 30332)

A system has been developed that uses audio-frequency surface seismic waves for the detection and imaging of buried landmines. The system is based on the measurement of seismic displacements immediately above buried mines using noncontacting vibrometers that interrogate the surface motion with either radar or ultrasonic signals. In laboratory tests and limited field tests the system has demonstrated the ability to detect a variety of inert antipersonnel and antitank mines with background contrast in excess of 20 dB. Current work on the system is focused on the transition from the laboratory into the field. To facilitate this, a series of experiments has been undertaken to measure the characteristics of several field test sites. The tradeoff between image contrast and scanning speed is of primary concern in evaluating the features of these sites. The field experiments have investigated the nature of ambient seismic noise, input impedance at the seismic source (a ground contacting shaker), modal content of the seismic interrogation signal, and the nature of the nonlinearities in the soil. Observed nonlinear phenomena have included harmonic generation, phase speed slowing, dispersion and spall. Although interesting, the differences between the field sites and the laboratory model do not appear to pose problems for seismic mine detection.

## Session 4pUW

## Underwater Acoustics and Acoustical Oceanography: David Weston Memorial Session II

Michael J. Buckingham, Cochair

*Scripps Institution of Oceanography, La Jolla, California 92093-0213*

Chester M. McKinney, Cochair

*4305 Farhills Drive, Austin, Texas 78731*

## Contributed Papers

1:00

**4pUW1. The feedback phenomenon applied to underwater acoustics.** Philippe Roux, Jason E. Jordan, and W. A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238)

People are familiar with the feedback phenomenon that results in the loud sound heard when a musician plays an electric instrument directly into a speaker. Feedback occurs when a source and a receiver are connected both acoustically through the propagation medium and electrically through an amplifier in such way that the received signal is simultaneously and continuously added to the emitted signal. A resonance is then obtained when the emitter and the receiver are in phase. The resonant frequency appears to be highly sensitive to fluctuations of the propagation medium. The feedback phenomenon has been experimentally demonstrated as a means to monitor the temperature fluctuation of a shallow water environment [“Acoustic monitoring of the sea medium variability: experimental testing of new methods,” by A. V. Furduev, *Acoust. Phys.* **47**, No. 3, 361–368 (2001)]. The goal of our work is to reproduce the feedback experiment using an alternative method that decomposes the feedback phenomenon into an iterative process. Successful reproduction of the feedback is accomplished using a step-by-step algorithm which details the evolution of the system from the initial signal to its steady-state form. These experimental and numerical results illustrate the potential of the feedback process for use in narrow-band acoustical tomography.

1:15

**4pUW2. System-orthogonal functions for sound velocity profile perturbation.** Wen Xu and Henrik Schmidt (MIT, 77 Massachusetts Ave., Rm. 5-204, Cambridge, MA 02139, wenuxu@mit.edu.)

Empirical orthogonal functions (EOFs) are derived from direct measurements of the sound velocity profile (SVP) and they are orthogonal in regard to the statistics of the SVP uncertainty. Viewed from the sonar output end, however, the effect of an error in one EOF is usually coupled with the effect of the error in another due to the nonlinear physical constraints. Thus the traditional EOF is not an optimum basis set to characterize the SVP uncertainty in regard to sonar performance measure. To obtain a system-orthogonal SVP representation, a new set of orthonormal functions is derived for SVP perturbations in a water column. The performance measure used is the Cramer–Rao bound (CRB) for SVP expansion coefficients, and a full-field random Gaussian signal model is assumed. The derived functions make the CRB matrix diagonal, decoupling the errors in the estimation of the expansion coefficients. Compared to the traditional EOFs, the new set of SVP basis functions depends on both the statistics of the sound speed uncertainty and the sound waveguide propagation property. It also includes the noise effect in measurements. Using this system-orthogonal function set makes possible the investigation of the relative significance of the individual basis functions. [Work supported by ONR.]

1:30

**4pUW3. The partial wave expansion for scattering in oceanic wave guides.** Dalcio K. Dacol (Acoust. Div., Naval Res. Lab., Washington, DC 20375-5350)

The partial wave expansion for target scattering in oceanic wave guides is obtained via the pseudo-potential method and discussed in detail. The case of sound scattering by multiple objects in an oceanic waveguide will also be discussed. The Fermi pseudo-potential was introduced in quantum mechanics as a means of simplifying problems involving scattering by a multicentered potential including applications to many-body problems. As such, it has direct applications in acoustics. It can also be used to discuss scattering off targets in confining geometries such as the interior of cavities or in wave guides. [Work supported by ONR.]

1:45

**4pUW4. A study of the connection between tidal velocities, soliton packets and acoustic signal losses.** Stanley A. Chin-Bing, Alex C. Warn-Varnas, David B. King (Naval Res. Lab., Stennis Space Center, MS 39529-5004, chinbing@nrlssc.navy.mil), Kevin G. Lamb (Univ. of Waterloo, Waterloo, ON N2L 3G1, Canada), James A. Hawkins (Planning Systems, Inc., Slidell, LA 70458), and Marvi Teixeira (Polytechnic Univ. of Puerto Rico, Hato Rey, PR 00988)

Coupled ocean model and acoustic model simulations of soliton packets in the Yellow Sea have indicated that the environmental conditions necessary for anomalous signal losses can occur several times in a 24 h period. These conditions and the subsequent signal losses were observed in simulations made over an 80 h space–time evolution of soliton packets that were generated by a 0.7 m/s tidal velocity [Chin-Bing *et al.*, *J. Acoust. Soc. Am.* **111**, 2459 (2002)]. This particular tidal velocity was used to initiate the Lamb soliton model because the soliton packets that were generated compared favorably with SAR measurements of soliton packets in the Yellow Sea. The tidal velocities in this region can range from 0.3 m/s to 1.2 m/s. In this work we extend our simulations and analyses to include soliton packets generated by other tidal velocities in the 0.3–1.2 m/s band. Anomalous signal losses are again observed. Examples will be shown that illustrate the connections between the tidal velocities, the soliton packets that are generated by these tidal velocities, and the signal losses that can occur when acoustic signals are propagated through these soliton packets. [Work supported by ONR/NRL and by a High Performance Computing DoD grant.]

2:00

**4pUW5. Seismo-acoustic propagation in environments that depend strongly on both range and depth.** Donald A. Outing, William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180), LeRoy M. Dorman (Scripps Inst. of Oceanogr., La Jolla, CA 92093), and Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

The parabolic equation method provides an excellent combination of accuracy and efficiency for range-dependent ocean acoustics and seismology problems. This approach is highly developed for problems in which the ocean bottom can be modeled as a fluid. For the elastic case, there

remain some accuracy limitations for problems involving sloping interfaces. Progress on this problem has been made by combining a new formulation of the elastic parabolic equation that handles layering more effectively [W. Jerzak, "Parabolic Equations for Layered Elastic Media," doctoral dissertation, Rensselaer Polytechnic Institute, Troy, NY (2001)] and a mapping approach that handles sloping interfaces accurately [J. Acoust. Soc. Am. **107**, 1937–1942 (2000)]. This approach makes it possible to handle problems involving complex layering and steep slopes, but the rate of change of the slope must be small. The method and its application to data will be described. Our immediate goal is to model propagation of seismic surface waves propagating across a transition between dry and marshy terrain. We have suitable data applicable to vehicle-tracking problems from Marine Corps Base Camp, Pendleton, CA. [Work supported by ONR.]

2:15

**4pUW6. The split-step Padé–Fourier solution.** Elizabeth Küsel, Flor A. Espinoza, William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180), and Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

At lower frequencies, the split-step Padé solution is at least comparable in efficiency to the split-step Fourier solution [J. Acoust. Soc. Am. **100**, 178–182 (1996)]. For deep-water problems in the kilohertz regime, the split-step Fourier solution provides greater efficiency but is not as accurate as the split-step Padé solution. The split-step Padé–Fourier solution combines the attributes of the two split-step solutions to achieve greater efficiency for some problems without sacrificing accuracy. This approach is based on factoring the exact exponential operator by multiplying and dividing by the approximate exponential that is used in the split-step Fourier solution. The multiplicative factor is implemented using the classical algorithm based on the fast Fourier transform. The other two factors are handled together with a rational approximation. Since the oscillation rate of these two factors is much lower than for the exact exponential, an efficient solution can be obtained using a very large range step and a relatively small number of terms in the rational approximation. [Work supported by ONR.]

2:30

**4pUW7. Internal solitary waves and coupled perturbed modes.** Chris J. Higham and Chris T. Tindle (Phys. Dept., Univ. of Auckland, Private Bag 92019, Auckland, New Zealand)

Coupled perturbed mode theory combines conventional coupled modes and perturbation theory for fast range-dependent normal mode calculations. The method is applied to sound propagation through shallow water internal solitary waves, or solitons. The solitons considered are depressions of the thermocline, separating well-mixed upper and lower layers. Shallow water solitons lead to strong mode coupling. The theory gives physical insight into soliton coupling physics through explicit mode coupling formulas.

2:45

**4pUW8. Coupled mode propagation in an elastic oceanic wedge.** Ahmad T. Abawi (SPAWAR Systems Ctr., 53560 Hull St., San Diego, CA 92152-5001)

The elastic version of the one-way coupled mode propagation model [Abawi, J. Acoust. Soc. Am. **111**, 160–167 (2002)] is used to compute the propagation of waves in an ocean overlaying a shear-supporting wedge-shaped bottom. The range-dependent ocean is approximated by a set of stair-step elastic waveguides. The elastic modes are obtained from the solution of the equations of motion in each stair-step and the solution of the range-dependent problem is obtained by solving a set of coupled differential equations for the mode amplitudes as a function of range. Various

field quantities such as the scalar and shear potentials, the compressional and shear pressures, and the displacements are computed and the results are compared with those obtained from the fast field propagation model, OASES.

3:00–3:15 Break

3:15

**4pUW9. Influence of shallow water currents on the performance of a broadband time-reversing array.** Karim G. Sabra and David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, 1231 Beal Ave., 2019 Lay Autolab., Ann Arbor, MI 48109-2121)

In recent shallow water experiments, the temporal and spatial focusing properties of time-reversing arrays (TRAs) were shown to be robust in a reciprocal medium and useful for underwater applications. The presence of oceanic currents in coastal environments leads to nonreciprocal acoustic propagation. In this case, time-reversal invariance is modified because the propagation speed inhomogeneity depends on the direction of acoustic propagation. Therefore, similarly to phase coherent reciprocal transmissions, a TRA will be influenced by the current-induced effects but not by the scalar contributions due to temperature or salinity. TRA performance, in the presence of steady currents, is investigated both theoretically using a simple first order normal mode formulation and numerically using a parabolic equation code for moving media, GCPEM [D. Mikhin, J. Acoust. Soc. Am. **105**, 1362 (1999)]. In a multipath shallow ocean environment, the retrofocus field is shifted relative to its location in a nonmoving medium. This shift depends on the current speed and the range-depth dependency of the ocean current profile because each acoustic mode is influenced differently. The possibility of using TRAs for monitoring coastal currents will be discussed.

3:30

**4pUW10. A range-dependent propagation model based on a combination of ray theory and plane-wave reflection coefficients.** Jens M. Hovem (Norwegian Univ. of Science and Technol., Trondheim, Norway) and D. P. Knobles (Univ. of Texas, Austin, TX 78713)

The paper describes a range-dependent propagation model based on a combination of range-dependent ray tracing and plane-wave bottom responses. The ray-tracing module of the model determines all the eigenrays between any source/receiver pairs and stores the ray histories. The received wave field is then synthesized by adding the contributions of all the eigenrays, taking into account the reflections from the bottom and the surface. The model can treat arbitrarily varying bottom topography and a layered elastic bottom as long as the layers are parallel. In the current version, the bottom is modeled with a sedimentary layer over an elastic half space, but more complicated structures are easily implemented. The new model has been tested against other models on several benchmark problems and also applied in the analysis and modeling of up-slope and down-slope propagation data recorded on a 52-element center-tapered array that was deployed at two locations about 70 miles east of Jacksonville, FL. The paper presents the results of these tests with an assessment of the potential use in connection with geo-acoustic inversion of range-dependent and elastic scenarios. [Work supported by Applied Research Laboratories, The University of Texas.]

3:45

**4pUW11. Interference patterns in range dependent environments.** Henry Cox and Kevin D. Heaney (ORINCON Industries, 4350 N. Fairfax Dr., Ste. 470, Arlington, VA 22203)

Interference patterns in space and frequency occur in ocean acoustics due to coherent multipath propagation. The wave guide invariant beta is commonly used to describe the behavior of these patterns [Brikhosvsky and Lysanov, *Fundamentals of Ocean Acoustics*, 2nd ed. (Springer-Verlag, New York, 1991)]. The standard derivation of this invariant uses a normal

mode expansion and is limited to range independent and weakly range dependent environments. A different approach is introduced that describes the interfering components in terms of their travel times. This leads to a very simple derivation of basic results that apply equally well to range dependent environments and to situations, such as Lloyd's Mirror, in which the normal mode representation is inconvenient. Beta can be expressed in terms of the local phase velocities and the difference in travel times for the interfering components. The travel time differences involve integration over the entire path trajectories from the source to the point where the interference pattern is observed. Numerical examples using ray theory will be presented.

4:00

**4pUW12. Evidence of three-dimensional waveguide propagation in SWARM 95 data.** Scott Frank, William L. Siegmann (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180), and Mohsen Badiy (Univ. of Delaware, Newark, DE 19716)

During a period of the SWARM95 experiment, strong nonlinear internal waves passed across two tracks that had airgun pulses propagating along them. Environmental data for this period indicate that the angles between the tracks and the internal wave-fronts, which were roughly planar, were very different—one angle being close to zero, and the other approximately 42 deg. Two-dimensional PE simulations for these waveguides show dramatically different results for depth-averaged, pulse-integrated energy variations. Specifically, the observed levels can be reproduced for the waveguide with the large incidence angle, but not for the one with the small incidence angle. For the latter case, data show significant variations in pulse shapes and in the integrated energy ( $\approx 5$  dB), while simulations show very small changes in both of these characteristics. Results from several recent computational and theoretical studies suggest that the cause may be three-dimensional effects from horizontal refraction and modal interference due to the nonlinear internal waves. The adiabatic mode parabolic equation [Collins, *J. Acoust. Soc. Am.* **94** (1993)] is used to quantify the three-dimensional influence of the internal waves on the integrated energy variations. The results demonstrate experimental evidence of three-dimensional effects from strong nonlinear internal waves. [Work supported by ONR.]

4:15

**4pUW13. Computing the two-point correlation function directly from the transport equation using split-step Padé solutions.** Chen-Fen Huang, Philippe Roux, and W. A. Kuperman (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0238, chenfen@mpl.ucsd.edu)

The two-point correlation function that describes the correlation between an acoustic field at two depths as a function of range obeys a transport equation. Without making a frequency or narrow angle approximation, this equation can be solved by parabolic equation methods. The solution algorithm is split into two depth-steps using the alternating direction implicit (ADI) method to advance one range-step. Then, at each depth substep, the split-step Padé solution [M. D. Collins, *J. Acoust. Soc. Am.* **96**, 382–385 (1994)] is used. Computations are confirmed against intensities computed from the indirect method that constructs the two-point function by an outer product of pressure vectors. Examples of the propagation of the vertical correlation for deep, shallow, and stochastic environments are presented.

4:30

**4pUW14. A study on simplification of the Biot–Stoll model in water-saturated porous medium.** Keunhwa Lee and Woojae Seong (Dept. of Ocean Eng., Seoul Natl. Univ., Seoul 151-742, Korea)

A practical approximate method based on the Biot–Stoll model to represent acoustic waves in the ocean in the presence of porous sediment is presented. The ocean sediment can be described individually by four models; visco-fluid, visco-elastic, equivalent fluid, and water-saturated po-

rous medium. The first two models are approximate forms of the Biot–Stoll model. The equivalent fluid model of Zhang and Tindle [*J. Acoust. Soc. Am.* **98**, 3391–3396 (1995)] derived for the fluid/solid interface is extended to the fluid/porous-medium interface for analytical derivation of the reflection coefficient. Then, the reflection characteristics as functions of frame stiffness ( $K_b/K_r$ ) and frequency region ( $f/f_c$ ) are discussed for all four models. In weak stiffness and high-frequency region, the equivalent fluid and visco-fluid bottom models describe ocean bottom better than the visco-elastic bottom model. Two-phase property is dominant in strong (high) stiffness or high-frequency region. This demonstrates that the visco-fluid and equivalent fluid models are useful approximate models in weak stiffness porous material such as the marine sediment.

4:45

**4pUW15. Asymptotic methods of evaluation of acoustic fields generated by underwater moving sources.** Vladimir S. Rabinovich (Instituto Politecnico Nacional de Mexico, ESIME-Zacatenco, edif. 1, Av. IPN, Mexico, D.F. 07738)

The problem of acoustic waves propagation from sources moving in stratified and almost-stratified waveguides simulating real oceanic waveguides is considered. It is assumed that acoustic sources move under water in the ocean with a subsonic variable velocity. Our approach for the evaluation of the acoustic field is based on the combination of two asymptotic methods: (1) The asymptotic analysis of the Green's function for almost-stratified waveguides by the operator ray method with respect to a small parameter which characterizes the variation of waveguides in the horizontal direction; (2) a subsequent asymptotic analysis of the integral representation of the field generated by a moving source by the two-dimensional stationary phase method with respect to a large parameter which characterizes the smallness of variations of the amplitude of source and the smallness of the vertical component of the source speed. Asymptotic formulas for the acoustic pressure, for the mode and time Doppler effects, are obtained. These formulas have a clear physical meaning and are convenient for the numerical simulation of the problem.

5:00

**4pUW16. Analysis of very high frequency propagation in sediments: Experimental results and modeling.** Richard Brothers, Sarah Page, Gary Heald (QinetiQ, Newton's Rd., Weymouth DT4 8UR, UK, rbrothers@qinetiq.com), Tim Leighton, Matt Simpson (ISVR, Southampton Univ., UK), and Justin Dix (SOC, Southampton Univ., UK)

A current QinetiQ study is investigating the propagation of sound waves into sediment at frequencies higher than 300 kHz. Previous work has found notable discrepancies between model predictions and experimental results and comparisons are inconsistent and unreliable. This new work package investigates the development of new scattering theories for frequencies ranging from 300 kHz to 1 MHz. In particular, the application of a pseudospectral time difference approach is analyzed. The model originally developed for lower frequency applications, is set up in various geometrical scenarios and for varying very high frequencies. Results show the received simulated pulses obtained for hydrophones placed within the sediment and source collocated. Furthermore, simulations are compared in tank experimental data. The controlled tank experiments were conducted by Southampton University and data are analyzed and discussed for various conditions and frequencies.



**Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics**

to be held jointly with the

**ANSI-Accredited U.S. Technical Advisory Group (TAG) Meeting for:  
ISO/TC 108/SC 4 Human Exposure to Mechanical Vibration and Shock**

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Human Exposure to Mechanical Vibration and Shock

*1325 Meadow Lane, Yellow Springs, Ohio 45387*

**Accredited Standards Committee S3 on Bioacoustics.** Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. There will be a report on the interface of S3 activities with those of ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock, including plans for future meetings of ISO/TC108/SC 4. The US Technical Advisory Group for TC 108/SC 4 consists of members of S3 and other persons not necessarily members of this Committee. Open discussion of committee reports is encouraged. *People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 8:00 a.m. on Thursday, 5 December 2002.*

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

**Meeting of Accredited Standards Committee (ASC) S1 Acoustics**

G. S. K. Wong, Chair S1

*Institute for National Measurement Standards, National Research Council, Montreal Road, Bldg. M36, Ottawa, Ontario K1A 0R6, Canada*

J. Seiler, Vice Chair S1

*U.S. Department of Labor, Mine Safety and Health Admin., P.O. Box 18233, Bldg. 38, Cochrans Mill Road, Pittsburgh, Pennsylvania 15236*

**Accredited Standards Committee S1 on Acoustics.** Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound, etc. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged. *People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 8:00 a.m. on Thursday, 5 December 2002.*

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