Session 2aAAa

Architectural Acoustics: Scattering Topics in Room Acoustics

Richard H. Campbell, Cochair

Bang-Campbell Associates, Box 47, Woods Hole, Massachusetts 02543

Carlos Alejandro Bidondo, Cochair

Freire 3766, Capital Federal, Buenos Aires CP 1429, Argentina

Chair's Introduction—8:00

Invited Paper

8:05

2aAAa1. The rediscovery of diffuse reflection in room acoustics prediction. Bengt-Inge Dalenbäck (CATT, Mariagatan 16A, SE-41471 Gothenburg, Sweden, bid@catt.se)

Around 1986 "hybrid methods" appeared where specular ray tracing was used to speed up image source validation and were implemented in a range of software (sometimes in the form of specular cone tracing). Unfortunately, these methods neglected diffuse reflection and room acoustics prediction suffered. Consequently, most software that started out with such methods have today either changed their algorithms to incorporate diffuse reflection or use them only to predict the early part of the echogram. A possible reason for the popularity of these methods was the more detailed point-receiver echogram, but already in 1980, Kuttruff published an *Acustica* paper clearly indicating the importance of handling diffuse reflection. Nevertheless, software exist today, and are presented in journals and doctoral theses, that neglect diffuse reflection. One reason could be that diffuse reflection does not *always* need to be taken into account since when rooms act *geometrically mixing* or have an *even absorption distribution* a specular-only prediction may suffice. However, if none of these conditions are met the RT may be severely overestimated while, on the other hand, Sabine/Eyring formulas will severely underestimate the RT. This paper presents examples where neglecting diffuse reflection leads to large prediction errors, especially regarding the RT.

Contributed Papers

8:25

2aAAa2. Perceptibility of double-slope reverberation decays. Derrick P. Knight (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854, dknight@jhacoustics.com), Yasushi Shimizu, and Rendell R. Torres (Rensselaer Polytechnic Inst., Troy, NY 12180)

No concert hall has a perfectly diffuse field, although many are close enough that their decay is perceived as linear. In recent years, concert hall acousticians have taken steps to ensure more exaggerated double-sloped (nonlinear) decays in their concert halls by using coupled volumes. Some acousticians feel that a coupled volume gives a hall a balance between clarity (subjectively speaking) and reverberance. However, there have been no studies done to determine when a nonlinear decay becomes perceptibly different from a linear decay. This work seeks to identify the threshold of perception for nonlinear decays. Nonlinear impulse responses of different lengths are generated by first computing uncoupled impulse responses of a concert hall and a coupled volume in CATT-Acoustic. The two linear impulse responses are convolved in MATLAB. These convolved impulse responses are manipulated to systematically vary the degree of nonlinear decay. The various nonlinear impulse responses are then convolved with anechoic signals with different temporal characteristics and presented to listeners for evaluation. From these evaluations, a criteria is derived to determine when a nonlinear decay becomes audibly different from a linear decay to a listener for various representative signals.

8:40

2aAAa3. Effects of size and density of diffusers on scattering coefficients measured in a 1:10 reverberation chamber. Jin Yong Jeon and Sung Chan Lee (School of Architectural Eng., Hanyang Univ., Seoul 133-791, Korea)

The degree of diffusion, or scattering coefficient, in surface materials has been known to be one of the most important aspects of the acoustical

qualities of concert halls. It has also been recognized that one of the best methods, which can reduce the errors in calculating the reverberation time and other acoustic parameters through computer modeling, is to calculate the scattering coefficient of surface materials. Based on the suggested ISO method, which measures the random-incidence scattering coefficient of surfaces in a diffuse field, the scattering coefficients of different size and density of wooden hemispheres were measured in a 1:10 reverberation room. As a result, the 17.5 cm hemisphere (real size) has the maximum scattering coefficient. It was also found that the scattering coefficient was growing higher when the coverage area increased from the center of the base plate and when the diffuse density increased until the density reached about 50%. Ceramic tiles designed by the calculations of scattering coefficients have been installed for the sidewalls of a 400-seat concert hall.

8:55

2aAAa4. Scattering and/or diffusing elements in a variety of recently completed music auditoria. Ronald L. McKay (McKay Conant Brook, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, rmckay@mcbinc.com)

Architectural elements which provide effective acoustic scattering and/or diffusion in a variety of recently completed auditoria for music performance will be presented. Color slides depicting the various elements will be shown. Each will be discussed with respect to its acoustic performance and architectural logic. Measured time-energy reflection patterns will be presented in many cases.

2a TUE. AM

Invited Paper

9:10

2aAAa5. Room acoustic computer modeling: The effects of scattering on transient power flow. Richard H. Campbell (Elec. and Computer Eng., Worcester Polytechnic Inst., 100 Institute Rd., Worcester, MA 01609)

The CATT© room acoustic modeling program allows the computation of acoustic power projected to three mutualy perpendicular planes at a point in space integrated over a specified time period. By selecting successive time periods it is possible to observe the oscillatory nature of the transient power flow from first arrival to the long-term settled value during diffuse-field reverberant decay. The computation is normalized, and displayed as a fraction of the total power. Introducing scattering on the room surfaces results in a significantly smaller peak amplitude of power flow oscillation. Several cases are shown involving different room sizes and shapes and different orientations of source and receiver. Conclusions are made on the number of mean-free path intervals required to ''settle'' the computer-modeled acoustic field.

Contributed Papers

9:30

2aAAa6. Midfrequency modeling of coupled-room performance spaces by computation of transmission coefficients of apertures and arrays of apertures. Jason E. Summers, Rendell R. Torres, and Yasushi Shimizu (Prog. in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180)

The shapes of the energy decays in a coupled-room system are functions of the energy transmitted by the coupling apertures relative to the irradiation strengths on the boundary surfaces of the rooms. At lower frequencies, the effective coupling areas depart from the geometrical aperture areas as the transmission coefficients of the apertures depart from unity. In order to better predict energy decays in coupled-room systems, transmission coefficients of apertures and arrays of apertures are computed. Power transmission coefficients are calculated for a number of source positions and numerically integrated to yield a random-incidence transmission coefficient. Computed aperture transmission coefficients are compared with analytical frequency-domain solutions for the transmission by circular and rectangular apertures. High-frequency statistical and geometrical models are modified to be applicable at midfrequencies by adjusting the coupling areas in the models according to their computed transmission coefficients. From these models the significance of non-geometrical aperture transmission on the reverberant decays in coupled-room systems is evaluated. Finally, the validity and low-frequency limits of these midfrequency models are evaluated by comparison with measurements of energy decay in actual coupled-room systems. [Research supported by the Bass Foundation.]

9:45

2aAAa7. Scattering from faceted surfaces in optimized room acoustics computations. Rendell R. Torres (Prog. in Architectural Acoust., Rensselaer Polytech. Inst., 110 8th St., Troy, NY 12180-3590), U. Peter Svensson (Norwegian Univ. of Sci. and Technol. (NTNU), Trondheim, Norway), and Nicolas de Rycker (Rensselaer Polytech. Inst., Troy, NY 12180-3590)

To minimize the computational demands of including scattering in auralization, it is appropriate to study how many orders of scattering are necessary. For this purpose, studying edge diffraction is especially appropriate as an elementary form of surface scattering. In a previous study [Torres *et al.,* J. Acoust. Soc. Am. **109**, 600–610 (2001)], it was found that higher orders and combinations of edge diffraction components were not usually as significant as first-order diffraction components. The primary reason was that the reference geometry (a large concert-hall stagehouse) was conservatively composed of large flat walls with dimensions larger

than most wavelengths of interest. In that case, significant edge-diffractions occurred at relatively low frequencies (below about 150 Hz). Other realistic reflecting surfaces in rooms, however, also include smaller-scale surface irregularities, e.g., facets for which higher-frequency wavelengths are typically a similar order or larger. This study examines a smaller test geometry consisting of reflector panel arrays similar to those found in concert halls, and we compare computations with various orders of diffraction. Studies of diffraction order are done to determine when inclusion of higher orders is necessary or may be neglected for applications such as interactive auralization.

10:00

2aAAa8. Room acoustics auralization studies of laterally incident boss-model scattering. Rendell R. Torres (Prog. in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180-3590, rrtorres@rpi.edu), Mendel Kleiner (Chalmers Univ. of Tech., SE-41296 Gothenburg, Sweden), and Georgios Natsiopoulos (Akustikon, SE-411 02 Gothenburg, Sweden)

Computation of room impulse responses by commercial auralization programs typically employs simplified Lambert models to account for nonspecular scattering. However, Lambert scattering models, developed primarily for diffuse lighting computation, have limited validity in acoustics. Instead, in this study, surface scattering from hemispherical bosses on walls is computed with numerical models based on exact classical solutions. Scattering impulse responses for auralization are calculated with various configurations of bosses on infinite planes on both sides of a binaural receiver (approximately 7–10 m away). The physical sound field is analyzed along with the resulting subjective effects of varying boss size and scatterer density.

10:15

2aAAa9. Acoustic diffusers III. Alejandro Bidondo (A. B. Ingenieria de Sonido, Argentina)

This acoustic diffusion research presents a pragmatic view, based more on effects than causes and 15 very useful in the project advance control process, where the sound field's diffusion coefficient, *sound field diffusivity* (SFD), for its evaluation. Further research suggestions are presented to obtain an octave frequency resolution of the SFD for precise design or acoustical corrections.

Session 2aAAb

Architectural Acoustics: Noise Isolation

Brandon Tinianov, Chair

Acoustical Laboratory, Johns Manville, 10100 West Ute Avenue, P.O. Box 625005, Littleton, Colorado 80162

Contributed Papers

10:45

2aAAb1. Prospects for a test procedure for rating floor toppings on joist floors. Alf Warnock (Natl. Res. Council Canada, M59, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, alf.warnock@nrc.ca)

Recently ASTM issued a new test method, E2179, for rating the reduction in impact noise when a floor topping is placed on a concrete slab. E2179 is essentially the same as ISO140-8. Measurements have shown that the reduction in impact noise obtained when a floor topping is placed on a concrete slab is not the same as when the topping is placed on a wooden subfloor supported on joists. To complicate matters, the improvement obtained for a given topping changes when the construction of the joist floor on which it is placed changes. Thus a standard test method with a rigorous definition of a reference joist floor might be created but the measured improvements would not necessarily be applicable to joist floors of different construction. The ISO and ASTM single number ratings for some joist floor systems with toppings do not always give good agreement because of the 8 dB rule that exists in ASTM E989 but is absent in ISO 717-2. This paper will review some of the activities in ASTM and ISO and present some measurement resuts obtained in several laboratories.

11:00

2aAAb2. Sound reduction by simple walls of brick and concrete. Rafael A. C. Laranja and Alberto Tamagna (UFRGS–DEMEC–GMAp, Rua Sarmento Leite, 425–Porto Alegre-RS-Brazil, 90050-110, raclaranja@hotmail.com)

Sound insulation can be difficult to forecast. In most cases it is necessary to take certain precautions in order to avoid degradation in the acoustical performance of walls. Knowing that sound transmission through walls depends on mass per unit area, bending stiffness, damping, mounting conditions, frequencies, etc., the sound transmission can be explained theoretically by several hypotheses. Even though sound transmission has been investigated for more than 100 years, there still remain a lot of issues to be solved. The purpose of this work is to present a discussion regarding the main analytical models used in sound reduction by simple walls, to help the engineer in noise control projects for industry or homes. Several model cases were analyzed by numerical estimation, and the best results were selected to perform experimental comparisons. Some comparative graphics are presented to compare numerical and experimental results. [Work supported by CAPES.] (To be presented in Portuguese.)

11:15

2aAAb3. Measurement and simulation of brick wall sound reduction index. Dinara X. da Paixao (Civil Eng., Federal Univ. of Santa Maria, Santa Maria, RS, Brazil, CEP: 97105-900), Elias B. Teodoro (Federal Univ. of Uberlandia, Uberlandia, MG, Brazil), and Samir N. Y. Gerges (Federal Univ. of Santa Catarina, Florianopolis, SC, Brazil)

This paper presents the procedures and results of an experimental, numerical simulation and analytical model of sound reduction index of solid brick wall. The solid brick used has the following dimensions: 22 $\times 10 \times 5$ cm and density of 1850 kg/m^3. The physic-mechanical charac-

teristics: density, resistance, and elasticity modulus were measured using a set of small walls (0.60×0.63 m). One wall of $4.10 \times 3.20 \times 0.10$ m (length/height/thickness) was built between two reverberant rooms, with 60 and 63 cubic meters in volume, respectively. Two kinds of junctions were used between the wall and the concrete walls of the reverberant rooms: elastic junction (with a thin layer of rubber, sealed with silicon rubber on both sides and the top of the wall) and rigid junction (using the normal mortar). The sound reduction index was measured according to ISO 140-3 using airborne transmission. The brick characteristics were used as input data for the statistical energy analysis (SEA) model. The results from SEA, analytical, and measured models are compared.

11:30

2aAAb4. Active control of the headpiece of noise barriers. Michael Möser and Hyo-In Koh (Inst. of Tech. Acoust., Tech. Univ. Berlin, Einsteinufer 25, 10587 Berlin, Germany, m.moeser@tu-berlin.de)

A theoretical and numerical study of the sound field around an active controlled edge of a noise barrier is described. Studies about noise barrier with passive methods have shown improvement in reduction of the sound level in the range of middle and high frequencies. In this work the sound field on the surface of the headpiece is controlled in such a way that the tangential power transport parallel to their surface is lowered by means of secondary sound sources. First the sound field on the surface of the cylindrical headpiece is locally minimized. The effect of the control position and number of the control point on the sound field near the headpiece and in far field at various frequencies is discussed. The radiated power in the far field of the shadow region is globally minimized, which is the target to reduce, and compared to previous results. Afterwards the method of modal restructing is shown whereby each modal amplitude is controlled individually. Theoretical computations show improved levels in the shadow zone. Optimal active control methods are discussed and a practically oriented simulation is planned.

11:45

2aAAb5. The acoustical performance of building façades designed for warm and humid climates. Elcione Maria Lobato de Moraes (Univ. of the Amazonia, Brazil, elcione@hotmail.com)

The effects of noise on man are known to range from simple nuisances to illnesses and trauma. Studies of acoustic isolation provided by buildings in urban centers assume that external openings such as windows and doors remain closed. However, this paper presents results of a study of the acoustic isolation of airborne noise for different types of façade construction in the city of Belén, Brazil. The study recognizes that windows and doors are opened as a function of the local climatic conditions since the city is located in a warm and humid tropical area. The purpose of the investigation is to provide information to planners so that they can better control acoustic contamination of buildings located in these kinds of climates. It is concluded that acoustic isolation is affected by the façade construction and by openings in the façade.

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Pan-American/Iberian Meeting on Acoustics   **2227**

Session 2aAB

Animal Bioacoustics: Amphibian Bioacoustics in Honor of Robert Capranica I

Peter M. Narins, Chair

Department of Physiological Science, University of California, Los Angeles, California 90095-1606

Chair's Introduction—8:00

Invited Papers

8:10

2aAB1. Bob meets bullfrogs: Love at first croak. Moise H. Goldstein, Jr.   (Marine Biol. Lab., Woods Hole, MA 02543)

After graduating from UC Berkeley, young Bob took at job at Bell Labs (BTL) and he and his lovely wife Patricia moved east. Somehow he found himself in a ''wet'' biological laboratory hidden in a corner of the BTL Murray Hill campus. Here Bob found Larry Frishkopf and the author, a MIT summer visitor at BTL, doing unit recording from bullfrog auditory nerves. Bob was pleased to help with the project. Sometimes, we would speculate about the tie between the electrophysiology and bullfrog behavior. As summer ended, Bob competed for and won a BTL Ph.D. fellowship. Bob chose to do his dissertation research at MIT and to address that question. I was his preceptor. He developed a novel method, sound evoked vocalizations, and skillfully demonstrated the relation of neural coding and behavior in bullfrogs. Soon after competing his dissertation research, Bob accepted a faculty position at Cornell. It is a great pleasure for me to take part in honoring Bob's brilliant career.

8:25

2aAB2. Neural mechanisms of call recognition in Pacific treefrogs. Gary J. Rose   (Dept. of Biol., Univ. of Utah, Salt Lake City, UT 84112)

Pacific treefrogs, Hyla regilla, use advertisement and encounter calls in their acoustic communication. These two call types differ primarily in the rate that pulses are repeated; at 17 °C, pulse repetition rates (PRRs) for advertisement and encounter calls are about 100 pulses/s and 30 pulses/s, respectively. Behavioral studies indicate that two discrete channels exist for processing these communication signals. Because sound pulses in these two call types are highly similar in spectral and temporal structure, the PRR selectivity of the advertisement filter is likely to result from an analysis of interpulse interval. Behavioral studies support the hypothesis that temporal integration might underlie this analysis. As a neural correlate to the behavioral studies, toral neurons have been recorded that respond selectively to PRRs found in the advertisement call. These cells fail to respond to encounter call PRRs. In support of the integration hypothesis, these units only respond after a threshold number of correct interpulse intervals have been presented; the average PRR is largely irrelevant. The interval-counting process can be reset by a single inappropriate interval. In the most selective cases, a single interval that is slightly longer or shorter than the optimal interval can reset the integration process. [Work supported by NIDCD.]

8:50

2aAB3. Nonlinearity in central auditory processing. Albert S. Feng   (Beckman Inst., Univ. of Illinois at Urbana–Champaign, 405 N. Mathews, Urbana, IL 61801)

The pioneering work of Robert Capranica in sound communication in bullfrogs revealed that sound pattern recognition involves a nonlinear operation. He showed in 1965 that, for bullfrogs, although the peripheral auditory system exhibits nonlinear properties, nonlinearity of the central auditory system is necessary for sound perception. Researchers studying sound processing in anurans as well as in birds and mammals have since shown that central auditory processing underlying various perceptual tasks (e.g., localization, ranging, and pattern recognition) mostly involves nonlinear operations; these are time- or frequency-dependent or both. Signal detection and discrimination in noise is another example. Frogs exhibit spatial unmasking, a space-dependent phenomenon, i.e., the masking effect is reduced when signal and noise sources are spatially separated. Spatial unmasking allows frogs to communicate effectively by sound in a dense chorus. Recent physiological evidence showed that spatial unmasking is largely attributed to nonlinear binaural processing in the CNS. The auditory periphery contributes to spatial unmasking but the processing therein is largely linear and its role is more limited. [Work supported by NIH/NIDCD.]

9:15

2aAB4. A reexamination of the teleost swimbladder as an acoustic source. Michael L. Fine   (Dept. of Biol., Virginia Commonwealth Univ., Richmond, VA 23284-2012, mfine@atlas.vcu.edu)

Classically swimbladders are considered pulsating resonant bubbles that are omnidirectional monopole sources capable of translating acoustic pressure to the ears. Swimbladder sounds are driven by sonic muscles (the fastest in vertebrates), yet high speed would seem unnecessary to excite a resonant structure. Recent studies in the oyster toadfish Opsanus tau and the weakfish Cynoscion regalis suggest that the classic generalizations may not apply to all fishes. The toadfish bladder is a low-Q inefficient (highly-damped) resonator that moves in a quadrupole fashion producing sound amplitude proportional to bladder velocity. Slow movements do not

produce audible sound. When the bladder is stimulated artificially, dominant frequency is determined by the waveform of the excitation stimulus and not fish size. The sound field, measured underwater, is mildly directional, and deflation of the bladder does not change auditory thresholds. The dominant frequency of weakfish sounds (pulses produced by individual muscle contractions) appears to be determined by timing parameters of muscle twitches and not the natural frequency of the bladder. [Work supported by NIH.]

2aAB5. Eighth-nerve regeneration, 20 years later. Andrea M. Simmons, Judith A. Chapman, and Alison Barnstable (Dept. of Psych., Brown Univ., Providence, RI 02912, Andrea_Simmons@brown.edu)

Over 20 years ago, Harold Zakon and Bob Capranica began a series of studies tracing the anatomical and physiological sequelae of damage to the eighth cranial nerve in ranid frogs. Their results indicated that the severed nerve could regenerate back into the brain, that proper anatomical connectivity was re-established, and that appropriate neural responses in target nuclei re-emerge. In an extension of these studies, patterns of gene expression in the crushed nerve and in target brainstem nuclei are being examined at various time points after damage. These molecular changes are correlated with the time course of recovery of normal vestibular function. Passive (head tilt) and active (response to rotation) vestibular function recover at different rates, with response to rotation still abnormal at 12 weeks after damage. This delayed recovery indicates a process of regeneration, rather than behavioral compensation. Differential display, a PCR-based technique, is used to elucidate which mRNAs are expressed at different time points following nerve crush. At periods less than 1 week after damage, mRNAs, both novel and previously identified, are differentially expressed in crushed versus sham-operated tissue. One gene product, expressed at 24 h post-crush, exhibits 91% coding region homology to human brain-specific protein, CGI-38.

10:05–10:25 Break

Contributed Papers

2aAB6. Auditory organs in *Xenopus laevis* and *Xenopus tropicalis*. Elba E. Serrano and Quincy A. Quick (Biol. Dept., NMSU, Las Cruces, NM 88003)

The genus *Xenopus* is comprised of over 20 genetically divergent species that occupy aquatic habitats such as lakes, rivers, and swamps south of the African Sahara. Like other anurans, advertisement calls are important signals for reproductive behavior and *Xenopus* have a vocal apparatus adapted for underwater sound production. The tetraploid *Xenopus laevis* is a well-established model organism for cell and developmental biology. *Xenopus tropicalis*, a diploid member of the *Silurana* group, is a newer *Xenopus* model for molecular genetics that is suited for transgenic studies. Both species are being used for investigations of inner ear organogenesis. Results presented here use light and confocal microscopy to examine the structure of the auditory organs of *X. laevis* and *X. tropicalis*. Images gathered from sectioned tissue stained with hematoxylin/eosin or micro-dissected organs labeled with Alexa 488 phalloidin (Molecular Probes) and propidium iodide illustrate the organization and innervation of the sensory epithelia of the sacculus, amphibian papilla and basilar papilla of the two species. The data show the similarities between the sensory fields and highlight the size differences of the organs in the two species. [Work supported by grant to EES (NIGMS, NIDCD) and awards to QQ (NASA NMSGC, NIGMS RISE).]

2aAB7. Auditory nerve recordings in *Xenopus laevis*. Darcy Kelley, Taffeta Elliot (Dept. of Biol. Sci., Columbia Univ., New York, NY 10027), and Jakob Christiasen-Dahlsgaard (Odense Univ.)

The South African clawed frog *X. laevis* communicates using a rich repertoire of underwater calls made up of clicks. Two calls, rapping and ticking, differ only by interclick interval [Tobias *et al.*, PNAS (1998)]. To begin to explore rate coding in the auditory system, extracellular recordings were obtained *in vivo* from the auditory nerve of frogs stimulated with both pure tones and recorded calls. To avoid problems of impedance matching, acoustic stimuli were delivered via vibration-excited probe directly to the frog's tympanum. Preferred frequencies across the population of 104 recorded fibers (taken from 5 females) fell into 2 or 3 groups: 1500, 600, and 200 Hz. Cells of all preferred frequencies fired in phase with clicks of the various calls. [Work supported by an NSF graduate fellowship to TME.]

2aAB8. Suppression of frog amphibian-papillar (AP) axons. Edwin R. Lewis (Dept. of EECS, Univ. of California, Berkeley, CA 94720, lewis@eecs.berkeley.edu), Pim van Dijk (Univ. Hospital Maastricht, The Netherlands), and Water M. Yamada (Univ. of Southern California, Los Angeles, CA 90089)

In response to white-noise stimuli, AP afferent axons of *Rana catesbeiana* and *R. esculenta* exhibit excitation, adaptation, and suppression simultaneously. After eigendecomposition of the second-order Wiener kernel, spectrotemporal properties of suppression and adaptation, in the individual axon, can be reconstructed by short-term averaging taken parallel to the main diagonal of the inhibitory subkernel. The results show suppressive spectra with deep, sharp notches in the vicinity of BEF. Spectral components outside the notch, on both sides of BEF, are effective in eliciting the suppression response; those very close to BEF are not. The results imply that the suppressive response (a negative dc shift in instantaneous spike rate) may occur with or without the presence of excitatory stimuli. It will be visible, however, only if there is background spike activity. The suppressive spectrum on the high side of BEF can extend beyond the highest AP BEF. In other words, AP axons can respond (negatively) to stimuli at frequencies beyond the highest BEF. The results further imply that a brief suppressive stimulus will be most effective if applied toward the end of a brief excitatory stimulus, or slightly after that excitatory stimulus has ended.

2aAB9. Amplitude modulation encoding in different auditory nuclei of the frog. G. Nikolay Bibikov (N. N. Andreyev Acoust. Inst., Schvernik St. 4, Moscow 117036, Russia)

Since R. R. Capranica published his pioneer articles dealing with the amplitude modulation (AM) encoding, we have explored the reproduction of sine AM in a thousand single units of the medullar, isthmal, and mid-brain acoustical centers of the European frogs. Typically, it was phasic units that demonstrated the best phase-locking to the 80%–100% modulated tone. However, onset units usually did not reproduce modulation with low modulation indexes (10%–20%). Medullar tonic units moderately reproduced deep AM and scarcely reproduced the 10% AM. However, the population response to a simple AM tone demonstrated a reliable reproduction of 10% modulated signal and a slight tendency to increase phase-locking from the first to the last modulation period. In the superior olive and lateral lemniscus nuclei this tendency became more evident. In torus semicirdularis a large population of the units increase their phase-locking response during the short-term (tenth of second) and long-term (tens of seconds) adaptation. In many units a dramatic enhancement of the

mean firing rate in the sustained state was observed after noise addition. Morever, the amplitude of the sine modulation of the instantaneous spike rate was enhanced on some levels of the noise. [Work supported by the RFBR Grant No. 02-04-48236.]

11:25

2aAB10. Multiple components in distortion product otoacoustic emissions from the amphibian ear. Sebastiaan W. F. Meenderink and Pim Van Dijk (Dept. of Otorhinolaryngology and Head & Neck Surgery, P.O. Box 5800, 6202 AZ Maastricht, The Netherlands)

Distortion product otoacoustic emissions (DPOAEs) are weak intermodulation distortions generated in the inner ear in response to two stimulus tones. Both the amphibian papilla and the basilar papilla in the inner of the frog may generate DPOAEs [Van Dijk and Manley, Hear. Res. **153**, 14–22 (2001)]. Here, we measured the level and phase of DPOAEs in the Leopard frog, *Rana pipiens*, in response to stimulus tones between 40- and 90-dB SPL. Results show that for stimulus tones in the amphibian papilla frequency range, two components contribute to DPOAEs. One component dominates for stimulus levels below about 70-dB SPL, while the other is most prominent at higher levels. The transition between both components is accompanied by a conspicuous phase change, and sometimes by a notch in the amplitude response curve. Similar results were obtained in the basilar papilla frequency range, but in addition a third component was present for stimuli below about 55-dB SPL. With the exception of this third component, our findings are remarkably similar to those in mammals, despite the structural differences between the mammalian and amphibian inner ear. [Work supported by NWO.]

TUESDAY MORNING, 3 DECEMBER 2002   CORAL GARDEN 2 AND 3, 8:00 A.M. TO 12:00 NOON

Session 2aAO

Acoustical Oceanography: Sensing the Basin Scale

Peter F. Worcester, Chair

Scripps Institute of Oceanography, University of California, San Diego, 9500 Gilman Drive, La Jolla, California 92093-0225

Contributed Papers

8:00

2aAO1. Observations of internal tide variability in the far field of the Hawaiian Ridge: The far field component of the Hawaii Ocean Mixing Experiment (HOME). Brian Dushaw (Appl. Phys. Lab., College of Ocean and Fishery Sci., Univ. of Washington, Seattle, WA 98105, dushaw@apl.washington.edu), Peter Worcester, Matthew Dzieciuch (Univ. of California, San Diego, San Diego, CA 92093-0225), and Doug Luther (Univ. of Hawai'i at Manoa, Honolulu, HI 96822)

As part of the Hawaii Ocean Mixing Experiment (HOME), observations of internal tides in two regions on either side of the Hawaiian Ridge were obtained by tomography, thermistors, and CTD casts from FLIP. The tomographic observations detect radiation of low internal-tide modes in broad areas, while the thermistors and CTD casts measure the "local" internal-tide variability. These observations are used to estimate the amount of energy carried away from the Ridge by the internal tides, to estimate the relative energies of low- and high-mode internal tides, and to test numerical models of internal-tide generation. Barotropic currents and pressure were also measured by tomography, electromagnetic, and pressure sensors so that, with careful modeling, the energy lost from the barotropic tides at the Ridge can be determined. Thermistor data obtained on one mooring showed that the M2, mode-1 internal tide was mainly phase-locked and carried 1.3 kW/m of energy. Modes 2 and 3 had amplitudes comparable to mode-1, but they were not phase locked. Energy fluxes at three other moorings were 1.2, 2.0, and 6.7 kW/m. Energy fluxes obtained by tomography were 0 (1 kW/m) or less; the line-integral data are less susceptible to the interference effects in the outgoing internal waves.

8:15

2aAO2. Acoustic thermometry along an Arctic Ocean path. Peter Mikhalevsky, Brian Sperry (SAIC, 1710 SAIC Dr., McLean, VA 22102), and Alexander Gavrilov (P. P. Shirshov Inst. of Oceanology, Moscow 117851, Russia)

Acoustic thermometry has been shown to be a very effective technique for monitoring average heat content and average temperature in the Arctic Ocean and in particular in the Arctic Intermediate Water (AIW) layer. As part of the U.S./Russian Arctic Climate Observations using Underwater Sound (ACOUS) program a 14-month time series of acoustic transmissions were analyzed along a 1250 km propagation path that extended from the Franz Victoria Strait to the Lincoln Sea from Oct. 1998 through Dec. 1999. The receive array mooring in the Lincoln Sea was recovered in April 2001. Modal travel times were estimated after pulse compression processing and mode filtering of the vertical line array. The interarrival time between mode 1 and modes 2 and 3 show net cooling during the first several months followed by a dramatic warming of the AIW along the propagation path. This warming is consistent with direct CTD measurements made along a central Arctic transect performed by the USS Hawkbill during the Scientific Ice Exercise (SCICEX) 2000. [Work supported by ONR, NSF, the Civilian Research and Development Foundation, and the Ministry of Industry, Science and Technology of the Russian Federation.]

8:30

2aAO3. Long-range detection of hydroacoustic signals from large Antarctic icebergs. Jacques Talandier, Olivier Hyvernaud, Pierre F. Piserchia (Departement Analyse, Surveillance, Environnement du Commissariat a l'Energie Atomique, BP 12, 9680 Bruyeres-Le-Chatel, France), and Emile A. Okal (Northwestern Univ.)

T-waves are commonly observed on coastal seismographs of the French Polynesian Seismic Network (RSP), when an oceanic earthquake or an underwater explosion occurs, even for small events. *T*-waves are trapped in the underwater channel and can propagate at very long distances before being converted into seismic waves close to the coastal seismic stations. During the 2000/2001 Austral summer, coastal seismic stations of the RSP detected unique series of *T*-waves from Antarctica about 60 away in the frequency band 2–15 Hz. Some of them last a few minutes while other wavetrains last several hours; some are broadband while others feature prominent frequencies, occasionally accompanied by overtones. Most of the hydroacoutics sources are relocated using the RSP stations and some Antarctic seismographs. It is shown that observed waves

have a very long underwater path but may also propagate in the ice sheet. Satellite monitoring demonstrates that hydroacoustic source locations are very well correlated in space and in time with icebergs B-15B and B-17 moving off the Ross Ice Shelf. These two icebergs appeared after the Iceberg B15 broke from the Ross Ice Shelf in March 2000.

8:45

2aAO4. The dynamics of abyssal T-phases. Ralph A. Stephen, Deborah K. Smith (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), and Clare Williams (MIT/WHOI Joint Prog. in Oceanogr., Woods Hole, MA 02543)

The characteristics of earthquakes, as revealed by T-phase observations, have the potential to provide important constraints on physical models of crustal processes under the oceans. Although it has been postulated that some form of scattering at or near the seafloor is necessary to convert the compressional and shear body waves from earthquakes into the low grazing angle paths necessary for propagation in the ocean sound channel, there are T-phase observations that cannot be explained by seafloor scattering alone. Water depth above the epicenter, for example, should have a strong effect on T-phase excitation. We use the time domain finite-difference method combined with ray theory to demonstrate these issues and we compare the theory to a series of events that occurred near the mid-Atlantic Ridge at the Kane Fracture Zone (MARK) in 1999 and 2000. There is evidence in this data set which suggests that topographic steering of T-phase locations occurs. Earthquake energy appears to preferentially enter the sound channel at topographic highs and epicentral locations are biased toward shallow bathymetry.

9:00

2aAO5. Propagation of sound through a spicy ocean: Analysis. Walter Munk, Matthew Dzieciuch, and Daniel Rudnick (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093)

We derive some of the parameters for a simple model of spice and internal wave scatter, using the canonical ocean sound channel. A distant goal is for long-range acoustic transmissions to provide a measure of upper ocean stirring from abyssal ocean acoustic signatures.

9:15

2aAO6. Propagation of sound through a spicy ocean: A numerical experiment. Matthew Dzieciuch, Walter Munk, and Daniel Rudnick (Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA 92093)

Using a closely sampled 1000 km hydrographic section in the eastern North Pacific, we separate the sound-speed finestructure into 2 component fields: (i) isopycnal tilt dominated by internal waves; and (ii) "spicy" (hot and salty) millifronts associated with upper ocean stirring. Scattering by the spicy millifronts is of the same order as internal wave scattering (the traditional view), and they both contribute to the penetration of sound into the shadow zone.

9:30

2aAO7. Travel time bias at sound reflection from an uneven surface: Implications for ocean remote sensing. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA/Environ. Technol. Lab., Boulder, CO 80305) and Iosif M. Fuks (ZEL Technologies, LLC and NOAA/Environ. Technol. Lab., Boulder, CO 80305)

Travel time is the acoustic quantity which is most frequently used to infer physical properties of the ocean or its bottom from acoustic measurements. Ocean boundaries typically have a complicated shape with smaller-scale features normally described as random roughness. The roughness

translates into fluctuations of reflected acoustic arrivals. In this paper, statistical properties of acoustic travel time are considered in homogeneous and a vertically-stratified 3-D ocean with rough boundaries and interfaces. On average, the rough surfaces can be either horizontal planes, corresponding to the ocean surface, or deterministic curved surfaces, corresponding to the sea floor or interfaces within the ocean bottom. It is shown that mean acoustic travel time differs from the travel time in an average medium. In particular, in agreement with Fermat's principle, small cross-range slopes of the reflecting surface always decrease ray travel time. The travel time bias is studied using the ray and the adiabatic normal mode theories. Implications are analyzed of the travel time bias on interpretation of measurements made with echo sounders, inverted echo sounders, and tomography systems employing surface- and/or bottom-reflected arrivals. [Work supported by ONR.]

9:45

2aAO8. Comments on "ray chaos" and ocean acoustic tomography. Brian Dushaw (Appl. Phys. Lab., Univ. of Washington, 1013 N.E. 40th St., Seattle, WA 98105) and John Colosi (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Recent publications describe "ray chaos" in the context of long-range acoustic propagation in the ocean. This work is relevant to long-range tomography, which relies on the identification of specific ray paths with pulse arrivals. However, the "ray chaos" work has mainly had a theoretical or numerical focus. Of necessity, artificial assumptions have been made as to internal-wave spectrum, infinite acoustic frequency, background sound speed profile, etc. The discussions of "ray chaos" have rarely incorporated the results of the long-range propagation experiments that have been conducted over the past 15 years. In spite of this disassociation between theory/numerics and experiment, discussions involving "ray chaos" have been critical of tomography/thermometry. One claim is that acoustic rays employed for purposes of tomography are an inappropriate description of the acoustic sampling associated with measured arrival patterns. Such criticism has occasionally implied that tomography cannot be employed for ocean studies at ranges larger than about 1 Mm as a result of "ray chaos" and other issues. Some aspects of the "chaos" view are obviously correct, while other aspects require a more rigorous test of modeling by experiment. However, the oceanographic measurements that result from using either "classical" or "chaotic" rays are practically indistinguishable.

10:00–10:15   Break

10:15

2aAO9. Discrepancies between ocean-acoustic fluctuations in parabolic-equation simulations and estimates from integral approximations. Michael D. Vera (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0225) and Stanley M. Flatté (Univ. of California, Santa Cruz, Santa Cruz, CA 95064)

Analytic, line-integral approximations to the acoustic path integral have been used to estimate the magnitude of internal-wave-induced fluctuations in a signal traveling through the ocean. These approximations for the bias and variance of travel time, the length scale of acoustic coherence in depth, and the spreading in time of acoustic intensity peaks are compared, in this discussion, to values from simulations that used the standard parabolic equation. Two different temperate-latitude sound-speed profiles were used in simulated 250 Hz acoustic propagations with a maximum range of 1000 km. The sound speed was perturbed by internal waves conforming to the Garrett–Munk (GM) spectral model with strengths of 0.5, 1, and 2 times the GM reference energy level. Though predictions of

the travel-time variance were largely successful, the other quantities examined did not correspond to simulation values. Calculated biases deviated from parabolic-equation results at ranges beyond a few hundred kilometers. The predicted depth-coherence lengths at 1000 km were significantly shorter than those extracted from the simulations. The estimated magnitudes of pulse spreading at 1000 km were much greater than the differences in widths between intensity peaks from simulations with and without internal-wave perturbations.

10:30

2aAO10. Calculations of ocean-acoustic fluctuations for use in tomography of internal waves. Stanley M. Flatté and Michael D. Vera (Phys. Dept., Univ. of California, Santa Cruz, CA 95064)

In order to use observations of the bias and variance of travel time, the length scale of acoustic coherence in depth, and the spreading of the acoustic pulse as information about the internal-wave field in the ocean (all for identifiable rays at long range), a reliable method of calculation of these quantities in the presence of a specific internal-wave model must be made. Attempts to use analytical methods having proven unreliable for long range, except for the variance of travel time; there are two possibilities: (1) restrict tomography of identifiable rays to travel-time variance; or (2) use simulation by parabolic equation to calculate expected values. The former is restricted and the latter is computer intensive. The needed resources for the latter will be discussed.

10:45

2aAO11. Limitations on perturbation theory applied to ocean acoustic inversion. B. Edward McDonald (Naval Res. Lab., Code 7145, Washington, DC 20375), Brian Sperry (SAIC, McLean, VA 22102) and Arthur Baggeroer (MIT, Cambridge, MA 02139)

Perturbation theory for ocean acoustic modal group speed responses to small environmental changes is investigated with regard to its applicability in ocean acoustic tomography. Assuming adiabaticity, the inverse problem for each vertical eigenmode is an integral equation whose kernel involves the eigenfunction and its frequency derivative. A new proof is given for the so-called "third term problem" which requires equivalence between two dissimilar forms of the integral equation. Numerical examples are given for the inversion kernel for four types of sound speed profiles, and then the parameter range (amplitude and scale size) in which perturbation theory is accurately examined. It is found that the range of validity is set not only by the amplitude of the perturbations, but also by their vertical scale size. [Work supported by ONR and Saclantcen.]

11:00

2aAO12. Long range nonlinear propagation in an ocean acoustic waveguide. Kaelig Castor, Peter Gerstoft, Philippe Roux, W. A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA 92037), and B. E. McDonald (Naval Res. Lab., Acoust. Div., Washington, DC 20375)

The Nonlinear Progressive Wave Equation (NPE) [McDonald and Kuperman, J. Acoust. Soc. Am. **81**, 1406–1417 (1987)] is an approximation of the Euler equations for nonlinear compressional waves in an inviscid fluid and, actually, is the nonlinear time domain counterpart of the frequency domain linear parabolic wave equation (PE) for small angle propagation. Simulations using a NPE code were used to study both harmonic (high frequency) and parametric (low frequency) generation. This code was coupled with a linear adiabatic normal mode program which allows

range-dependent cases treatment to study propagation in shallow or deep water with longer range propagation paths. Included in the modeling are both shock dissipation and linear attenuation in the sediment layer. The results of these studies are presented.

11:15

2aAO13. Downslope measurements from a bottom mounted tomography source. Kevin D. Heaney (ORINCON Industries, 4350 N. Fairfax Dr., Ste. 470, Arlington, VA 22203), Brett Castille, Arthur Teranishi, and Daniel Sternlicht (ORINCON Industries, San Diego, CA 92121)

Long-range tomography experiments seek to measure the temporal acoustic fluctuations due to thermal changes in the ocean structure. To map these changes to specific depths in the ocean, accurate travel times, and the ability to predict ray paths are required. Both of these become difficult in an environment that is bottom interacting (even though the water may be deep). To examine near source effects of interaction with the bottom for a tomographic source set on the bottom (at a depth of 800 m) measurements were taken off of Kauai. A short VLA was deployed from a small vessel and broadband recordings were taken at 6 ranges, from directly overhead to 55 km away. The issues to be addressed are the accuracy of the source timing and the relative strength of the bottom bounce and direct path. Evidence exists that interaction with the sea-floor can lead to up to a 0.5 s delay in the measured travel time, from that predicted.

11:30

2aAO14. Horizontal refraction and coherence of acoustic signals propagating over a long-range in the ocean. Alexander G. Voronovich, Vladimir E. Ostashev (NOAA/Environ. Technol. Lab., 325 Broadway, Boulder, CO 80305), and The NPAL Group[a]

The paper is devoted to experimental and theoretical studies of horizontal refraction and coherence of acoustic signals recorded during the NPAL experiment with the use of the billboard acoustic array [The NPAL Group, J. Acoust. Soc. Am. **109**, 2384 (2001)]. Both ray- and mode representations of the acoustic field are used for signal-processing. In the first approach, signals at different pairs of hydrophones located at approximately the same depth are cross-correlated. This allows us to obtain the mean horizontal refraction angle, its time-dependence and variance, and the coherence radius of the sound wavefront. The variance and the coherence radius are also estimated using a theory developed. In this theory, the energy of the sound field scattered by internal waves spreads in horizontal directions according to a diffusion law. The diffusion coefficient is evaluated numerically for the Garret–Munk spectrum and canonical Munk profile. It is shown that the variance of the horizontal refraction angle and the coherence radius of the wavefront calculated theoretically agree qualitatively with their values obtained using the NPAL data. [Work supported by ONR.] [a] J. A. Colosi, B. D. Cornuelle, B. D. Dushaw, M. A. Dzieciuch, B. M. Howe, J. A. Mercer, R. C. Spindel, and P. F. Worcester.

11:45

2aAO15. NPAL horizontal refraction: RAKE correlator estimates. Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA 92093) and the NPAL Group[a] (SIO-UCSD, APL-UW, WHOI)

A purpose of the NPAL billboard array data set was to measure the horizontal refraction of low-frequency (75 Hz), long-range (4000 km) timefronts. Simple beamforming yields time series that are too noisy for an accurate estimate of horizontal refraction variability. Since the data show partial horizontal coherence, a RAKE correlator is designed to account for the signal variance across the array and to improve the performance of the linear beamformer. [a] J. A. Colosi, B. D. Cornuelle, B. D. Dushaw, M. A. Dzieciuch, B. M. Howe, J. A. Mercer, R. C. Spindel, and P. F. Worcester.

Session 2aBB

Biomedical Ultrasound/Bioresponse to Vibration: History of Biomedical Ultrasound/ Bioresponse to Vibration

Lawrence A. Crum, Chair

Applied Physics Laboratory, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105

Chair's Introduction—10:30

Invited Paper

10:35

2aBB1. Origins and evolution of the developments which led to echo-Doppler duplex color flow diagnostic methodology. Donald W. Baker (Dept. of Bioengineering (ret.), Univ. of Washington, Seattle, WA 98105 and 13706 94th Ave. NE, Kirkland, WA 98034-1842, jdwbaker@attbi.com)

Research efforts to develop instrumentation for animal physiologic research to better characterize the cardiovascular system in engineering terms ultimately evolved for application on man and led to the Pacific Northwest becoming the current foci of the medical ultrasound industry. This presentation will trace the events from my being a student in Electrical Engineering to heading the Cardiovascular Instrument Development Program originally begun by Dr. Robert Rushmer in 1957. This narrative will range from early instruments for measurements on research animals to their development for noninvasive use on man. The instruments covered will be the transit time flow-meter, CW Doppler, pulsed Doppler, duplex scanner, color flow mapping. The role of collaboration in both engineering and many specialties of medicine will be demonstrated. Many of the original instruments have been in the Smithsonian Museum of American History and will in the near future be on permanent exhibit there.

TUESDAY MORNING, 3 DECEMBER 2002   CORAL KINGDOM 1, 8:00 TO 10:20 A.M.

Session 2aEA

Engineering Acoustics: Air Acoustics Devices, Techniques and Measurements

Zemar M. D. Soares, Cochair

Electroacoustics Laboratory, INMETRO, Av. N. S. das Gracas 50, Xerem, Rio de Janeiro 25250-020, Brazil

Gilles A. Daigle, Cochair

Institute for Microstructural Sciences, National Research Council, Ottawa, Ontario K1A 0R6, Canada

Chair's Introduction—8:00

Contributed Papers

8:05

2aEA1. Vibration balanced miniature loudspeaker. David E. Schafer, Mekell Jiles, Thomas E. Miller, and Stephen C. Thompson (Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60143)

The vibration that is generated by the receiver (loudspeaker) in a hearing aid can be a cause of feedback oscillation. Oscillation can occur if the microphone senses the receiver vibration at sufficient amplitude and appropriate phase. Feedback oscillation from this and other causes is a major problem for those who manufacture, prescribe, and use hearing aids. The receivers normally used in hearing aids are of the balanced armature-type that has a significant moving mass. The reaction force from this moving mass is the source of the vibration. A modification of the balanced armature transducer has been developed that balances the vibration of its internal parts in a way that significantly reduces the vibration force transmitted outside of the receiver case. This transducer design concept, and some of

its early prototype test data will be shown. The data indicate that it should be possible to manufacture transducers that generate less vibration than equivalent present models by 15–30 dB.

8:20

2aEA2. Measurement and numerical simulation of the changes in the open-loop transfer function in hearing aid as a function telephone handset proximity. Gilles A. Daigle and Michael R. Stinson (Inst. for Microstructural Sci., Natl. Res. Council, Ottawa, ON K1A 0R6, Canada)

The presence of a nearby object (telephone handset, cupped hand, etc.) can cause acoustical feedback to occur in a hearing aid. The object reflects or scatters additional sound energy to the microphone position causing the open-loop transfer function (OLTF) to increase. Feedback can occur when the OLTF>0 dB. To investigate this problem, measurements of the OLTF were made for three hearing aids (BTE, ITC, ITE) mounted on a KEMAR manikin. A telephone handset, positioned initially in a typical user posi-

2a TUE. AM

tion, was translated to positions between 0 and 100 mm away from the pinna, repeatedly, using a linear translation system. Changes of up to 15 dB or more were observed as the handset moved, particularly for positions within 20 mm of the pinna. In parallel, numerical simulations were made using a boundary element method. Computed changes in OLTF were consistent with the measured changes.

8:35

2aEA3. Derivation of moving-coil loudspeaker parameters using acoustical testing techniques. Brian E. Anderson (Dept. of Phys., Brigham Young Univ., Provo, UT 84602, loudspeakerdesign@hotmail.com) and Timothy W. Leishman (Brigham Young Univ., Provo, UT 84602)

A novel acoustical method of measuring small-signal moving-coil loudspeaker parameters has recently been developed. This technique involves the use of a plane wave tube to measure acoustical properties (e.g., reflection and transmission coefficients) of a driver under test (DUT). From this data, small-signal parameters are derived using curve-fitting techniques. Current loudspeaker parameter measurement techniques require measurement of the electrical impedance of the DUT. This paper will discuss the acoustical measurement apparatus, system modeling (via equivalent circuits), and a comparison of measured parameters to those derived using electrical techniques.

8:50

2aEA4. New acoustic test facility at Georgia Tech. Van Biesel and Kenneth Cunefare (Georgia Inst. of Technol., Atlanta, GA 30332, van.biesel@me.gatech.edu)

Georgia Tech's Integrated Acoustics Laboratory (IAL) is a state of the art research facility dedicated to the study of acoustics and vibration. The centerpiece of the laboratory is a 24 ft$\times$24 ft$\times$20 ft full anechoic chamber, which has been in operation since 1998. The IAL is currently expanding to include a reverberation room and hemi-anechoic chamber, designed and built by Acoustic Systems. These two chambers will be joined by an 8 ft$\times$8 ft transmission loss opening, allowing for a detailed measurement and analysis of complex barriers. Both chambers will accommodate vehicles and similarly large structures. The reverberation room will have adequate volume for standardized absorption measurements. Each chamber will be equipped with dedicated multichannel data acquisition systems and instrumentation for the support of simultaneous research in all areas of the laboratory. The new test chambers are funded by a grant from the Ford Motor Company and are planned to be completed and fully functional by 1 January 2003.

9:05

2aEA5. PC interface for a stepped filter octave band analyzer. Miguel A. Horta and Marco A. Vazquez (Instituto Politecnico Nacional, Mexico)

Spectrum analyzers are a basic tool for an acoustic engineer. Since almost anyone in a classroom has access to a PC or notebook computer, a low cost alternative for measuring many different parameters with a spectrum analyzer (such as reverberation time, transmission loss, or equivalent noise level) is a digital interface that processes the signal combined with the proper software for the computer. This is a design of a spectrum analyzer that does such a task. The external interface sequentially processes the signal in an octave band filter bank and converts it to data for the computer to display. Depending on the software, the data can be used to measure any of the signal's desired parameters.

9:20

2aEA6. Secondary microphone calibration: Advantages on the use of the constant envelope sweeps. Zemar M. D. Soares, Walter E. Hoffmann (Electroacoustics Lab., INMETRO, Av. N. S. das Gracas 50, Xerem RJ, 25250-020, Brazil, zmsoares@inmetro.gov.br), and Swen Muller (INMETRO, Xerem RJ, 25250-020, Brazil)

With the objective of reducing the costs for the accreditation of secondary laboratories in the electroacoustical area without impairing the quality of the calibration of microphones, the Laboratory of Electroacoustics of INMETRO (Brazil) has investigated the advantages of using sweeps with user-defined spectral distribution and constant temporal envelope to obtain the impulse response between sound source and microphone. In applications in which the sensibility of the microphone in the free field is the objective, the use of anechoic chambers is fundamental. However, they can be substituted by applying a windowing of the impulse response to isolate the direct sound. The same technique can be used to separate reflections in the Jig for calibration of microphones in pressure field, as proposed by IEC61094-5. The calibration process presented here is based on FFT techniques, using a special sweep as excitation signal. The sweep is custom tailored in a way that its energy contents compensates the background noise spectrum. This way, the excitation signals' signal-to-noise ratio becomes independent of frequency, while the sweep keeps an almost constant temporal envelope which contains the maximum possible energy. (To be presented in Portuguese.)

9:35

2aEA7. Free-field calibration of measurement microphones at frequencies up to 80 kHz. Allan J. Zuckerwar and Gregory C. Herring (NASA Langley Res. Ctr., M.S. 493, Hampton, VA 23681, a.j.zuckerwar@larc.nasa.gov)

Civil-aviation noise-reduction programs, that make use of scaled-down aircraft models in wind tunnel tests, require knowledge of microphone pressure (i.e., not free-field) sensitivities beyond 20 kHz—since noise wavelengths also scale down with decreasing model size. Furthermore, not all microphone types (e.g., electrets) are easily calibrated with the electrostatic technique, while enclosed cavity calibrations typically have an upper limit for the useful frequency range. Thus, work was initiated to perform a high-frequency pressure calibration of Panasonic electret microphones using a substitution free-field method in a small anechoic chamber. First, a standard variable-frequency pistonphone was used to obtain the pressure calibration up to 16 kHz. Above 16 kHz, to avoid spatially irregular sound fields due to dephasing of loudspeaker diaphragms, a series of resonant ceramic piezoelectric crystals was used at five specific ultrasonic frequencies as the free-field calibration sound source. Then, the free-field sensitivity was converted to a pressure sensitivity with an electrostatic calibration of the reference microphone (an air condenser type), for which the free-field correction is known. Combining the low- and high-frequency data sets, a full frequency calibration of pressure sensitivity for an electret microphone was generated from 63 Hz to 80 kHz.

9:50

2aEA8. Broadband self-calibrating micromachined microphones with integrated optical displacement detection. Neal A. Hall, Wook Lee, and F. Levent Degertekin (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Love Bldg., 771 Ferst Dr., Rm. 320, Atlanta, GA 30332-0105)

An optical displacement detection method for micromachined microphones is described and experimental results are presented. The microphone membrane is fabricated on a transparent substrate and the back electrode is patterned in the form of diffraction fingers. This structure forms a phase sensitive diffraction grating, providing the displacement sensitivity of an optical interferometer. The diffraction fingers are also used for electrostatic actuation, providing sensitivity adjustment and self-calibration capabilities. Optically semitransparent coatings are also employed to create Fabry–Perot resonant cavities that enhance the optical detection sensitivity by an order of 10 dB. The high electrical sensitivity provided by optical displacement detection relaxes requirements on me-

chanical sensitivity, and small microphone membranes on the order of 200 μm with vacuum-sealed and air-sealed cavities are used to fabricate microphones with a flat response from dc to over 200 kHz. The optical detection and electrostatic actuation capabilities are demonstrated on fully integrated devices with aluminum microphone membranes micromachined on quartz substrates and bonded to microfabricated silicon photodiodes. [Work supported by DARPA.]

10:05

2aEA9. Passive subtractive beamformer applied to line sound sources. Mitsunori Mizumachi and Satoshi Nakamura (ATR, 2-2-2 Hikaridai, ''Keihanna Sci. City,'' Kyoto 619-0288, Japan)

Speech is an attractive interface for mobile equipment if it is clearly received. A microphone array aids in reducing noise and enhancing speech. A small-scale microphone array generally adopts the subtractive beamforming technique, which constructs sharp notches in a beampattern in either an active or a passive way, so that target sound sources are assumed as point sound sources. However, the shapes of actual sound sources are various and not always points. In particular, the mouth needs to be modeled as line sound source or plane sound source. Therefore, beamformers should target line sound sources. A passive subtractive beamformer is proposed to expand width of a sharp notch by combining several 2ch subtractive beamformers. In multiple sound source conditions, it is necessary to avoid the effect of spatial aliasing against nontarget sound sources. Then, a hybrid technique is applied to realize an optimal connection of the quadruple, double, and single subtractive beamformer for the low, middle, and high frequency region. The feasibility of the proposed subtractive beamformer is confirmed by performance evaluation in suppressing the signals from line sound sources. [Work supported by the Telecommunications Advancement Organization of Japan.]

TUESDAY MORNING, 3 DECEMBER 2002   CORAL GARDEN 1, 8:30 TO 11:50 A.M.

Session 2aED

Education in Acoustics: Development of Acoustics Programs in Latin America

Daniel R. Raichel, Cochair

2727 Moore Lane, Fort Collins, Colorado 80526

Moyses Zindeluk, Cochair

Mechanical Engineering Department, COPPE, University of Rio de Janeiro, Caixa Postal 68503, Rio de Janeiro 21945-000, Brazil

Chair's Introduction—8:30

Invited Papers

8:35

2aED1. Education in acoustics and vibration at UFSC—Brazil. Samir N. Y. Gerges (Dept. of Mech. Eng., Federal Univ. of Santa Catarina, Cx.P. 476, Florianopolis, SC, Brazil)

In the 1970s, Brazil invested heavily on postgraduate program of all areas, especially in acoustics and vibration. Several universities achieved benefits from these investments, namely the Federal University of Santa Catarina (UFSC), the Federal University of Rio de Janeiro (UFRJ), and the Federal University of Santa Maria (UFSM). Part of the undergraduate and postgraduate studies at the Mechanical Engineering Department (EMC) of the Federal University of Santa Catarina relates to vibration and noise. On the undergraduate level an optional course called Noise Control, totaling 54 hours, is offered, which covers basic acoustics and noise control concepts. In the postgraduate program Master's and Doctorate degrees students can attend courses and pursue studies in the area of noise and vibration. This area of concentration is supported by a well equipped laboratory consisting of two reverberation chambers, with a third one, for transmission loss measurements, under construction, together with a hemianechoic room and equipment for the measurement and analysis of noise and vibration. Part of this laboratory, the Industrial Noise Laboratory, is accredited by the Brazilian authorities for measurements of and research on hearing protectors. [Work supported by the Federal Government, and industry.]

8:55

2aED2. Education in acoustics in Argentina. Federico Miyara (Acoust. and Electroacoust. Lab., Natl. Univ. of Rosario, Riobamba 245 bis, 2000 Rosario, Argentina, fmiyara@fceia.unr.edu.ar)

Over the last decades, education in acoustics (EA) in Argentina has experienced ups and downs due to economic and political issues interfering with long term projects. Unlike other countries, like Chile, where EA has reached maturity in spite of the acoustical industry having shown little development, Argentina has several well-established manufacturers of acoustic materials and equipment but no specific career with a major in acoustics. At the university level, acoustics is taught as a complementary—often elective—course for careers such as architecture, communication engineering, or music. In spite of this there are several research centers with programs covering environmental and community noise, effects of noise on man, acoustic signal processing, musical acoustics and acoustic emission, and several national and international meetings are held each year in which results are communicated and

discussed. Several books on a variety of topics such as sound system, architectural acoustics, and noise control have been published as well. Another chapter in EA is technical and vocational education, ranging between secondary and postsecondary levels, with technical training on sound system operation or design. Over the last years there have been several attempts to implement master degrees in acoustics or audio engineering, with little or no success.

9:15

2aED3. Efforts regarding acoustical education for architectural students at the Universidad Peruana de Ciencias Aplicadas (Peruvian University of Applied Sciences), UPC. Jorge Moy (Universidad Peruana de Ciencias Aplicadas, Av. Prol. Primavera 2390, Surco, Lima, Peru)

The lack of knowledge in acoustics among the vast majority of Peruvian architects results in acoustical problems in buildings from a lack of such considerations in the design stage. This paucity of knowledge on the part of the architects may be attributed to the lack of emphasis on the role of acoustics in most architectural curricula in Peruvian universities. A purpose in this paper is to present a brief report on last year's efforts to implement courses in Architectural Acoustics and Noise Control for architecture students at UPC.

9:35

2aED4. Acoustics lecturing in Mexico. Sergio Beristain (ESIME, IPN, IMA, Mexico, sberista@hotmail.com)

Some thirty years ago acoustics lecturing started in Mexico at the National Polytechnic Institute in Mexico City, as part of the Bachelor of Science degree in Communications and Electronics Engineering curricula, including the widest program on this field in the whole country. This program has been producing acoustics specialists ever since. Nowadays many universities and superior education institutions around the country are teaching students at the B.Sc. level and postgraduate level many topics related to acoustics, such as Architectural Acoustics, Seismology, Mechanical Vibrations, Noise Control, Audio, Audiology, Music, etc. Also many institutions have started research programs in related fields, with participation of medical doctors, psychologists, musicians, engineers, etc. Details will be given on particular topics and development.

9:55

2aED5. Engineering acoustics: A pioneer undergraduate program at Rio de Janeiro. Roberto A. Tenenbaum and Moyses Zindeluk (Acoust. and Vib. Lab., Mech. Eng. Dept., Federal Univ. of Rio de Janeiro, CP 68503, 21945-970 Rio de Janeiro, Brazil, moyses@ufrj.br)

Acoustics, essentially a multidisciplinary subject, still has in Brazil a small but increasing number of professionals with a solid background to deal with various aspects of this area. Since 1970 the faculty of the Acoustics and Vibration Laboratory, COPPE/UFRJ, offers graduate (M.Sc and D.Sc) programs, and some undergraduate courses in acoustics, vibration, and signal processing. In January 2000, this group launched a formal undergraduate engineering acoustics program in the Mechanical Engineering Department of the Federal University of Rio de Janeiro. After three years of mechanical engineering, with a firm foundation in physics, applied mathematics, and engineering basics, the undergraduate student may elect to take the engineering acoustics program for the remaining two years. In this program, a wide number of courses are offered, including basic acoustics, room acoustics, signal processing, musical acoustics, machine diagnosis, etc. Approximately 30 different courses may be chosen from. However, the student is not completely free, since the courses selected must fit within a subject concentration profile, e.g., noise control or musical acoustics. In this paper the programs curriculum are presented and its impact on the students is discussed. A first evaluation of the qualifications achieved by the graduate students in the area is also presented.

10:15–10:25   Break

10:25

2aED6. On helping Latin American countries in education in acoustics. Daniel R. Raichel (Douglas Eilar & Assoc., Encinitas, CA 92024-3130 and the Grad. Ctr., CUNY)

The science and applications of acoustics are just as important in Latin America as they are in North America and elsewhere. However, resources in academia are harder to come by in nearly all of the Central American and South American nations; and therefore it would behoove U.S. and European acousticians to help their Latin-American counterparts in achieving their goals of quality education in acoustics, particularly in architectural acoustics, noise control, biomedical usages of ultrasound, signal analyses, and measurement techniques. Among the means of helping are scholarly exchanges, more support by the U.S. government for such exchanges (particularly through Fulbright programs—it is unfortunate that the Fulbright Senior Specialist Program does not recognize acoustics as being one of the environmental sciences), collaboration on research projects, long-term equipment loans and/or outright donations, etc. Advice by experienced practitioners in establishing or improving acoustics laboratories can optimize equipment selection and development of the curriculum.

10:45

2aED7. Acoustics: A branch of engineering at the Universidad Austral de Chile (UACh). Victor Poblete, Jorge P. Arenas, and Jorge Sommerhoff (Inst. of Acoust., Universidad Austral de Chile, P.O. Box 567, Valdivia, Chile, vpoblete@uach.cl)

At the end of the 1960s, the first acousticians graduating at UACh had acquired an education in applied physics and musical arts, since there was no College of Engineering at that time. Initially, they had a (rather modest) four-year undergraduate program, and most of the faculty were not specialized teachers. The graduates from such a program received a sound engineering degree and they were skilled for jobs in the musical industry and sound reinforcement companies. In addition, they worked as sound engineers and producers. Later, because of the scientific, industrial and educational changes in Chile during the 1980s, the higher education system

had massive changes that affected all of the undergraduate and graduate programs of the 61 universities in Chile. The UACh College of Engineering was officially founded in 1989. Then, acoustics as an area of expertise was included, widened and developed as an interdisciplinary subject. Currently, the undergraduate program in acoustics at UACh offers a degree in engineering sciences and a 6-year professional studies in Civil Engineering (Acoustics), having two main fields: Sound and Image, and Environment and Industry.

Contributed Papers

11:05

2aED8. Acoustic Engineering program at the Universidad Austral de Chile (UACh). Jorge Sommerhoff, Victor Poblete, and Jorge P. Arenas (Inst. of Acoust., Universidad Austral de Chile, P.O. Box 567, Valdivia, Chile, jsommerh@uach.cl)

From the beginning of the acoustics program at UACh in 1968, the studies of Acoustic Engineering have been modified and developed according to the vision and human resources of its developers. Three different stages of growth can be seen. When the program began, it was totally aimed at forming skilled professionals in audio and recording. In this way, the professional title given was Sound Engineer. At that time, each applicant was required to have ''good musical hearing,'' which had to be demonstrated through a special musical audition test. The second stage was characterized by the incorporation of acoustics subjects which allowed students, with no musical abilities, to competently work on acoustic engineering activities not related to music. Then, the professional title was changed to Acoustic Engineer. Thus, job opportunities were diversified and access was allowed by all types of students. In the last stage, the study plan was modified as a response to the new vision and requirements of the globalized world in which the environmental component has a great importance. In this work the development of a program that dates from 35 years ago is presented and justified.

11:20

2aED9. Using dynamic means to teach a dynamic subject: A new concept for a graduate vibrations course in Mexico. Salvador Echeverria-Villagomez (Apdo. Postal 1-100, C.P. 76000, Queretaro, Mexico)

Mechanical vibrations is a subject that belongs in every undergraduate curricula in mechanical engineering, and to many graduate programs in mechanical design and manufacturing. It is a subject in which many disciplines come in, from mechanics, through materials properties, to mecha-

nisms and special mathematics tools. It is also a most interesting discipline from many points of view: from the theoretical models used to analyze the dynamical behavior of systems, through the modern numerical and computational tools, to the experimental techniques used to measure and test their performance. The paper presents a concept to teach vibrations and acoustics. This concept provides a framework for using the mentioned tools, which is homologous in structure to the subject being taught, thus enhancing the power and effectiveness of teaching and training in the field. The main characteristics of the concept are their systemic approach, fluid logic, use of conceptual virtual maps and dynamic means.

11:35

2aED10. Acoustic outsourcing: New employment possibilities for the specialists. Patricia Perez, Heriberto Rios, Armando Andrade, and Mario Ramirez (Laboratorio de Desarrollo Tecnologico en Bioingenieria, ESIME-IPN, Mexico)

The need for companies to be more competitive has led them to resort to training, external consultantship, continuous improvement programs, but with the aim of achieving maximum productivity, the big companies go even further: they are opting to focus on their high-priority activities, leaving some nonstrategic functions in the hands of third parties (organizations or individuals). Acoustic outsourcing presents immense business opportunities for the specialists in this area when offering services or completing a production process that the company carries out in an internal way but that is not its main function or activity. Outsourcing contemplates a serious long term commitment between the two parties; a kind of strategic alliance, all with the purpose of increasing efficiency and the quality of the products that the company develops, besides solving acoustic problems related to the production stage. (To be presented in Spanish.)

TUESDAY MORNING, 3 DECEMBER 2002   CORAL SEA 1 AND 2, 8:00 TO 11:45 A.M.

Session 2aMU

Musical Acoustics: Analysis, Synthesis, Perception and Classification of Musical Sounds

James W. Beauchamp, Chair

School of Music, Department of Electrical and Computer Engineering, University of Illinois, Urbana, Illinois 61801

Invited Papers

8:00

2aMU1. Spectral modeling, analysis, and synthesis of musical sounds. Sylvain Marchand and Myriam Desainte-Catherine (LaBRI, Univ. of Bordeaux 1, 351 cours de la Liberation, F-33405 Talence cedex, France)

Spectral models provide general representations for sound well-suited for expressive musical transformations. These models allow us to extract and modify perceptually-relevant parameters such as amplitude, frequency, and spectrum. Thus, they are of great interest for the classification of musical sounds. A new analysis method was proposed to accurately extract the spectral parameters for the model from existing sounds. This method extends the classic short-time Fourier analysis by also considering the derivatives of the sound signal, and it can work with very short analysis windows. Although originally designed for stationary sounds with no noise, this method shows excellent results in the presence of noise and it is currently being extended in order to handle nonstationary sounds as

well. A very efficient synthesis algorithm, based on a recursive description of the sine function, is able to reproduce sound in real time from the model parameters. This algorithm allows an extremely fine control of the partials of the sounds while avoiding signal discontinuities as well as numerical imprecision, and with a nearly optimal number of operations per partial. Psychoacoustic phenomena such as masking are considered in order to reduce on the fly the number of partials to be synthesized.

8:20

2aMU2. Easily extensible unix software for spectral analysis, display, modification, and synthesis of musical sounds. James W. Beauchamp (School of Music and Dept. of Elec. and Computer Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801, j-beauch@uiuc.edu)

Software has been developed which enables users to perform time-varying spectral analysis of individual musical tones or successions of them and to perform further processing of the data. The package, called SNDAN, is freely available in source code, uses EPS graphics for display, and is written in ANSI C for ease of code modification and extension. Two analyzers, a fixed-filter-bank phase vocoder ("pvan") and a frequency-tracking analyzer ("mqan") constitute the analysis front end of the package. While pvan's output consists of continuous amplitudes and frequencies of harmonics, mqan produces disjoint "tracks." However, another program extracts a fundamental frequency and separates harmonics from the tracks, resulting in a continuous harmonic output. "monan" is a program used to display harmonic data in a variety of formats, perform various spectral modifications, and perform additive resynthesis of the harmonic partials, including possible pitch-shifting and time-scaling. Sounds can also be synthesized according to a musical score using a companion synthesis language, Music 4C. Several other programs in the SNDAN suite can be used for specialized tasks, such as signal display and editing. Applications of the software include producing specialized sounds for music compositions or psychoacoustic experiments or as a basis for developing new synthesis algorithms.

8:40

2aMU3. Analysis-synthesis of musical sounds by hybrid models. S. Ystad (CNRS, Laboratoire de Mecanique et d'Acoustique 31, Chemin Joseph Aiguier, 13402 Marseille cedex 20, France)

Analysis-synthesis consists of constructing synthetic sounds from natural sounds by algorithmic synthesis methods. The models used for this purpose are of two kinds: physical models which take into account the physical characteristics of the instrument and signal models which take into account perceptual criteria. By combining physical and signal models hybrid models can be constructed taking advantage of the positive aspects of both methods. In this presentation I show how hybrid models can be adapted to specific instruments producing both sustained and plucked sounds. In these cases signal models are used to model the nonlinear source signal. The parameters of these models are obtained from perceptual criteria such as the spectral centroid or the tristimulus. The source signal is further injected into the physical model which consists of a digital wave guide model. The parameters of the physical model are extracted from the natural sound by analysis based on linear time-frequency representations such as the Gabor and the wavelet transforms. The models which will be presented are real-time compatible and in the flute case an interface adapted to a traditional flute which pilots a hybrid model will be described.

9:00

2aMU4. Recent developments in automatic classification of musical instruments. Bozena Kostek (Sound & Vision Eng. Dept., Gdansk Univ. of Technol., Narutowicza 11/12, 80-952 Gdansk, Poland)

In this paper recent developments in automatic classification of musical instrument domain are presented. Issues related to automatic classification of music are data representation of musical instrument sounds, automatic musical sound recognition, musical duet separation, music recognition, etc. These problems belong to the so-called Musical Information Retrieval domain. The best developed is the automatic recognition of individual musical sounds. In rich literature on this subject many references can be found. Another issue deals with music information retrieval understood as searching for music-related features such as song titles, etc. A query-by-humming can be also cited as one of the MIR topics. The most difficult problem that deals with automatic recognition of multipitch excerpts still remains unsolved, however, recently some approaches to this issue can be found in the literature. Some of the mentioned problems were subjects of the research carried out at the Sound & Vision Department of the Gdansk University of Technology. The developed solutions in the domain of automatic classification of individual sounds, duet separation, and music recognition will be presented as examples of possible case-studies in the MIR domain. The proposed approach was evaluated on musical datebases created at the Department. [Work supported by KBN, Grant No. 4 T11D 014 22.]

9:20

2aMU5. The timbre model. Kristoffer Jensen (Dept. of Datalogy, Univ. of Copenhagen, 2100 Copenhagen, Denmark, http://www.diku.dk)

A timbre model is proposed for use in multiple applications. This model, which encompasses all voiced isolated musical instruments, has an intuitive parameter set, fixed size, and separates the sounds in dimensions akin to the timbre dimensions as proposed in timbre research. The analysis of the model parameters is fully documented, and it proposes, in particular, a method for the estimation of the difficult decay/release split-point. The main parameters of the model are the spectral envelope, the attack/release durations and relative amplitudes, and the inharmonicity and the shimmer and jitter (which provide both for the slow random variations of the frequencies and amplitudes, and also for additive noises). Some of the applications include synthesis, where a real-time application is being developed with an intuitive gui, classification, and search of sounds based on the content of the sounds, and a further understanding of acoustic musical instrument behavior. In order to present the background of the model, this presentation will start with sinusoidal A/S, some timbre perception research, then present the timbre model, show the validity for individual music instrument sounds, and finally introduce some expression additions to the model.

9:40

2aMU6. Hypersignal analyses of orchestral instrument signals as correlated with perception of timbre. Roger A. Kendall (Music Cognition and Acoust. Lab., Schoenberg Hall, UCLA, Los Angeles, CA 90024)

Experiments were conducted to assess the relationships among signal analyses and timbral perception across the playing range of bassoon, trombone, tenor saxophone, alto saxophone, soprano saxophone, French horn, violin, oboe, flute, clarinet, and trumpet. Spectral analyses employed Hypersignal using 9th order Zoom FFT on 22.05 samples/s signals. Spectral centroid and spectral flux measures were calculated. Perceptual experiments included similarity scaling and identification at various pitch chroma across the playing range of the instruments. In addition, a pilot experiment assessing the interaction of pitch chroma and timbre was conducted where timbral judgements were made across, rather than within, pitch chroma. Results suggest that instruments with relatively low tessitura produce higher centroid ranges since the larger air column yeilds a large number of vibrational modes. In contrast, higher tessitura instruments, using smaller air columns, produce fewer modes of vibration with increasing pitch chroma, to the point that the centroids coverge near Bb5. Perceptual data correspond to the spectral, resulting in less specificity among instruments at their higher tessituras. It is suggested that spectral centroid, which maps strongly near A4 in the majority of studies, must be viewed with caution as a predictor of timbre at tessitura extremes.

10:00

2aMU7. A confirmatory analysis of four acoustic correlates of timbre space. Stephen McAdams, Anne Caclin, and Bennett K. Smith (Ircam–CNRS, 1 pl. Igor Stravinsky, F-75004 Paris, France)

Exploratory multidimensional scaling studies of musical instrument timbres generally yield two- to four-dimensional perceptual spaces. Acoustic parameters have been derived that correlate moderately to highly with the perceptual dimensions. In a confirmatory study, two three-dimensional sets of synthetic, harmonic sounds equalized for fundamental frequency, loudness, and perceived duration were designed. The first two dimensions corresponded to attack time and spectral centroid in both sound sets. The third dimension corresponded to spectral flux (variation of the spectral centroid over time) in the first set and to the energy ratio of odd to even harmonics in the second set. Group analyses of dissimilarity judgments for all pairs of sounds homogeneously distributed in each space revealed a two-dimensional solution for the first set and a three-dimensional solution for the second set. Log attack time and spectral centroid were confirmed as perceptual dimensions in both solutions. The even/odd energy ratio was confirmed as a third dimension in the second set. Spectral flux was not confirmed in the first set, suggesting that this parameter should be re-examined. Analyses of individual data sets tested for differences across listeners in the mapping of acoustic parameters to perceptual dimensions. [Work supported by the CTI program of the CNRS.]

10:20–10:30   Break

Contributed Papers

10:30

2aMU8. Piano string modeling: From partial differential equations to digital wave-guide model. J. Bensa (CNRS, Laboratoire de Mecanique et d'Acoustique 31, Chemin Joseph Aiguier, 13402 Marseille cedex 20, France), S. Bilbao (Stanford Univ., Stanford, CA), R. Kronland-Martinet (CNRS, 13402 Marseille cedex 20, France), and Julius O. Smith III (Stanford Univ., Stanford, CA)

A new class of partial differential equations (PDE) is proposed for transverse vibration in stiff, lossy strings, such as piano strings. While only second-order in time, it models both frequency-dependent losses and dispersion effects. By restricting the time-order to 2, valuable advantages are achieved: First, the frequency-domain analysis is simplified, making it easy to obtain explicit formulas for dispersion and loss versus frequency; for the same reason, exact bounds on sampling in associated finite-difference-schemes (FDS) can be derived. Second, it can be shown that the associated FDS is ''well posed'' in the sense that it is stable, in the limit, as the sampling period goes to zero. Finally, the new PDE class can be used as a starting point for digital wave-guide modeling [a digital wave-guide factors one-dimensional wave propagation as purely lossless throughout the length of the string, with losses and dispersion lumped in a low-order digital filter at the string endpoint(s)]. We perform numerical simulations comparing the finite-difference and digital wave-guide approaches, illustrating the advantages of the latter. We examine a procedure allowing the resynthesis of natural string vibration; using experimental data obtained from a grand piano, the parameters of the physical model are estimated over most of the keyboard range.

10:45

2aMU9. The wave digital piano hammer. Stefan D. Bilbao, Julius O. Smith III (Ctr. for Computer Res. in Music and Acoust., Dept. of Music, Stanford Univ., Stanford, CA 94305), Julien Bensa, and Richard Kronland-Martinet (S2M-LMA-CNRS, Marseille, France)

For sound synthesis purposes, the vibration of a piano string may be simply modeled using bidirectional delay lines or digital waveguides which transport traveling wavelike signals in both directions. Such a digital wave-type formulation, in addition to yielding a particularly computationally efficient simulation routine, also possesses other important advantages. In particular, it is possible to couple the delay lines to a nonlinear exciting mechanism (the hammer) without compromising stability; in fact, if the hammer and string are lossless, their digital counterparts will be exactly lossless as well. The key to this good property (which can be carried over to other nonlinear elements in musical systems) is that all operations are framed in terms of the passive scattering of discrete signals in the network, the sum of the squares of which serves as a discrete-time Lyapunov function for the system as a whole. Simulations are presented.

11:00

2aMU10. Musical sound analysis/synthesis using vector-quantized time-varying spectra. Andreas F. Ehmann and James W. Beauchamp (Univ. of Illinois at Urbana–Champaign, 5308 Music Bldg., 1114 W. Nevada St., Urbana, IL 61801)

A fundamental goal of computer music sound synthesis is accurate, yet efficient resynthesis of musical sounds, with the possibility of extending the synthesis into new territories using control of perceptually intuitive

parameters. A data clustering technique known as vector quantization (VQ) is used to extract a globally optimum set of representative spectra from phase vocoder analyses of instrument tones. This set of spectra, called a Codebook, is used for sinusoidal additive synthesis or, more efficiently, for wavetable synthesis. Instantaneous spectra are synthesized by first determining the Codebook indices corresponding to the best least-squares matches to the original time-varying spectrum. Spectral index versus time functions are then smoothed, and interpolation is employed to provide smooth transitions between Codebook spectra. Furthermore, spectral frames are pre-flattened and their slope, or tilt, extracted before clustering is applied. This allows spectral tilt, closely related to the perceptual parameter ''brightness,'' to be independently controlled during synthesis. The result is a highly compressed format consisting of the Codebook spectra and time-varying tilt, amplitude, and Codebook index parameters. This technique has been applied to a variety of harmonic musical instrument sounds with the resulting resynthesized tones providing good matches to the originals.

11:15

2aMU11. The syrinx: Nature's hybrid wind instrument. Tamara Smyth and Julius O. Smith III (Ctr. for Computer Res. in Music and Acoust., Stanford Univ., Stanford, CA 94305)

Birdsong is commonly associated with the sound of a flute. Although the pure, often high pitched, tone of a bird is undeniably flutelike, its sound production mechanism more closely resembles that of the human voice, with the syringeal membrane (the bird's primary vocal organ) acting like vocal folds and a beak acting as a conical bore. Airflow in the song bird's vocal tract begins from the lungs and passes through two bronchi, two nonlinear vibrating membranes (one in each bronchial tube), the trachea, the mouth, and finally propagates to the surrounding air by way of the beak. Classic waveguide synthesis is used for modeling the bronchi and trachea tubes, based on the model of Fletcher [J. Acoust. Soc. Am. (1988, 1999)]. The nonlinearity of the vibrating syringeal membrane is simulated by finite-difference methods. This nonlinear valve, driven by a steady pressure from the bronchi, generates an oscillatory pressure entering the trachea.

11:30

2aMU12. Electrophysiological correlates of musical timbre perception. Anne Caclin (Ircam–CNRS, 1 pl. Igor Stravinsky, F-75004 Paris, France), Elvira Brattico (Univ. of Helsinki, Helsinki, Finland), Bennett K. Smith (Ircam–CNRS, F-75004 Paris, France), Mari Tervaniemi (Univ. of Helsinki, Helsinki, Finland), Marie-Hilhne Giard (Inserm U280, F-69424 Lyon 03, France), and Stephen McAdams (Ircam–CNRS, F-75004 Paris, France)

Timbre perception has been studied by deriving a multidimensional space of the perceptual attributes from listeners' behavioral responses. The neural bases of timbre space were sought. First, a psychophysical timbre dissimilarity experiment was conducted. A three-dimensional space of 16 synthetic sounds equalized for fundamental frequency, loudness, and perceived duration was designed. Sounds varied in attack time, spectral center of gravity, and energy ratio of odd/even harmonics. Multidimensional scaling revealed a three-dimensional perceptual space with linear or exponential relations between perceptual and physical dimensions. Second, in an electrophysiological experiment, the mismatch negativity (MMN) component of event-related potentials was recorded. The MMN is elicited by infrequently presented sounds differing in one or more dimensions from more frequent ones. Although elicited without the focus of attention, it correlates with the subjects' behavioral responses, revealing the neural bases of preattentive discrimination. Eight sounds were chosen within the perceptual space. Changes along individual and combined dimensions elicited an MMN response. MMN latency varied depending on the dimension changed. In addition, preliminary analyses tend to show an additivity of the MMN waves for some pairs of dimensions. These results shed light on the neural processes underlying the perceptual representation of multidimensional sounds. [Work supported by the CTI program of the CNRS.]

TUESDAY MORNING, 3 DECEMBER 2002

CORAL ISLAND 1 AND 2, 8:00 A.M. TO 12:00 NOON

Session 2aPA

Physical Acoustics: Bubbles, Drops and Foams I

R. Glynn Holt, Chair
*Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street,
Boston, Massachusetts 02215*

Chair's Introduction—8:00

Invited Papers

8:05

2aPA1. Radiation pressure of standing waves on liquid columns and small diffusion flames. David B. Thiessen, Mark J. Marr-Lyon, Wei Wei, and Philip L. Marston (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814)

The radiation pressure of standing ultrasonic waves in air is demonstrated in this investigation to influence the dynamics of liquid columns and small flames. With the appropriate choice of the acoustic amplitude and wavelength, the natural tendency of long columns to break because of surface tension was suppressed in reduced gravity [M. J. Marr-Lyon, D. B. Thiessen, and P. L. Marston, Phys. Rev. Lett. **86**, 2293–2296 (2001); **87**(20), 9001(E) (2001)]. Evaluation of the radiation force shows that narrow liquid columns are attracted to velocity antinodes. The response of a small vertical diffusion flame to ultrasonic radiation pressure in a horizontal standing wave was observed in normal gravity. In agreement with our predictions of the distribution of ultrasonic radiation stress on the flame, the flame is attracted to a pressure antinode and becomes slightly elliptical with the major axis in the plane of the antinode. The radiation pressure distribution and the direction of the radiation force follow from the dominance of the dipole scattering for small flames. Understanding radiation stress on flames is relevant to the control of hot fluid objects. [Work supported by NASA.]

8:30

2aPA2. Foam rheology in the wet and dry limits. J. Gregory McDaniel, R. Glynn Holt (Aerosp. and Mech. Engr. Dept., Boston Univ., 110 Cummington St., Boston, MA 02115), and Iskander Sh. Akhatov (Bashkir State Univ., 450000 Ufa, Russia)

Understanding the rheological behavior of wet foams is important as a basic problem in fluid physics, and as a practical problem in many industries. This lecture will describe research into the wet and dry limits of foam rheology by a relatively new experimental technique in which foam drops are acoustically levitated and driven into motion. In the dry limit, the drops behave as viscoelastic solids. The effective moduli of the foam are estimated by observing the resonances of the drops and matching them to an analytical model for viscoelastic sphere vibrations. Analytical explorations of the wet limit have proceeded by considering the dynamics of a single bubble in a volume of liquid determined by the foam's void fraction. The linearized result is a wave equation, from which the natural frequencies and mode shapes of wet foam drops are determined. Relationships between this wave equation and those of classical effective medium theories will be described. [Work supported by NASA.]

8:55

2aPA3. Airborne chemistry single cell level. Staffan Nilsson, Peter Viberg, Peter Spegel, Sabina Santesson (Tech. Analytical Chemistry, Lund Univ., P.O. 124, SE-221 00 Lund, Sweden), Eila Cedergren, Eva Degerman, Tomas Johansson, and Johan Nilsson (Lund Univ., Lund, Sweden)

A miniaturized analysis system for the studying of living cells and biochemical reactions in microdrops was developed. Cell studies were performed using single adipocytes in 250-nL drops. Continuous flow-through droplet dispensers, developed in-house, were used for additions to the levitated droplet. Addition of b-adrenergic agonists stimulates the lipolysis in the adipocytes, leading to free fatty acid release and a consequent pH decrease of the surrounding buffer, a change that can be easily followed using a pH-dependent fluorophore continuously monitored by fluorescence imaging detection. An analytical method using capillary electrophoresis and nanospray mass spectrometry for measurement of the cAMP level in activated single adipocytes are now being developed for future use in combination with the levitation technique. The levitation approach was also employed for the screening of nucleation conditions for macromolecules. Here, the acoustic levitator offers a simplified way to determine the main features of the phase diagram (i.e., precipitation diagram). Using the droplet dispensers, different types and amounts of precipitation agents are injected into the levitated drop, allowing a systematic search for nucleation conditions that is not possible using standard crystallization methods. Once the precipitation diagram has been obtained, optimization using standard methods is employed to grow the crystals.

9:20

2aPA4. The role of bubbles and cavitation in the production of thermal lesions from high-intensity focused ultrasound. Ronald A. Roy, R. Glynn Holt, Xinmai Yang, and Patrick Edson (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, ronroy@bu.edu)

Rapid hyperthermia resulting in tissue necrosis is a key physical mechanism for focused ultrasound surgery (FUS). At therapeutic intensities, tissue heating is often accompanied by cavitation activity. Although it is well known that bubbles promote mechanical damage, *in vitro* and *in vivo* experiments have shown that under certain conditions bubble activity can double the heating rate. With a view towards harnessing bubbles and cavitation for useful clinical work, we report the results of *in vitro* experiments and modeling for the dynamic and thermal behavior of bubbles subjected to 1-megahertz ultrasound at mega-pascal pressures. The dominant bubble-related heating mechanism depends critically on the bubble size distribution which, in turn, depends on insonation control parameters (acoustic pressure, pulse duration), medium properties (notably dissolved gas concentration), and bubble-destroying shape instabilities. The evidence points to a range of control parameters for which bubble-enhanced FUS can be assured. [Work supported by DARPA and the U.S. Army.]

9:45–10:00 Break

Contributed Papers

10:00

2aPA5. The effect of dissolved gas concentration on bubble-enhanced heating in tissue-mimetic materials. Xinmai Yang, Ronald A. Roy, and R. Glynn Holt (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

Bubble-enhanced heating is a key mechanism to cause tissue damage in ultrasound surgery. We have conducted experiments in an agar-based tissue phantom. We found that the difference of air concentration in the tissue phantom has a small but measurable effect on the enhanced heating. Notably, high air concentration samples exhibit very good repeatability. We have passively monitored broadband acoustic emissions from the bubbles in order to determine if diagnostic information could be gleaned from such signals. Finally we investigate the effect of bubble size distribution on bubble-enhanced heating by employing bubble-based contrast agents to control the initial bubble size distribution. [Work supported by DARPA and the U.S. Army.]

10:15

2aPA6. A theoretical model for bubble enhanced ultrasound heating due to time-dependent bubble size distributions. Yang Xinmai, R. Glynn Holt, Patrick Edson, and Ronald A. Roy (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, xmyang@bu.edu)

Substantial *in vitro* and *in vivo* evidence shows that cavitation activity can affect tissue heating in focused ultrasound surgery and acoustic hemostasis applications. In particular, the heating rate in tissue increases significantly after cavitation sets in. Exploitation of this phenomenon for clinical use requires knowledge of, among other parameters, the time-dependent bubble size distribution sustained during insonation. Difficulties associated with the measurement of bubble sizes during *in vitro* or *in vivo* experiments call for a theoretical approach to the problem. We will present a theoretical model that estimates the time-dependent distribution of bubble equilibrium radii. Shape instability thresholds and rectified diffusion thresholds bound asymptotic bubble size distributions, and the instan-

taneous size distributions are governed by growth rates. The temperature rise caused by such bubble activity is calculated and compared with experimental data. [Work supported by DARPA and the U.S. Army.]

10:30

2aPA7. Low-power, cylindrical, air-coupled acoustic levitation/concentration devices: Symmetry breaking of the levitation volume. Gregory Kaduchak, Aleksandr S. Kogan, Christopher S. Kwiatkowski, and Dipen N. Sinha (Los Alamos Natl. Lab., MS D429, Los Alamos, NM 87545)

A cylindrical acoustic device for levitation and/or concentration of aerosols and small liquid/solid samples (up to several millimeters in diameter) in air has been developed [Kaduchak *et al.*, Rev. Sci. Instrum. **73**, 1332–1336]. It is inexpensive, low-power, and, in its simplest embodiment, does not require accurate alignment of a resonant cavity. It is constructed from a cylindrical PZT tube with thickness-to-radius ratio h/a ~0.03. The novelty of the device is that the lowest-order breathing mode of the tube is tuned to match a resonant mode of the interior air-filled cylindrical cavity. A high-Q cavity results that is driven very efficiently; drops of water in excess of 1-mm diameter are levitated for approximately 100 mW of input electrical power. The present research addresses modifying the different spatial configurations of the standing wave field within the cavity. By breaking the cylindrical symmetry, it is shown that pressure nodes can be localized for collection or separation of aerosolds or other particulate matter. Several different symmetry-breaking configurations are demonstrated. It is shown that experimental observations of the nodal arrangements agree with theoretical predictions.

10:45

2aPA8. Static and oscillatory response measurements of acoustically levitated foam drops. Li Liu, J. Gregory McDaniel, and R. Glynn Holt (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

Small samples of aqueous foam of varying gas volume fraction are acoustically levitated in an ultrasonic field. The drops are subjected to both static and time-varying pressures. Normal mode frequencies and inferred rheological properties (yield stress, shear modulus) for foams as a function of gas volume fraction will be presented. We compare the experimental results to recent theoretical descriptions of such modal oscillations [McDaniel and Holt, Phys. Rev. E **61**, 2204 (2000); McDaniel, Akhatov, and Holt, Phys Fluids **14**, 1886 (2002)]. [Work supported by NASA.]

11:00

2aPA9. Transport parameters for pulsed ultrasonic waves propagating in an aluminum foam. Arnaud Tourin, Arnaud Derode, Victor Mamou, Mathias Fink (LOA, ESPCI 10, rue Vauquelin 75005 Paris, France), John Page, and Michael L. Cowan (Univ. of Manitoba, Winnipeg, MB R3T 2N2, Canada)

Aluminum foams have now been studied for many years in large part because of their applications as light-weight elastic materials (e.g., car bumpers, aerospace engineering applications). The pore size and the spatial distribution of the pores govern the mechanical behavior of the foam and can vary enormously depending on the method of manufacturing. Thus, new methods for the nondestructive characterization of these materials are needed. We present here a set of experimental ultrasonic methods in a range of frequencies where the ultrasonic waves are multiply scattered in the medium. In this regime, the propagation is described by ultrasonic transport parameters which are related to the microstructure of the foam. The diffusion coefficient and the absorption mean free path have been determined in pulse transmission experiments by fitting the solution of the diffusion equation to the average intensity, the so-called time of flight distribution. To more fully characterize the medium, the transport mean path and the diffusion coefficient have been measured in backscattering experiments using the static and dynamic coherent backscattering effects. For both methods, the properties of the sample interfaces have been taken into account.

11:15

2aPA10. Low frequency cavitation erosion. Sally J. Pardue and Gautam Chandekar (Dept. of Mech. Eng., Tennessee Technolog. Univ., Box 5014, Cookeville, TN 38505, spardue@tntech.edu)

Damage of diesel engine piston sleeve liners due to cavitation of the coolant fluid can be severe. Coolant fluid additives are used to inhibit cavitation damage, and are evaluated by industry suppliers using ASTM G32-98 Standard Test Method for Cavitation Erosion Using Vibratory Apparatus. The ASTM G32-98 test procedure uses an ultrasonic horn at 20 kHz to vibrate a test button in the coolant. The test button mass loss and surface appearance are studied to sort the performance of new coolant additives. Mismatch between good lab performers and actual engine test runs has raised concerns over the current lab test. The frequency range of the current test has been targeted for investigation. A low frequency, less than 2000 Hz, test rig was built to explore the cavitation damage. The test system did produce cavitation on the surface of the test button for a period of 36 h, with minimal mass loss. The test rig experienced cyclic fatigue when test times were extended. The work is now focusing on designing a better test rig for long duration tests and on developing numerical models in order to explore the effects of cavitation excitation frequency on surface erosion.

11:30

2aPA11. A novel cavitation probe design and some preliminary measurements of its application to megasonic cleaning. Lawrence A. Crum (Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105) and Gary Ferrell (SEZ America, Inc., Mountain View, CA 94043)

An initial prototype design for a cavitation probe that uses the property of a collapsing cavitation bubble to produce visible photons (sonoluminescence) has been designed and constructed. These light emissions can be easily detected within a small, finite volume and thus this probe provides a direct means of measuring the cavitation density (activity/per unit volume) within a cavitating fluid and the delivery of ultrasonic energy at an engineered surface. As a result, ultrasonic methods treating a surface can be directly monitored and controlled in real-time, leading to the ability to improve and predict the performance of the resulting structure. This probe provides the potential for constructing a real-time monitor of ultrasonic/megasonic cleaner efficiency and effectiveness. In addition, because the entire three-dimensional cavitation field can be measured with this probe, it can also serve as a useful tool in ultrasonic/megasonic cleaner design. A real-time cavitation-density measuring device would have great utility in the semiconductor cleaning industry and thus this probe provides considerable promise for commercial development. A description of the probe will be presented as well as some preliminary data on cavitation density within a commercial megasonic cleaner. [Work supported in part by the NSF.]

11:45

2aPA12. Bubble dynamics in an acoustic flow field. Dmitry V. Voronin, Georgij N. Sankin (Lavrentyev Inst. of Hydrodynamics, Prosp. Acad. Lavrentyeva, 15, Novosibirsk 630090, Russia, voron@hydro.nsc.ru), Robert Mettin (Universitaet Goettingen, 37073 Goettingen, Germany), Vyacheslav S. Teslenko (Lavrentyev Inst. of Hydrodynamics, Novosibirsk 630090, Russia), and Werner Lauterborn (Universitaet Goettingen, 37073 Goettingen, Germany)

Dynamics of interaction between cavitational bubbles is investigated when a complex of a compression and a rarefaction pulse passes through a liquid with pre-existing micro bubbles. Cavitation was generated experimentally with the help of electromagnetic generator of a flat and a convergent acoustic pulse (2-μs duration, 1–20 MPa) having the form of a hollow sphere segment. A modeling was performed within the frame of two-dimensional axisymmetric nonstationary approach on the basis of conservation laws for a model of an ideal compressible liquid. A thermodynamic flow field was computed both in liquid and inside bubbles. Behind the rarefaction wave the microbubbles begin to grow and generate secondary compression shocks, the amplitude of which may exceed that of

the incident pulse under certain conditions. It is shown that the process of bubble interaction within a cluster is accompanied by bubble coalescence, fragmentation, and collapse of the initial bubble or its fragments. Simultaneously, high temperature spots appear in the bubble compressing by the secondary wave. Adiabatic heating of gas either inside a bubble or near the neck between a bubble and its fragment may result in sonoluminescence, also observed in experiments. [Work supported by ASA, DAAD, and RFBR.]

<div align="center">TUESDAY MORNING, 3 DECEMBER 2002   CORAL GALLERY FOYER, 8:00 TO 11:45 A.M.</div>

<div align="center">Session 2aPP</div>

<div align="center">Psychological and Physiological Acoustics: General Topics in Psychological Acoustics</div>

<div align="center">Jont B. Allen, Cochair

Mimosa Acoustics, 382 Forest Hill Way, Mountainside, New Jersey 07092

Rodrigo Ordoñez, Cochair

Department of Acoustics, Aalborg University, Fredrik Bajers Vej 7 B4, DK-9220 Aalborg, Denmark</div>

<div align="center">*Contributed Papers*</div>

8:00

2aPP1. Attentional focus and the method of adjustment revisited. Charles S. Watson, Gary R. Kidd, and Soriya V. Pok (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405, watson@indiana.edu)

In past reports we have described a technique by which listeners may be trained to focus their auditory attention on a particular spectral-temporal region of a complex acoustic stimulus, using the psychophysical method of adjustment. Previous work will be reviewed, and the results of a new experiment will be described, in which listeners were trained under both the adjustment method and a standard adaptive tracking technique to detect changes in the frequency of a tone within a nine-tone sequence. Under some circumstances, the adjustment procedure can enable listeners to learn to detect very small changes within a few minutes, whereas several hours of training under adaptive methods may be required to achieve the same detection or discrimination performance. Other differences between the two methods will be described. [Work supported by the NIH/NIDCD.]

8:15

2aPP2. Psychophysical analysis of sound and vibration in the cabin of passenger aircrafts. Volker Mellert, Ingo Baumann, Nils Freese, Roland Kruse, Reinhard Weber (Dept. of Phys., Oldenburg Univ., 26111 Oldenburg, Germany), Hermann Remmers, and Michael Bellmann (ITAP GmbH, 26129 Oldenburg, Germany)

The vibroacoustics within the fuselage of several types of aircrafts is recorded with microphones, ear-related devices, and accelerometers at different locations of passengers seats, and the workplace of the cabin and cockpit crew. The signals are analyzed according to standard psychoacoustic and vibration parameters. The requirements for the reproduction of the signals in a ground-based test-bed (e.g., mock-up) are identified. Results are reported on how well test facilities at ground meet real-flight conditions. [Work supported by the European Community (www.heace.org).]

8:30

2aPP3. Distinguishing sound from noise—The significance of attention and noise sensitivity. George Dodd (Acoust. Res. Ctr., Univ. of Auckland, Private Bag 92019, Auckland, New Zealand, g.dodd@auckland.ac.nz)

There is often a large discrepancy between the accuracies of physical measurements and the precision ascribed to subjective responses to sound. Consequently the criteria by which sound and noise are assessed can ap-pear somewhat loose. In previous work it was proposed that there is a need to formalize definitions for noise and non-noise sounds in order to render subjective reactions more readily quantifiable. This is necessary to give greater recognition to the significance of differences in individuals responses, and also to put criteria for environmental sound on a more scientific basis. This presentation reviews our research looking for physiological responses which correlate with a person's attention to sound, and presents results from our study of noise sensitivity. In this work noise sensitivity is defined as being a tendency to be distracted by sound and is viewed as a stable characteristic of people differing between individuals and distinct from noise annoyance experienced at a particular time. The results of assessing noise sensitivity by self-assessment questionaires and other measures are presented and how they relate to individual listening habits.

8:45

2aPP4. Spectral pattern and harmonic relations as factors governing the perceptual cohesion of low-numbered components in complex tones. Brian Roberts (School of Psych., Univ. of Birmingham, Edgbaston, Birmingham B15 2TT, UK, B.Roberts@Bham.ac.uk) and Jeffrey M. Brunstrom (Loughborough Univ., Loughborough, Leicestershire LE11 3TU, UK)

Mistuning a harmonic changes its pitch more than expected and also increases its salience. Both effects can be used to explore the auditory organization of complex tones. In this study we extend our findings indicating that the effects of a spectral pattern on grouping are not restricted to harmonic relations. Stimuli were either harmonic ($F0 = 200$ Hz) or frequency shifted by 25% of $F0$. Component 1 or 2 was replaced by one of a set of sinusoidal probes in the same spectral region. In experiment 1, listeners adjusted the frequency of a pure tone to match the probe pitch. Inflections of the pitch-shift functions were close to the two expected values in the harmonic condition. In the shifted condition, they were close to the suboctave (225 Hz) and the frequency (450 Hz) of component 2. In experiment 2, listeners matched the probe loudness by adjusting the level of a tone of identical frequency. Loudness minima corresponded closely to the pitch-shift inflections. Experiment 3 showed that a pitch-shift inflection close to 450 Hz requires the presence of component 1. These results suggest that the first component of a shifted complex is grouped differently from the others, based on harmonicity rather than spectral spacing.

9:00

2aPP5. Sound localization: Effects of gender, aging, head related transfer function, and auditory performance. Pedro Menezes, Ilka Soares, Silvio Caldas Neto, and Mauricy Motta (UFPE, UNCISAL, Av. Prof. Moraes Rego, 25-Cidade Universitria, Recife, PE, Brasil. Cep. 50960.870, son@hotlink.com.br)

The sound localization resolution in 80 normal hearing subjects of both sexes will be compared to audiometric parameters, audiometry, timpanometry, and stapedian reflexes. Moreover, the pinna and concha length and width so as the interaural distance, ear–head angle, sex, and age were compared too, in a reverberating room. Three tones of square waves with 1, 2, and 3 kHz, were present with an intensity of 70-dB SPL in order of the speakers and sequence of frequencies randomized. The subjects were trained to indicate the origin of the sound in a control console with push-buttons representing the space disposition of the speakers, being 8 in the horizontal plane, 5 in the medial sagital plane, and 5 in the medial frontal plane. The identification of each speaker is done by pressing the respective push-button. The angle between the speaker's axis in the same plane is of 45, and the distance of 1 meter from the analyzed subject. The preliminary results from 50 subjects showed a better localization precision at 1 kHz frequency, without sex or age predominance. The medial sagital plane presented more errors, in all the frequencies, according to the specialized literature. More subjects are actually being and soon they will be available. (To be presented in Portuguese.)

9:15

2aPP6. Visual bias on sound location modulated by content based processes. Ilja Frissen and Beatrice de Gelder (Tilburg Univ., Tilburg, The Netherlands)

Ventriloquism refers to a perceptual phenomenon in which the apparent location of a sound source is displaced in the direction of a synchronous but spatially disparate visual stimulus. It is generally accepted that spatial and temporal proximity are factors facilitating crossmodal integration. Here we investigate whether content based processes could also play a role. In order to control for strategic factors, a psychophysical staircase method (Bertelson and Aschersleben, 1998) was adopted. Auditory stimuli were digital recordings of vowels (/i/ and /o/). Visual stimuli were digital pictures of talking faces articulating the same vowels, and a scrambled face. We ran eight concurrent staircases. In half of these the auditory stimuli were paired with the corresponding face, and in the other half with the scrambled face. Half the staircases started from the extreme left and the other from the extreme right. Presentation of staircases was randomized. Participants were asked to judge whether the sound was coming from the left, or from the right, of the median plane. On the staircases with a face stimulus, reversals started to occur significantly earlier than with a nonface. Thus, a ''realistic'' stimulus pairing enhances crossmodal integration.

9:30–9:45 Break

9:45

2aPP7. Auditory localization in the horizontal plane with single and double hearing protection. Douglas S. Brungart (Air Force Res. Lab., Wright–Patterson AFB, OH 45433), Alexander J. Kordik (Sytronics, Inc., Dayton, OH 45432), and Brian D. Simpson (Veridian, Dayton, OH 45431)

Although most occupational noise problems can be adequately addressed with either earplugs or earmuffs, some extreme noise environments require listeners to wear both earplugs and earmuffs at the same time. However, little is known about the impact that double hearing protection has on sound localization. In this experiment, listeners wearing single and double hearing protection were asked to localize pink noise signals originating from 24 evenly spaced loudspeakers in the horizontal plane. In the single hearing protection conditions, localization accuracy was severely degraded with a short (250 ms) stimulus, but only modestly degraded with a continuous stimulus that allowed listeners to make exploratory head movements. In the double hearing protection conditions, localization accuracy was near chance level with the short stimulus and was only slightly better than chance with the continuous stimulus. A second experiment showed that listeners wearing double hearing protection were routinely unable to identify the lateral positions of stimuli originating from loudspeakers located at $\pm 45°$ in azimuth. The results suggest that double hearing protection reduces the air-conducted signals in the ear canals to the point that bone and tissue conducted signals disrupt the interaural difference cues listeners normally use to localize sound. [Work supported by AFOSR.]

10:00

2aPP8. Standardization of infrasounds and low-frequency noises for health benefits on humans. Zbigniew Damijan (Structural Acoust. and Biomed. Eng. Lab., Staszic Univ., AGH Krakow, Poland), Ryszard Panuszka (Staszic Univ., AGH Krakow, Poland), and James McGlothlin (Purdue Univ., West Lafayette, IN)

Annoyances from low-frequency noises and infrasound are effects of using technologies, transportation, manufacturing equipment, and large air-condition systems. Present developed procedures for evaluations of annoyance of low-frequency noise (LFN) on humans and occupational health are based on international and national standards. Comparisons show that there is a large difference between permitted values of acoustic pressure levels used. These standards are based on levels of thresholds heard by the human auditory system and subjective observations of impacts from vibrations of infrasound waves on the human body. A newly discovered phenomenon shows a follow-up effect in the brain and is an important reason to check and investigate attenuation of infrasound, especially near and below 10 Hz. Previous investigations showed that nonlinear reaction is observed according to the external infrasound field pressure on the human body. New studies need to investigate electrical reactions of the human brain and electrodermal reactions by influences of infrasound. [Work supported by the Kosciuszko Foundation, Inc., an American Center for Polish Culture, with Funding provided by the Alfred Juzykowski Foundation and KBN Warsaw.]

10:15

2aPP9. Gap detection and location in the precedence effect. Liang Li and Bruce A. Schneider (Dept. of Psych., Univ. of Toronto at Mississauga, Mississauga, ON L5L 1C6, Canada)

The nature of the precedence effect was investigated by introducing a gap into (1) both the leading and lagging sounds, (2) the lagging but not the leading sound, and (3) the leading but not the lagging sound. When a 50-ms gap was introduced into both sounds with an onset asynchrony equal to the delay between the leading and lagging sounds, the gap was perceived to occur only on the leading side as long as the delay between leading and lagging sounds did not exceed approximately 15 ms, even though the precedence effect itself broke down when the delay between the leading and lagging sounds exceeded approximately 9 ms. Gaps presented only in lagging sounds were always heard as occurring in the source position of the leading sound, but no gaps were perceived when the gap occurred only in the leading sound, rather, the listener heard a noise burst from the position of the lagging (suppressed) sound. The present results indicate that gaps in the lagging sound are perceived as belonging

to the leading sound, whereas gaps in the leading sound release the lagging sound from "echo suppression," indicating that higher-order (top–down) processes are involved in the precedence effect.

10:30

2aPP10. Artificial environment mapping from acoustic information. Rodolfo Martinez (CIIDIR, IPN, OAXACA, Mexico, rodolfo_mc@yahoo.com) and Sergio Beristain (Acoustics Lab., ESIME, IPN, Mexico)

Living creatures have the capacity to build maps of the surrounding world from their hearing or ultrasonic perception systems. This mapping is generally assumed to allow survival within a complex environment. When this ability is applied to artificial life or artificial intelligence, the problem becomes very complex. This paper describes some reference models for several species.

10:45

2aPP11. Pitch and loudness memory in musicians and nonmusicians. Peter Bailey and Stuart Dobinson (Dept. of Psych., Univ. of York, York YO10 5DD, UK, pjb1@york.ac.uk)

The finding that pitch and loudness traces decay at different rates [S. Clement, L. Demany, and C. Semal, J. Acoust. Soc. Am. **106**, 2805–2811 (1999)] is one of several results indicating that the processes of pitch and loudness memory may be distinct. A speculation raised by Clement *et al.*, among others, is that these specialized memory subsystems might be differently influenced by musical experience. To explore this hypothesis, difference limens (DLs) for the fundamental frequency (DLF) and intensity (DLI) of complex tones were measured for groups of musically-experienced and musically-naive participants, using a roving-standard, 2-interval procedure in which the duration and content of the interstimulus interval (ISI) within a trial were manipulated. In the first experiment ISIs were silent and 0.5 s or 2.0 s in duration. For both groups DLs increased with ISI; DLFs were smaller for musicians than nonmusicians at both ISIs, but DLIs were smaller for musicians only when ISI=0.5 s, and did not differ when ISI=2.0 s. In a second experiment the ISI included interpolated tones. DLFs were larger for nonmusicians than musicians, but DLIs were similar for both groups. The results suggest that musical experience has different effects on memory fo rpitch and loudness.

11:00

2aPP12. Effect of envelope lowpass filtering on consonant and melody recognition. Arthur P. Lobo, Felipe Toledos, Philip C. Loizou (Dept. of Elec. Eng., Univ. of Texas, Dallas, Richardson, TX 75083), and Michael F. Dorman (Arizona State Univ., Tempe, AZ 75827)

Recent work [Smith *et al.*, Nature **416**, 87–90 (2002)] has shown that the speech envelope contains fine temporal information which is used in pitch perception and spatial localization. This study was performed on normal hearing subjects. In this paper, we investigated the effect of lowpass filtering of the envelope on consonant and melody recognition in subjects using the Clarion cochlear implant. The subjects were originally fitted with the simultaneous analog stimulation (SAS) speech processing strategy, a strategy known to provide fine time-envelope information. The consonants and instrumental music were bandpass filtered into seven channels and the envelope of each channel was lowpass filtered with cutoff frequencies ranging between 100 and 1200 Hz. Initial results on the consonant recognition task showed that some subjects performed equally well for all envelope cutoff frequencies. On the melody recognition task, some subjects performed best at a particular envelope cutoff frequency. Results for the full set of subjects who participated in this study will be presented. [Work supported by NIH.]

11:15

2aPP13. Visual speech recalibrates auditory speech identification. Paul Bertelson (Universite Libre de Bruxelles, Bruxelles, Belgium), Jean Vroomen, and Beatrice de Gelder (Tilburg Univ., The Netherlands)

Exposure to spatially incongruent auditory and visual inputs produces both immediate crossmodal biases and aftereffects. But for event identification, rather than localization, only biases have been demonstrated so far. Taking the case of incongruent audiovisual speech, which produces the well-known McGurk bias effect, we show that, contrary to earlier reports (e.g., Roberts and Summerfield, 1981), aftereffects can be obtained. Exposure to an ambivalent auditory token from an /aba–ada/ continuum combined with the visual presentation of a face articulating /aba/ (or /ada/) increased the tendency to interpret test auditory tokens as /aba/ (or /ada/). The earlier results that were taken as disproving the possibility of visual recalibration of auditory speech identification were obtained with exposure to nonambiguous auditory tokens that (as we confirm in another experiment) create an auditory contrast effect in a direction opposite that of recalibration, and presumably masked the effect of recalibration.

11:30

2aPP14. Factors affecting frequency discrimination by poor readers. Peter Bailey, Maggie Snowling (Dept. of Psych., Univ. of York, York YO10 5DD, UK, pjb1@york.ac.uk), Yvonne Griffiths, and Nick Hill (Univ. of Essex, Colchester CO4 3SQ, UK, ymg@essex.ac.uk)

Some of the experiments on frequency discrimination by groups of poor readers have shown impairments relative to normal-reading control groups, while other experiments have shown no reliable group differences. It remains uncertain to what extent these differences in outcome are attributable to individual differences in the severity of a sensory processing deficit associated with dyslexia, or to use different psychophysical procedures that make different demands of higher-level cognitive processes (such as attention and memory) which may be compromised by dyslexia. To explore these issues, pure-tone frequency difference limens (DLFs) were measured for groups of dyslexic adults and normal-reading controls in conditions incorporating a range of procedural manipulations. Dyslexic and control participants' DLFs did not differ reliably when the procedure involved four-interval trials, but dyslexics' DLFs were larger than controls using two-interval trials. The *relative* difference between dyslexic and control participants' DLFs found using two-interval trials did not differ systematically across conditions involving a fixed or roving standard frequency, long or short duration stimuli, long or short interstimulus intervals, or interstimulus intervals that were either silent or included interpolated tones. The results suggest no obvious link between elevated DLFs and impaired short-term pitch memory in these dyslexic participants.

2a TUE. AM

Session 2aSAa

Structural Acoustics and Vibration: Analysis, Measurements, and Control of Structural Intensity

Sabih I. Hayek, Chair

Department of Engineering Science and Mechanics, Pennsylvania State University, State College, Pennsylvania 16802-6812

Chair's Introduction—8:00

Invited Papers

8:05

2aSAa1. The intensity potential approach. Jean Louis Guyader and Michael Thivant (LVA, INSA de Lyon, 69621 Villeurbanne, France)

Sound intensity vectors can be decomposed into irrotational and curl components. The irrotational part describes intensity propagation, and the curl component vortices and near field of sources. To predict power flow one can limit the problem to irrotational intensity, that is to say to the determination of intensity potential. The resulting equation is analogous to heat transfer, permitting one to use standard heat transfer solvers to predict acoustic power flow. The intensity potential approach is presented for acoustic propagation from sources in cavities with apertures. The modelization of absorbing materials through thermal convection factor is discussed. Finally, comparison with exact prediction and experimental results are presented.

8:35

2aSAa2. Active control of structural intensity and radiated acoustic power from an infinite point-excited submerged Mindlin plate. Jungyun Won and Sabih Hayek (Active Vib. Control Lab., Dept. of Eng. Sci. and Mech., 212 EES Bldg., Penn State Univ., University Park, PA 16802)

In this paper, the active vibrational structural intensity (VSI) in, and the radiated acoustic power from an infinite elastic plate in contact with a heavy fluid is modeled by the Mindlin plate theory. The plate is excited by a point force, which generates a vector-active VSI field in the plate. The resulting acoustic radiation generates an active acoustic intensity (AI) in the fluid medium. The displacement, shear deformation, VSI vector map, radiated acoustic pressure, and the AI vector map are computed. One, two, or four synchronous point controllers are placed symmetrically with respect to the point force on the plate. Minimization of either the structural intensity at a reference point or the total radiated acoustic power is achieved. Below coincidence, a significant portion of the point force input power is trapped in the plate in the form of VSI. The total radiated power is calculated by use of the input power from the source, the controllers, and the VSI. Above coincidence, a significant portion of the input source power is leaked to the fluid in the form of AI, so that the acoustic radiated power is equal to the input power from the source and the controllers.

Session 2aSAb

Structural Acoustics and Vibration: Vibration and Noise Control

J. Gregory McDaniel, Chair

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

Contributed Papers

9:15

2aSAb1. Active control of cantilever-beam vibration. M. Roman Serbyn (Morgan State Univ., Baltimore, MD 21251, rserbyn@morgan.edu)

A bang–bang control system previously developed for the stabilization of a rigid platform [ISA Trans. **21**, 55–59 (1982)] has been adapted to the problem of reducing flexural vibrations of a beam. The electromechanical system develops an appropriate control signal for the actuator from samples of the disturbance by analog and digital signal processing using integrated circuits. The effectiveness of this approach is predicated upon the sampling rate being much higher than the maximum vibration frequency to be silenced. It is also robust with respect to the waveform of the disturbance. Noise reductions of 10–20 dB have been achieved, depending on the bandwidth of the noise. The cantilever, chosen because of its mechanical and theoretical simplicity, provides a good foundation for the study of more complex structures, like airfoils and nonrigid platforms. In both experimental and analytical investigations the emphasis has been on

the optimization of control parameters, particularly with regard to the application of the cancellation signal. Reduction in size and cost of the control unit is possible by incorporating the latest technological advances in electronic and electromechanical devices, such as FPGA boards and MEMS components.

9:30

2aSAb2. An active structural acoustic control approach for the reduction of the structure-borne road noise. Hugo Douville, Alain Berry, and Patrice Masson (Dept. of Mech. Eng., Universite de Sherbrooke, 2500 Boul. Universite, Sherbrooke, QC J1K 2R1, Canada)

The reduction of the structure-borne road noise generated inside the cabin of an automobile is investigated using an Active Structural Acoustic Control (ASAC) approach. First, a laboratory test bench consisting of a wheel/suspension/lower suspension A-arm assembly has been developed in order to identify the vibroacoustic transfer paths (up to 250 Hz) for realistic road noise excitation of the wheel. Frequency Response Function (FRF) measurements between the excitation/control actuators and each suspension/chassis linkage are used to characterize the different transfer paths that transmit energy through the chassis of the car. Second, a FE/BE model (Finite/Boundary Elements) was developed to simulate the acoustic field of an automobile cab interior. This model is used to predict the acoustic field inside the cabin as a response to the measured forces applied on the suspension/chassis linkages. Finally, an experimental implementation of ASAC is presented. The control approach relies on the use of inertial actuators to modify the vibration behavior of the suspension and the automotive chassis such that its noise radiation efficiency is decreased. The implemented algorithm consists of a MIMO (Multiple-Input–Multiple-Output) feedforward configuration with a filtered-X LMS algorithm using an advanced reference signal (width FIR filters) using the Simulink/Dspace environment for control prototyping.

9:45

2aSAb3. Active control of noise radiated through rectangular plates using piezeletric patches. Danuza Cristina Santana, Marcus Antonio Viana Duarte, and Domingos Alves Rade (School of Mech. Eng., Federal Univ. of Uberlandia, P.O. Box 593, CEP 38400-902 Uberlandia, MG, Brazil)

Due to problems caused by noise in industrial environment and in human daily life, techniques of noise control have received increasing attention from engineers and researchers lately. More recently, the use of piezeletric elements as sensors and/or actuators in noise and vibration control systems has been extensively investigated. The main advantage of the use of such devices is that they can be easily integrated to the mechanical system with little added mass and relatively high control authority. The present paper addresses a technique of active control of sound transmitted through a rectangular, thin, simply supported plate by employing multiple piezeletric patches bonded to the plate's surface. A harmonic plane wave incident on one side of the plate is considered to be the primary noise source. Aiming at minimizing the noise transmitted to the other side of the plate, bending motion is induced through the piezeletric patches so that the plate behaves as a secondary sound source. The paper brings the development of the system mathematical model which enables to obtain the spatial distribution of sound pressure radiated through the plate in the far field. An optimal control technique providing the voltage control signals for the activation of the piezoelectric patches is presented, based on the minimization of a cost function representing the mean square integral of the sound pressure radiated in a semi-sphere in far field. It is also proposed a methodology for the optimal placement on the piezelectric patches using Genetic Algorithms. (To be presented in Portuguese.)

2aSAb4. Mechanical realization of passive scalar transfer functions. Pierre E. Dupont and Wenyuan Chen (Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, pierre@bu.edu)

There are typically an infinite number of mass-spring-damper systems that share the same input–output characteristics as described by a drive-point acceleration (passive) transfer function. The realization problem of solving for one or more of these mechanical systems arises in several application areas. These include the scaled acoustic testing of complex ship systems as well as the design of mechanical filters. The theory for scalar transfer functions involving no damping or proportional damping is well known. The family of solutions includes realizations consisting of masses connected entirely in parallel or in series; and algorithms are available for computing the components values. The assumption of proportional damping is purely for mathematical convenience, however, and does not usually concur with the reality of experimental data. This talk addresses the mechanical realization problem for scalar acceleration transfer functions with arbitrary viscous damping. A time domain approach is utilized to obtain a parameterized description of the solution set. Numerical methods are presened which can be used to search the solution set for realizations satisfying criteria relating to ease of fabrication, e.g., involving the fewest components. Laboratory experiments are included to validate the approach. [Work supported by ONR.]

10:15

2aSAb5. A vibration technique for measuring railroad rail stress. Vesna Damljanovic and Richard L. Weaver (Dept. of Theoret. & Appl. Mech., Univ. of Illinois, 104 S. Wright St., Urbana, IL 61801, damljano@uiuc.edu)

Longitudinal rail stress, related to constrained thermal contractions and expansions, leads to broken and buckled rails, and consequent service delays and derailments. There is a broad consensus on the need for cost-effective and reliable methods for the measurement of rail stress. Vibration techniques for assessing rail stress, based on the effect of longitudinal force on the free vibrations of beams, have long been proposed. It is well understood that compressive stresses decrease the flexural frequencies, while tensile stresses increase them. Past efforts attempting to use this for measurements of stress in railroad rails have failed due to an inability to control or adequately measure other parameters, most particularly the placement and stiffness of the supports. We are developing a new method in which the influence of support parameters should be minimal. Scanned laser vibrometry measurements of vibration fields, followed by a comparison with guided wave theory for the complex cross section of the rail, promises to allow the stress to be determined with the requisite precision. Here we report on the status of this work. [Work supported by Association of American Railroads and the Transportation Research Board.]

10:30

2aSAb6. Theoretical and experimental study of vibration, generated by monorail trains. Samuil A. Rybak (N. N. Andreev Acoust. Inst., Moscow 117036, Russia), Sergey A. Makhortykh (Russian Acad. of Sci., Pushchino, Moscow Region 142290, Russia), and Stanislav A. Kostarev (Lab. of Acoust. and Vib. Tunnel Assoc., Moscow 107217, Russia)

Monorail transport as all other city transport vehicles is the source of high noise and vibration levels. It is less widespread than cars or underground transport but its influence in modern cities enhances. Now in Moscow the first monorail road with trains on tires is designed, therefore the problem of vibration and noise assessments and prediction of its impact on the residential region appears. To assess the levels of generated vibration a physical model of interaction in the system wagon-tire-road coating-

2a TUE. AM

viaduct-soil has been proposed and then numerically analyzed. The model is based on the known from publications facts of automobile transport vibration and our own practice concerning underground trains vibration generation. To verify computer simulation results and adjust model param-eters the series of measurements of noise and vibration near experimental monorail road was carried out. In the report the results of calculations and measurements will be presented and some outcomes of possible acoustical ecologic situation near monorail roads will be proposed.

TUESDAY MORNING, 3 DECEMBER 2002

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

Session 2aSC

Speech Communication: Vowel Acoustics and Perception (Poster Session)

Michael J. Kiefte, Cochair

Human Communication Disorders, Dalhousie University, 5599 Fenwick Street, Halifax, Nova Scotia B3H 1R2, Canada

Gonzalo Corvera, Cochair

Monte Athos 116, 11000, D.F. Mexico

Contributed Papers

All posters will be on display from 8:30 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 8:30 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m.

2aSC1. Causes and consequences of the unequal distribution of vowels in the vowel space. Peter Ladefoged (Phonet. Lab., Linguist., UCLA, Los Angeles, CA 90095-1543)

A set of possible vocal tract shapes for vowels was generated, using a vocal tract model that operates in terms of the two tongue shape parameters determined by Harshman, Ladefoged, and Goldstein (1975) and one parameter of lip opening. Each of the tongue shape parameters was varied through 15 equal steps. Many of the combinations of the two parameters produced impossible distortions of the tongue or tongue shapes associated with consonants, but 147 of the 225 vowel shapes were humanly possible. These are 147 equally spaced vocal tract shapes as defined by the two parameters. Each of these tongue shapes was combined with seven degrees of lip rounding. When the first two formants of these 1029 vowels were plotted, some parts of this formant space are more densely populated than others. There are few vowels with low $F1$ and low $F2$. The difference among front vowels, such as those in heed, hid, head, had, can be made simply by varying the vocal tract shape, but the back vowels, such as those in hawed, hood, who'd, require added lip rounding. In the world's languages there is often an asymmetry in the height of front and back vowels.

2aSC2. Realization of the English [voice] contrast in $F1$ and $F2$. Elliott Moreton (Dept. of Cognit. Sci., Krieger Hall, Johns Hopkins Univ., Baltimore, MD 21218)

Before a [voice] coda, $F1$ is higher for monophthongs but lower for /aI/ than before [+voice]. We test the hypothesis that this is due to local hyperarticulation before voiceless obstruents. Experiment 1, with 16 American English speakers, found the /aI/ pattern of more peripheral $F1$ and $F2$ in the offglides /oI eI aU/ as well, showing that it is part of the realization of [voice] rather than a historical property of /aI/. Some of the $F2$ increase in /aI oI eI/ cannot be accounted for by articulatory raising alone, but must be ascribed to fronting. The diphthong nuclei tended to change in the same direction as the offglides. Experiments 2 and 3, each with a different 16 American English speakers, collected "tight"-"tide" (Exp. 2) or "ate"-"aid" (Exp. 3) judgments of a synthetic stimulus in which offglide $F1$, offglide $F2$, and nuclear duration were varied independently. "Tight" and "ate" responses were facilitated by lower $F1$, by

higher $F2$, and by shorter nuclei. Log-linear analysis showed that the three factors contributed independently, and that $F2$ was a stronger cue than $F1$ in terms of logits per Bark. Thus [voice] is correlated with, and cued by, peripheralization of diphthong offglides.

2aSC3. Influence of formant transitions and linguistic relevance on vowel imitation. Gautam K Vallabha and Betty Tuller (Ctr. for Complex Systems & Brain Sci., Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33431, vallabha@walt.ccs.fau.edu)

Speakers are unable to imitate the formants of isolated self-produced vowels accurately. The pattern of bias directions cannot be ascribed solely to articulatory or perceptual noise, and is different for each speaker [G. K. Vallabha and B. Tuller, J. Acoust. Soc. Am. **110**, 2657 (2001)]. The present experiment examines whether the bias pattern persists when the vowels are embedded in a CV syllable. A dialectically uniform group of male and female American English [AE] speakers produced vocalic sounds in three conditions: (1) they read a list of /hVd/ words 10 times each, (2) they imitated /V/ and /dV/ targets containing monophthongal AE vowels and (3) they imitated /V/ and /dV/ targets that were systematically distributed over the $F1 \times F2$ space. All targets were self-produced (hence perfectly reproducible in principle) and were imitated 10 times each. Results will be discussed in terms of the bias and variability of AE versus non-AE targets, the imitation bias of /dV/ vs /V/ targets, and whether the bias pattern differences among subjects are reduced by the dialect uniformity. [Work supported by NIMH.]

2aSC4. Context effects on the perception of vowel spectral properties. Michael Kiefte (School of Human Commun. Disord., Dalhousie Univ., 5599 Fenwick St., Halifax, NS B3H 1R2, Canada) and Keith R. Kluender (Univ. of Wisconsin, Madison, WI 53706-1696)

Previous studies [M. Kiefte and K. R. Kluender, J. Acoust. Soc. Am. **57**, 55–62 (2001)] have shown that long-term stationary spectral characteristics have a very strong influence on the perception of specific cues in monophthongs. For example, F_2 will be perceptually neglected if a precursor sentence is filtered by a one-pole filter corresponding to the center

frequency and bandwidth of the target vowel. The present work continues this research and presents results from experiments using nonspeech precursors. The first set of experiments use speechlike stimuli that have been LPC vocoded from a time-reversed sentence having similar spectro-temporal properties to natural speech. The second set of experiments use FM-formant glides as precursors to test multiple hypotheses including the time course of these effects over varying precursor durations. The second set of experiments also test the hypothesis that the duration of the precursor plays a role in the magnitude of the context effect on the perception of the following vowel.

2aSC5. Acoustic correlates of English glottal [t] allophone. Rina Kreitman (Dept. of Linguist., Cornell Univ., 203 Morrill Hall, Ithaca, NY 14850, rk83@cornell.edu)

This experiment investigates the acoustics of American English coda [t], which is often glottalized. A list of monosyllabic and bisyllabic words with matching vowel quality in target syllables across pairs was devised. The matching pairs were controlled for lexical frequency from the CELEX database. The words were read in isolation and in a frame sentence, where the target word was at the end of an intonational phrase. Results show that there is glottalization on the vowel preceding [t], which is interpreted as laryngeal coarticulation from the glottal allophone of [t]. Additionally, results show that for some speakers the proportion of the vowel that is glottalized is longer in sentences than in isolated words. Results of a cepstral measure of harmonics-to-noise-ratio (Hillenbrand *et al.,* 1994) are different from what was found for Ju–hoansi glottalized consonants (Miller-Ockhuizen, 2002), but similar to results for glottalized vowels. That is, Gamnitude of R1 increased toward the end of the vowel. Since glottalization, as measured via a low H1–H2 (NiChasaide and Gobl, 1997) and aperiodicity seen through waveform inspection, does occur on the vowel, this suggests an articulatory mechanism for glottalization different from constriction of the glottis, such as false vocal-fold contraction (Fujimura and Sawashima, 1971).

2aSC6. Effects of frequency shifts on vowel category judgments. Catherine M. Glidden, Peter F. Assmann, and Terrance M. Nearey (School of Human Development, Univ. of Texas, Dallas, Box 830688, Richardson, TX 75083)

To investigate the effects of fundamental frequency ($F0$) and formant frequency shifts on vowel identification, a high-quality vocoder ("STRAIGHT") was used to process the syllables "bit" and "bet" spoken by an adult female talker. From these two endpoints a nine-step continuum was generated by interpolation of the time-varying spectral envelope. Upward and downward frequency shifts in spectral envelope (scale factors of 0.75, 1.0, or 1.33) were combined with shifts in $F0$ (scale factors of 0.5, 1.0, or 1.25). Downward frequency shifts generally resulted in malelike voices whereas upward shifts were perceived as childlike. Matched frequency shifts, in which $F0$ and spectral envelope (i.e., formant frequencies) were shifted in the same direction, had relatively little effect on phoneme boundaries. Mismatched frequency shifts, in which $F0$ was modified independently of spectral envelope or *vice versa*, resulted in systematic boundary shifts. The changes in the identifications functions were qualitatively consistent with predictions of a model trained using acoustic measurements derived from a database of naturally spoken vowel tokens from men, women, and children. The empirical and modeling results are consistent with the idea that vowel boundary shifts are a consequence of listeners sensitivity to the statistical structure of natural speech.

2aSC7. Context dependencies in vowel identification in ablated CVC syllables. Jim Talley (Dept. of Linguist., Univ. of Texas, Austin, TX 78712, jtalley@acm.org)

In previously reported work [Talley, J. Acoust. Soc. Am. **108**, 2601 (2000)], novel results from a new perceptual study of human vowel identification under ablation conditions were discussed. That study, which used ten American English (AE) vowels in each of four simple CVC consonantal contexts, found highly significant effects of ablation condition and consonantal context on vowel identifiability. However, little insight was available at the time regarding the specifics of the vowel–context interactions. This paper extends that work providing detailed analysis of vowel identification sensitivity relative to consonantal context under differing ablation conditions.

2aSC8. Lexical duration effects in Japanese function particles. Setsuko Shirai (Dept. of Linguist., Univ. of Washington, Box 354340, Seattle, WA 98195-4340)

It is well known that the lexical status, i.e., content versus function, influences vowel quality and duration in English. However, it is not clear if function particles in Japanese are reduced in a way that is equivalent to English. Previous studies [for example, N. Campbell, in *Speech Production and Linguistic Structure,* 1992] point out that Japanese particles tend to be long because of phrase-final lengthening effects and that function particles and content words tend to have different segmental content. Therefore, any study of the reduction in function particles must control for word and phrase position and must compare segmentally matched tokens. In this study function particles containing three vowels, [e, a, o], are compared with equivalent syllables in content words controlling for segmental context, the number of syllables, and for word and phrase position. For example, a content word [kiga] is matched with a token containing a function particle [ki-ga]. For 11 speakers of Tokyo Japanese, there was a significant effect for lexical category [$F(1, 426)$, $p < 0.01$]. When split for vowel, [a] maintained a significant effect, while the other two vowels only maintained a trend towards reduced duration in function particles.

2aSC9. Speech perception based on spectral peaks versus spectral shape. James M. Hillenbrand (Speech Pathol. and Audiol., Western Michigan Univ., Kalamazoo, MI 49008) and Robert A. Houde (RIT Res. Corp., Rochester, NY 14623)

Some spectral details are more intimately associated with the transmission of phonetic information than others, and a good deal of phonetic perception research has involved drawing inferences about the nature of the spectral representations that mediate phonetic recognition by conducting listening experiments using speech signals which are contrived in such a way as to retain only some characteristics of the speech signal, while purposely removing or distorting other spectral details. The present study was designed to address one aspect of this problem having to do with the relative contributions of spectral envelope peaks versus the detailed shape of the spectral envelope. The problem was addressed by asking listeners to identify nonsense syllables that were generated by two structurally identical source-filter synthesizers, one of which constructs the filter function based on the detailed spectral envelope shape, while the other uses a coarse estimate that is constructed entirely from the distribution of peaks in the envelope. Results suggest that nearly as much phonetic information is conveyed by the relatively coarse peaks-only representation method as by the method that preserves the fine details of the original envelope. However, there is a modest but reliable increase in the transmission of phonetic information when the detailed envelope shape is preserved.

2aSC10. Temporal characteristics of Chinese-accented English: Preliminary findings. Bruce L. Smith (Dept. of Commun. Disord., Univ. of Utah, 390 S. 1530 E., Rm. 1201 Behavioral Sci. Bldg., Salt Lake City, UT 84112-0252), Ann R. Bradlow, and Tessa Bent (Northwestern Univ., Evanston, IL 60208)

This study investigated several temporal features of English to determine the extent of their occurrence in the speech of talkers of Chinese-accented English who had relatively limited experience with spoken English. Specifically, the extent to which these speakers produced the following temporal contrasts was examined: (1) tense versus lax vowel duration, (2) vowel duration before voiced versus voiceless consonants, and (3) vowel and consonant duration in sentence-final versus nonfinal position. Preliminary data from sentences produced by eight non-native and eight native talkers indicates that the native English speakers and the Chinese-accented talkers did not differ in the extent to which they realized the inherent duration difference between tense and lax vowels. However, the native English subjects tended to show substantially greater vowel lengthening before voiced versus before voiceless consonants than the Chinese-accented talkers. In addition, while the two groups did not differ significantly in the extent to which they lengthened sentence-final consonants relative to nonfinal consonants, the native English talkers showed greater sentence-final vowel lengthening than the Chinese-accented talkers. When group differences for a given temporal parameter were found, 1 or 2 of the non-native subjects typically fell within the range of performance shown by the native speakers.

2aSC11. Duration and rate effects on American English vowel identification by native Danish listeners. Terry L. Gottfried (Dept. of Psych., Lawrence Univ., Appleton, WI 54912) and Ocke-Schwen Bohn (Aarhus Univ., DK-8000 Aarhus C, Denmark)

Native listeners alter their identification of American English vowel contrasts according to speaking rate, apparently making judgments about the relative duration of vowels. Berman *et al.* [J. Acoust. Soc. Am. **105**, 1402 (1999)] created a series of spectral continua (varying $F1$ and $F2$) from "beat" to "bit," "pat" to "pet," and "cot" to "cut," also varying syllable duration according to natural speech endpoints. These syllables were inserted into natural speech sentence contexts of two rates (normal and fast). Berman *et al.* found that longer syllable duration led to more long vowel ("beat," "pat," "cot") responses; faster rate contexts also led to more long vowel responses. In the present research we tested native speakers of Danish on their use of duration and rate context in identifying these English vowels. Danish listeners were significantly affected by the duration of syllables in their vowel identification. However, despite vowel duration being phonemic in Danish, native Danish listeners were not significantly affected by rate context in their identification of these English vowels. This might be explained by perceived sufficiency of spectral information in these English vowel contrasts for Danish listeners, or by the lack of rate dependent vowel processing in their native language. [Work supported by Danish-American Fulbright Commission.]

2aSC12. Effects of perceptual assmilation on the production of English vowels by native Japanese speakers. Takeshi Nozawa (Kansai Univ. of Intl. Studies, 1-18 Aoyama Shjimi-cho Miki, Hyogo 673-0521, Japan) and Elaina M. Frieda (Auburn Univ., Auburn, AL 36849)

In our previous studies, it was found that English vowel contrasts that were difficult for Japanese speakers to discriminate were for the most part identified with the same Japanese vowels. This finding complies with one the Perceptual Assimilation Model (Best, 1995) postulates. The present study investigated how perceptual assimilation affects production of English vowels by Japanese speakers. Three experienced Japanese learners in Columbus, Ohio, and five inexperienced Japanese learners in Kobe, Japan produced English vowels, repeating after two different native speakers' utterances. Each subject heard the same token twice in a different order, so they each produced four tokens of each English vowel. The results support what our previous studies have found, and show the effects of perception

on production. The production of the vowels difficult to discriminate tended to be close to each other within subjects' production vowel space. The production of the vowels identified as good exemplars of Japanese vowels were more stable than those identified as poor exemplars. Thus, the subjects /i/ tokens were almost always identified as /i/, whereas their /æ/ tokens were often identified as /ɛ/, /ɑ/, or /ʌ/. Experienced learners' utterances were far more correctly identified.

2aSC13. On the relationship between perception and production of American English vowels by native speakers of Japanese: A pair of case studies. Kanae Nishi (Speech & Hearing Sci., City Univ. of New York–Grad. Ctr., 365 Fifth Ave., New York, NY 10016) and Catherine L. Rogers (Univ. of South Florida, Tampa, FL 33620)

In a previous study, 15 American English (AE) vowels in /hVd/ words recorded by native speakers of Japanese (J) were presented in pairs to native speakers of AE. From these data, native-perceived vowel spaces of J-accented AE were obtained (Nishi, 2001). In the present study, two male native speakers of Japanese who had served as speakers in the previous study listened to the 15 AE vowels produced by two male native speakers of AE in a similarity rating task. Their perceptual data were analyzed using multidimensional scaling. J-perceived vowel spaces of AE vowels produced by AE speakers were compared to AE-perceived vowel spaces of J-produced AE vowels. Vowels produced by both AE and J speakers were also subjected to acoustic analysis. The acoustic vowel spaces obtained were then compared to the AE and J perceptual vowel spaces. Results revealed considerable differences between the two J speakers in terms of their perception of AE vowels. These differences were found to be strongly correlated with the speakers' vowel spaces as perceived by AE listeners, as well as with their acoustic vowel spaces.

2aSC14. Assimilation and discrimination of Canadian French vowels by English-speaking adults. Linda Polka (School of Commun. Sci. and Disord., McGill Univ., 1266 Pine Ave. W., Montreal, QC H3G 1A8, Canada, linda.polka@mcgill.ca), Paola Escudero (Utrecht Univ., The Netherlands), and Shelly Matchett (McGill Univ., Montreal, QC H3G 1A8, Canada)

According to the Perceptual Assimilation Model (PAM) discriminability of non-native contrasts depends on the perceived similarity to native phonetic categories [Best (1994)]. With respect to vowel contrasts, it has been claimed that PAM predictions hold so long as context-specific realizations of a given contrast are considered [Strange *et al.* (2001); Levy (2002)]. To further assess these claims, English adults were tested on perception of Canadian French tense high vowels, /i/, /y/, /u/ and their respective context-conditioned variants [I], [Y], and [U]. Adults completed 2 tasks with both citation and sentence context stimuli; stimuli for both tasks were natural /bVs/ tokens, produced by male and female talkers. For each vowel, subjects completed an identification and rating task (using English vowel response categories) which provided data to assess the assimilation pattern for four non-native contrasts: /i-y/, /y-u/, /I-Y/, and /Y-U/. Discrimination was assessed for each contrast using a categorical AXB task, in which each token is produced by a different talker. The ability to predict discrimination differences from assimilation data was assessed. The findings were examined together with acoustic analysis to determine whether acoustic differences and assimilation differences are equally predictive of relative discriminability.

2aSC15. The identification and discrimination of English vowels produced by native Mandarin speakers. Yang Chen (Univ. of Wyoming P.O. Box 3311, Laramie, WY 82072) and Michael Robb (Univ. of Connecticut, U85, Storrs, CT 06269)

Individuals who speak an Asian language as their first language (L1) are reported to show phonetic inaccuracies in their production of English spoken as a second language (L2) [Flege, 1989]. Phonetic inaccuracies are

assumed to arise from the phonetic differences between the two languages, whereby L1 competes with L2 production [Flege *et al.*, 1997]. A small database is currently available in regarding the influence of Mandarin on the production of American English vowels. The present investigation is to perform a detailed acoustic and perceptual evaluation of American English tense–lax vowel pairs produced by 40 native Mandarin speakers. The Mandarin language does not contain lax vowels, therefore, examining the production of American English tense–lax pairs will provide a critical assessment of a native Mandarin-speaking individual's ability to differentially produce English vowels. The first hypothesis to be tested will be that Mandarin speakers will demonstrate indistinctive acoustic tense–lax vowel differentiation in their English productions. The second hypothesis to be tested will be that Mandarin speakers will be judged perceptually as less successful in differentially producing the adjacent tense–lax English vowel pairs as compared to the American speakers. The phonetic influences of L1 (Mandarin) on L2 (English) will be discussed.

2aSC16. Improving English vowel perception and production by Spanish-speaking adults. Karen Stenning and Donald G. Jamieson (Natl. Ctr. for Audiol., Univ. of Western Ontario, London, ON N6G 1H1, Canada)

This study investigated the effects of perceptual and production training on the abilities of adult native speakers of Spanish to identify and produce the English vowels /i, I, e, ɛ, æ/. Testing was performed prior to and following an average of 12 h of perceptual training in a category inclusion task, then again following an average of 7.5 h of production training involving visual feedback of vowel formant ($F1$ and $F2$) values. A lagged control group of participants who were delayed in starting training showed no improvement in perception and production skills during the control period but changed equivalently during training. The mean improvement in perceptual identification accuracy for /i, I, e, ɛ/ was 17%. The mean improvement in the intelligibility of participants productions of /I, ɛ, æ/ following training was 11%. For both perception and production, most improvement occurred during perceptual training.

2aSC17. Production and perception of vowels in Karitiana. Didier Demolin (Free Univ. of Brussels, 50 av. F. D. Roosevelt, 1050 Brussels, Belgium) and Luciana Storto (Universitade de Sao Paolo, S.P., Brazil)

The main acoustic features of vowels in Karitiana, a language of the Tupi stock spoken in Brazil are examined. This language has 5 vowel qualities [i,e,a,o,ł] which can be oral (short and long) and nasal (short and long). This vowel system has no high back vowel [u]. The main characteristics are that short oral vowels have a wider distribution than long oral vowels; nasal vowels are centralized when compared to oral vowels; length measurements show that central vowels are very short; whatever their quality short nasal vowels have similar length; the difference between short and long nasals is less than in the oral dimension. The dynamic characteristics of nasal vowels where there is a characteristic rising movement of $F2$ are also examined. Finally a perception test was carried out to understand how Karitiana vowels are perceived by native speakers. The test used a set of 58 synthetic stimuli which covers the vowel space. The main results show that speakers easily recognize peripheric vowels while central vowels are less salient. Karitiana speakers did not identify any stimuli in the area of the high back vowel.

2aSC18. Effects of hearing status and perturbation with a bite block on vowel production. Jennell C. Vick, Joseph S. Perkell, Harlan Lane, Melanie Matthies, Majid Zandipour, Ellen Stockmann, Frank Guenther, and Mark Tiede (Speech Commun. Group, Res. Lab. of Electron., MIT, Cambridge, MA 02139)

This study explores the effect of hearing status on adaptation to a bite block in vowel productions of normal hearing (NH) adults and adults who use cochlear implants (CI). CI speakers are tested prior to and following

experience with the implant. Different sized bite blocks (BB) are used to create unusual degrees of mandibular opening for vowel productions in an /hVd/ context (had, head, heed, hid, and hod). Four conditions are elicited from each NH and CI speaker: (1) no BB with hearing (CI processor on), (2) no BB with no hearing (NH speakers with masking noise and CI speakers with processor off), (3) BB with no hearing, and (4) BB with hearing. Prior to fitting with the implant, CI speakers are tested without hearing in two conditions: (1) no BB and (2) BB. Spectra of the vowel productions are analyzed for dispersion of tokens in the $F1$–$F2$ plane in the four conditions. Pilot results support the hypothesis that prior to fitting, CI users are less able to adapt to perturbations than NH speakers and that experience with a CI improves adaptation. The current study is exploring this result further with larger groups of subjects. [Research supported by NIH.]

2aSC19. Mothers exaggerated acoustic-phonetic characteristics in infant-directed speech are highly correlated with infant's speech discrimination skills in the first year of life. Hue-Mei Liu, Patricia K. Kuhl, and Feng-Ming Tsao (Ctr. for Mind, Brain and Learning, CMBL 357988, Univ. of Washington, Seattle, WA 98195, liumei@u.washington.edu)

In addition to the well documented suprasegmental features of infant-directed speech, recent studies demonstrate that critical acoustic-phonetic features are exaggerated at the level of segments in infant-directed speech (IDS) [Kuhl *et al.* (1997); Liu *et al.* (2000); Burnham *et al.* (2002)]. The speech directed to infants contains acoustically more extreme vowels when compared to adult-directed speech, resulting in an expanded vowel space. The relationship between the exaggerated characteristics of IDS and infants' speech perception skills has not previously been examined. The present study tested the hypothesis that there is a correlation between characteristics of speech input and speech perception sensitivity for individual infants in the first year of life. Thirty-two Mandarin Chinese mothers' speech samples were recorded in infant-directed and adult-directed speech conditions. The acoustic analysis demonstrated that Mandarin mothers modify their speech when speaking to their infants, amplifying the important acoustic cues. More importantly, the acoustic-phonetic characteristics in a particular mother's infant-directed speech are significantly correlated with her infant's speech perception. This result supports the view that hyperarticulation of the phonetic units in infant-directed speech facilitates infant's language development. [Work supported by NIH grant (HD37954) and Talaris Research Institute.]

2aSC20. Effect of age and context on vowel area. Megan M. Hodge (Dept. of Speech Pathol. & Audiol., Univ. of Alberta, Rm. 2-70 Corbett Hall, Edmonton, AB T6G 2G4, Canada, megan.hodge@ualberta.ca)

Vowel quadrilateral area, based on first and second formant measures of the corner vowels, has been shown to be correlated positively with intelligibility scores in studies of adults with and without dysarthria [Weismer *et al.,* 2001] and children with and without dysarthria [Higgins and Hodge, 2002]. This study compared vowel areas of three groups of typical talkers (3 year-olds, 5-year-olds, and women) using a log Hz scale in two different speaking conditions. The first speaking condition was production of multiple tokens of isolated /hV/ syllables for each corner vowel, and the second was production of single word items containing the four corner vowels taken from a childrens' test of intelligibility [Hodge, 1996]. An interaction of age with phonetic context was found. Vowel areas for the 3-year-olds did not differ between the two conditions and were the largest of the three groups. The 5-year-olds' and women's vowel areas were of similar size and both were smaller in the word condition. The women

showed the greatest effect of phonetic context with a significantly smaller mean vowel area in the word than isolated vowel condition. [Work supported by Glenrose Rehabilitation Hospital and Canadian Languaage and Literacy Research Network.]

2aSC21. Rhythmic patterns in the speech of developmental apraxia of speech and articulation-disordered children. Maria Passadakes, Fredericka Bell-Berti (Dept. of Speech, Commun. Sci., and Theatre, St. John's Univ., Jamaica, NY 11439, bellf@mail.stjohns.edu), Joanne Paoli (St. John's Univ., Jamaica, NY 11439), and Carole Gelfer (William Paterson Univ., Wayne, NJ 07470)

Many disorders, particularly those of neurological origin, involve disturbances in timing control. Research on children's speech is needed to better understand children's acquisition of phonological processes, and how it goes awry. Studies investigating timing aspects of the speech of developmentally apraxic and articulation-disordered children are lacking.

The purpose of this research is to describe both phrase-final lengthening and compensatory shortening patterns of apraxic and articulation-disordered children, as well as normal children, under 5 years old.

2aSC22. Guidelines for the acoustical conditioning of classrooms in educational buildings. Jorge Alberto Mastroizzi, Carmen Montes, Susana Amura, and Maria Amelia Mastroizzi (Universidad Argentina John F. Kennedy, Gabinete de Investigacin y Vinculacin Technolgica, Rivarola 139 37, (1015) Buenos Aires, Argentina)

The purpose of this work is to identify deficiencies in acoustical conditions within classrooms. Taking the classroom as a representative educational space, acoustical requirements will be determined as a function of the grade level of the students being taught. Sources of noise that affect the classrooms will be examined, assessed, and corrective measures proposed.

TUESDAY MORNING, 3 DECEMBER 2002

GRAND CORAL 1, 7:55 A.M. TO 12:00 NOON

Session 2aUW

Underwater Acoustics: Geoclutter and Boundary Characterization I

Charles W. Holland, Cochair
Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804-0030

John R. Preston, Cochair
Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804-0030

Chair's Introduction—7:55

Contributed Papers

8:00

2aUW1. The Boundary Characterization 2001 Experiment. Charles Holland, Kevin LePage, Chris Harrison (SACLANT Undersea Res. Ctr.), Paul Hines, Dale Ellis, John Osler, Dan Hutt (DRDC-A), Roger Gauss, Redwood Nero (Naval Res. Lab.), and John Preston (Appl. Res. Lab., Penn State Univ., State College, PA 16804)

The weakest link in performance prediction for naval systems operating in coastal regions is the environmental data that drive the models. In shallow water downward refracting environments, the seabed properties and morphology often are the controlling environmental factors. In the Boundary 2001 Experiment, seabed, surface, and biologic scattering, seabed reflection, propagation, reverberation, and ambient noise data were collected in order to develop and refine measurement techniques for key environmental model inputs. Both Rapid Environmental Assessment (REA) methods and high-resolution measurement techniques were employed from 0.1–10 kHz. Supporting oceanographic, geologic, and geophysical data were also collected. The experiment was conducted in May 2001 in two littoral regions: the New Jersey shelf and the Scotian Shelf. This paper provides an overview of the experiment objectives, hypotheses and conduct. [Research supported by NATO SACLANT Undersea Research Centre, ONR, and DRDC-A.]

8:15

2aUW2. Measurements of acoustic backscatter at shallow grazing angles at low kHz frequencies. Paul C. Hines, John C. Osler, and Darcy J. MacDougald (Defence R&D Canada Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, paul.hines@drdc-rddc.gc.ca)

The acoustic backscattering strength of the seabed has been demonstrated to be one of the key inputs required in sonar performance prediction models. Yet, direct measurement of acoustic scattering from the seabed at shallow grazing angles presents a considerable challenge in littoral

waters. The DRDC Atlantic (formerly DREA) Wide Band Sonar system, which consists of a parametric transmitter and a superdirective receiver, is ideally suited to make this measurement. The system was used to measure backscatter as a function of grazing angle and azimuth on the Scotian Shelf off the coast of Nova Scotia and on the Strataform site off the coast of New Jersey. Interpretation of the data set is enhanced with swath bathymetry measurements made at one of the experimental sites. In this paper the experimental geometry is described and the backscatter measurements are presented and discussed in light of the swath bathymetry results.

8:30

2aUW3. Measurements of signal spread and coherence on the New Jersey Shelf and in the Straits of Sicily using time-forward and time-reversed signals. Roger C. Gauss and Richard Menis (Naval Res. Lab., Code 7144, Washington, DC 20375-5350, roger.gauss@nrl.navy.mil)

Mid-frequency shallow-water propagation measurements were made at a variety sites on the New Jersey Shelf and in the Straits of Sicily during three joint trials with the SACLANTCEN (Boundary Characterization: 2000–2002) in order to extract measures of signal spread and coherence, and to evaluate the spatial robustness of time-reversal techniques. The experiments had the NRV Alliance periodically transmitting a 1-s, 200-Hz-bandwidth LFM while it traversed an arc about a stationary platform, which captured these signals and transmitted back time-forward and a set of time-reversed versions of them, as well as transmitting its own version of the original signal. Analysis of normalized matched-filter data indicate that while the two-way time spreads were generally modest, marked decorrelation was observed and that time reversal did well in reconstructing the original impulse response of the LFMs. As the various stored time-reversed versions of the original signals corresponded to different source–receiver paths, a comparison of their signal correlations provided a mea-

sure of the spatial robustness of phase conjugation in the different shelf environments. That statistically the stored time-reversed signals often did as well as *in situ* time-reversed signals suggests the potential for bistatic applications of phase-conjugation concepts. [Work supported by ONR.]

8:45

2aUW4. Analysis of monostatic and bistatic reverberation measurements on the Scotian Shelf. Dale D. Ellis (DRDC Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada) and John R. Preston (Appl. Res. Lab., Penn State Univ., State College, PA 16804)

During the Boundary 2001 sea trial, a number of long-range reverberation measurements were made at a site on the Scotian Shelf between Halifax and Sable Island. The water depth was about 80 m over a sandy bottom. The sources were SUS charges dropped from either the NATO research vessel Alliance or the Canadian research vessel CFAV Quest. The receiver was the SACLANTCEN 254-m towed array aboard Alliance. The data were analyzed in two array apertures and frequencies from 80 Hz to 1400 Hz. Model-data comparisons were made using the Generic Sonar Model, with bottom parameters being extracted using both a manual procedure and an automated procedure. Direct measures of the scattering and bottom properties were made by other researchers, including Hines, Holland, and Osler. The bottom topography was fairly smooth near the site, but at long ranges there were numerous scattering features. The polar plots of the data, and model-data differences, are compared with the bathymetric features in the area. [Work supported in part by ONR Code 32, Grant No. N00014-97-1-1034.]

9:00

2aUW5. Shallow-water reverberation highlights and bottom parameter extractions from the STRATAFORM. John R. Preston (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804) and Dale D. Ellis (DRDC Atlantic, Dartmouth, NS B2Y 3Z7, Canada)

Together with SACLANTCEN, the authors recently participated in the Boundary Characterization Experiment to measure shallow-water bottom reverberation in the STRATAFORM off New Jersey. SUS charges were used as monostatic sources. The receivers were horizontal arrays. Data were analyzed in bands from 160–1500 Hz. The STRATAFORM is known to have benign surface morphology but contains many buried river channels. Highlights of the reverberant returns are discussed that include returns from over the shelf break. Some comparisons in reverberation characteristics between SUS and coherent pulses are noted. Another objective of these reverberation experiments was to quickly invert for bottom scattering and bottom loss parameters. An automated geo-acoustic parameter extraction method was used together with the Generic Sonar Model and a Jackson-Mourad model for scattering. After automatically adjusting bottom loss and scattering strength, good agreement is achieved between the diffuse reverberation data and model predictions in relatively flat areas. Model/data differences are generally correlated with bottom scattering features. Since reverberation typically lasts 10–20 s or more, extracted parameters apply over wide areas. Local bottom loss and backscattering measurements were made by Holland in these areas. A comparison with Holland's results is given. [Work supported by ONR Code 32, Grant No. N00014-97-1-1034.]

9:15

2aUW6. Scattering strength uncertainty. Chris H. Harrison (SACLANT Undersea Res. Ctr., Viale San Bartolomeo, 400, 19138 La Spezia, Italy, harrison@saclantc.nato.int)

A serious weakness in modeling shallow water reverberation is the uncertainty in bottom scattering strength and its angle-dependence. If the bottom scattering law is assumed to be a separable function of an incoming and outgoing angle it follows that the reverberation contains separable incoming and outgoing propagation terms. Thus the returning multipaths from a scattering patch are weighted directly by (the outgoing part of) the scattering law. This means that comparisons of reverberation and propagation angle-dependence on a vertical receiving array have the potential to

reveal the scattering law directly. In this paper we discuss a reverberation experiment with complementary propagation measurements using a VLA and a broadband source to deduce scattering law angle-dependence and absolute scattering strength. The approach is justified by some analysis, and findings are compared with the numerical results of a new multistatic sonar model, SUPREMO. The experiment was conducted in a fairly flat bottomed part of the Mediterranean south of Sicily during BOUNDARY2002.

9:30

2aUW7. Measurements of mid-frequency boundary scattering on the New Jersey Shelf and in the Straits of Sicily. Edward L. Kunz and Roger C. Gauss (Naval Res. Lab, Code 7140, Washington, DC 20375-5350, edward.kunz@nrl.navy.mil)

During joint trials with the SACLANTCEN, direct-path monostatic scattering measurements were conducted at 18 sites across the New Jersey Shelf (Boundary Characterization 2001) and at 9 sites in the Straits of Sicily (Boundary Characterization 2002). Using combinations of short duration cw and LFM signals, both the mean (scattering strength) and statistical (probability density function) characteristics of the bottom- and surface-interaction zones were measured at each site. Bottom-zone scattering strength results show that many sandy and rocky sites exhibited a generally flat dependence on grazing angle (over 20–70 deg) and a moderate dependence (~ 5 dB) on frequency over 2.5–5 kHz. In contrast, the measured surface-zone scattering strengths exhibited a strong dependence on grazing angle (and a mild dependence on frequency), consistent with scattering from the rough air–sea interface. Using physics-based scattering models, coupled with the supporting environmental measurement results, estimates of both the relative contributions of different bottom-zone scattering mechanisms (water-sediment interface, sediment volume and, at low grazing angles, near-bottom fish) and geophysical quantities (such as bottom roughness spectral parameters) were derived and will be presented along with the acoustic data results. [Work supported by ONR.]

9:45

2aUW8. Geoacoustic characterization of seabed scattering experiment locations. John Osler and Blair Lock (Defence R&D Canada Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada)

Measurements of acoustic forward and backscattering have been made by DRDC Atlantic during collaborative sea-trials on the Scotian Shelf off the coast of Nova Scotia and on the Strataform site off the coast of New Jersey. In this paper, the geoacoustic properties and roughness parameters that are necessary to interpret and model the scattering measurements are presented. They have been determined using complementary *in situ* and acoustic techniques. The *in situ* measurements have been made using grab samples and a free fall cone penetrometer that has been fitted with a resistivity module. The probe provides two independent means of calculating the undrained shear strength, an empirical sediment classification and sediment bulk density. The acoustic measurements include inversions for geoacoustic parameters using the WARBLE [Holland and Osler, J. Acoust. Soc. Am. (2000)] and normal incidence sediment classification [Hines and Heald, Proc. Inst. Acoust. (2001)] techniques. At the scattering experimental locations, these measurements have been combined with surveys using commercial equipment: sidescan sonar, multibeam bathymetry, and subbottom profilers to characterize the seabed.

10:00–10:15   Break

10:15

2aUW9. Bistatic reverberation modeling for range-dependent waveguides. Kevin D. LePage[a] (Naval Res. Lab., Washington, DC 20375-5350, kevin.lepage@nrl.navy.mil)

The BiStaR bistatic reverberation model has been developed at SACLANTCEN to model general reverberation scenarios in range-dependent waveguides. Features of the model include coherent propagation to and from the scattering patch, a coherent scattering patch model

which includes anisotropy in the scatterer correlation length scales, and explicit inclusion of system parameters such as bandwidth and source and receiver directivity. The model is built upon the C-SNAP coupled mode model developed at SACLANTCEN by Ferla and Porter. In this talk the theoretical features of the model are presented and the characteristics of coherent and range-dependent bistatic reverberation predicted with the model are discussed. The model is also exercised to help interpret reverberation phenomena observed during the Geoclutter/Boundary Characterization Cruise conducted off the coast of New Jersey during the spring of 2001, including reverberation from beyond the shelf break and scattering from features buried beneath the sediment. [a] Previously at the SACLANT Undersea Research Centre, La Spezia, Italy.

10:30

2aUW10. Comparison of *in situ* compressional wave speed and attenuation measurements to Biot–Stoll model predictions. Barbara J. Kraft, Larry A. Mayer (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824), Peter G. Simpkin (IKB Technologies Ltd., Bedford, NS B4B 1B4, Canada), and John A. Goff (Univ. of Texas Inst. for Geophys., Austin, TX 78759)

The importance of estimating acoustic wave properties in saturated marine sediments is well known in geophysics and underwater acoustics. As part of the ONR sponsored Geoclutter program, *in situ* acoustic measurements were obtained using *in situ* sound speed and attenuation probe (ISSAP), a device developed and built by the Center for Coastal and Ocean Mapping (CCOM). The location of the Geoclutter field area was the mid–outer continental shelf off New Jersey. Over 30 gigabytes of seawater and surficial sediment data was collected at 99 station locations selected to represent a range of seafloor backscatter types. At each station, the ISSAP device recorded 65 kHz waveform data across five acoustic paths with nominal probe spacing of 20 or 30 cm. The recorded waveforms were processed for compressional wave speed and attenuation. Experimental results are compared to predicted values obtained using the Biot–Stoll theory of acoustic wave propagation. Several methods are examined to estimate the required model parameters. The contribution of loss mechanisms to the effective attenuation is considered. [Research supported by ONR Grant No. N00014-00-1-0821 under the direction of Roy Wilkens and Dawn Lavoie.]

10:45

2aUW11. Modeling of reverberation in the East China Sea. T. W. Yudichak, D. P. Knobles (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029l, yudi@arlut.utexas.edu), P. Cable, Y. Dorfman (BBN Technologies, Mystic, CT 06355-3641), R. Zhang, Z. Peng, F. Li, Z. Li (Chinese Acad. of Sci., Beijing 100080, PROC), P. H. Dahl (Univ. of Washington, Seattle, WA 98105), J. H. Miller, and G. R. Potty (Univ. of Rhode Island, Narragansett, RI 02882)

Reverberant time series recorded in the East China Sea component of the Asian Seas International Acoustics Experiment are analyzed with the aid of a model of acoustic scattering from inhomogeneities in the seabed. Wideband sources deployed by the IOA were used to produce the time series, which were recorded on a thirty-two element VLA also deployed by the IOA. The model employs perturbation theory with the framework of a normal mode approach to evaluate scattering by fluctuations in the sound speed and density in the sediment volume as well as of the roughness of the water–sediment interface. The relative importance of surface and volume scattering at low frequencies in shallow-water environments, such as the East China Sea experimental location, is considered. Also, by incorporating this model in an environmental inversion scheme, information about the distribution of inhomogeneities may be gained. [Work supported by ONR.]

11:00

2aUW12. Effects of bathymetric and oceanographic variations on short range high frequency acoustic propagation in shallow water. Christian de Moustier (Ctr. for Coastal & Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824-3525), Seth Mogk, Melissa Hock, and Gerald D'Spain (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0205)

Detailed bathymetry of shoaling regions at two shallow water sites of the San Clemente Offshore Range in Southern California is combined with oceanographic data, taken with CTD, XBT, and bottom moored ADCPs during the winter and summer conditions, to evaluate the effects of environmental variability on acoustic transmission loss and ambient noise measurements made at 16 discrete frequencies ranging from 3 kHz to 30 kHz over 10 km path lengths. Over such short ranges, bottom slopes and tidal effects have the strongest influence on acoustic propagation. [Work supported by the U.S. Naval Oceanographic Office.]

11:15

2aUW13. A comparison of high frequency acoustic transmission data from a smooth water/sand interface with a composite poroelastic model. Marcia Isakson, Daniel Weigl, Erik Bigelow, and Nicholas Chotiros (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029, misakson@arlut.utexas.edu)

Reflection data taken from a smooth water/sand interface have been successfully modeled using a composite poroelastic model. The model was able to account for a decrease in reflectivity at subcritical angles while maintaining physically realistic input parameters. In this study, transmission data have been measured at a number of grazing angles including subcritical angles through a smooth water/sand interface. Because of the flat interface, there should be no contribution to transmission from Bragg scattering or similar scattering effects. The data will be compared with simulations based on a composite poroelastic model using Biot parameters determined from the reflection data. The fit will be considered for its accuracy in predicting the arrivals of the fast, evanescent, and slow waves. [Work supported by ONR, Ocean Acoustics.]

11:30

2aUW14. Separation of scattering mechanisms for the Asian Sea International Acoustics Experiment East China Sea reverberation measurements. Y. Dorfman, P. G. Cable (BBN Technologies), D. P. Knobles, T. W. Yudichak (Appl. Res. Labs., Univ. of Texas, Austin, TX), R. Zhang, Z. Peng, F. Li, Z. Li (Inst. of Ocean Acoust., Beijing, PROC), G. H. Miller, and G. R. Potty (Univ. of Rhode Island, Narragansett, RI)

Low-frequency monostatic reverberation data collected in the East China Sea on a VLA are analyzed to infer bottom scattering strength characteristics. The VLA was deployed and the data collected by researchers from the Institute of Ocean Acoustics, Beijing, China. Reverberation data originating from ranges from 3–15 km are analyzed using coherent array processing methods to determine scattering strengths as a function of frequency and angle at the scatterer. Subaperture processing is used to separate sea surface and bottom contributions and to gain physical insight into the scattering mechanisms responsible for the observed reverberation level. Modeled transmission loss obtained from analyses of measured forward propagation data is employed within the framework of the subaperture signal processing to enable the extraction of the scattering strength. [Work supported by ONR.]

11:45

2aUW15. Sensitivity of broadband propagation and reverberation to oceanographic and bottom variability in shallow water wave guides. Kevin D. LePage (Naval Res. Lab., Washington, DC 20375-5350, kevin.lepage@nrl.navy.mil)

In a previous paper the author derived closed form expressions for the average intensity of broadband time series averaged over oceanographic variability in range independent wave guides [LePage, J. Comput. Acoust. **4** (2001)]. Here, equivalent expressions are derived for the expected value of reverberation intensity. Examples are computed which show that for time series, oceanographic variability most strongly affects the earliest arrivals, while bottom variability most strongly affects the late arrivals. For reverberation, the relative sensitivity to bottom and oceanographic variability were explored using the new model. [Work supported under the ONR capturing uncertainty DRI.]

TUESDAY AFTERNOON, 3 DECEMBER 2002　　　　GRAND CORAL 2, 1:30 TO 3:05 P.M.

Session 2pAAa

Architectural Acoustics and Noise: Music Buildings in Latin America

J. Christopher Jaffe, Chair
Jaffe Holden Acoustics, 114A Washington Street, Norwalk, Connecticut 06854

Chair's Introduction—1:30

Invited Papers

1:35

2pAAa1. The acoustics for speech of eight auditoriums in the city of Sao Paulo. Sylvio R. Bistafa (Dept. of Mech. Eng., Polytechnic School, Univ. of Sao Paulo, Sao Paulo, 05508-900, SP, Brazil, sbistafa@usp.br)

Eight auditoriums with a proscenium type of stage, which usually operate as dramatic theaters in the city of Sao Paulo, were acoustically surveyed in terms of their adequacy to unassisted speech. Reverberation times, early decay times, and speech levels were measured in different positions, together with objective measures of speech intelligibility. The measurements revealed reverberation time values rather uniform throughout the rooms, whereas significant variations were found in the values of the other acoustical measures with position. The early decay time was found to be better correlated with the objective measures of speech intelligibility than the reverberation time. The results from the objective measurements of speech intelligibility revealed that the speech transmission index STI, and its simplified version RaSTI, are strongly correlated with the early-to-late sound ratio C50 (1 kHz). However, it was found that the criterion value of acceptability of the latter is more easily met than the former. The results from these measurements enable to understand how the characteristics of the architectural design determine the acoustical quality for speech. Measurements of ST1-Gade were made as an attempt to validate it as an objective measure of "support" for the actor. The preliminary diagnosing results with ray tracing simulations will also be presented.

2:00

2pAAa2. Three halls for music performance in Chile. Jaime Delannoy, Carolina Heuffemann, Daniel Ramirez, and Fernando Galvez (Universidad Perez Rosales, Brown Norte 290 Nunoa, Santiago, Chile)

The primary purpose of this work was to investigate about the present acoustic conditions of used architectonic spaces in Santiago of Chile for orchestras of classic music performance. The studied halls were three: Aula Magna Universidad de Santiago, Teatro Municipal de Nunoa, and Teatro Baquedano. The used methodology was based on studies made by L. Beranek, M. Barron, among others, in concert halls worldwide. As it guides, for the measurement procedure, physical parameters RT, EDT, C50, C80, LF, BR, G, U50 were evaluated according to norm ISO 3382. On the other hand, it has been defined, to proposal way, a questionnaire of subjective valuation directed to musicians, specialized conductors, and listeners.

2:25

2pAAa3. The first vineyard concert hall in North America. Christopher Jaffe and Carlos Rivera (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854)

The first vineyard or surround concert hall designed and built in the Western Hemisphere is the Sala Nezahualcoyotl in Mexico City. The Hall was completed in 1976 and is part of the Cultural Center at the Universidad Nacional Autonoma de Mexico. The hall was named after a Toltec poet, architect, and musician who lived in the 15th century and was the Renaissance man of his day. In order to provide the familiar traditional sound of the rectangular (shoebox) European Hall, the acoustic designers set the criteria for reverberation times through the frequency spectrum and the Initial Time Delay Gap at every seat in the house to match the measure-

ments taken at the Grosser Musik vereinssaal in Vienna and Boston Symphony Hall. In this paper we discuss the techniques used to create the traditional sound in a vineyard hall and the reaction of musicians and audiences to the completed facility. The Sala was the model for Suntory Hall in Japan which in turn spawned a number of vineyard halls in Japan. Most recently, the vineyard style seems to be appealing to more and more symphonic organizations in Europe and North America.

Contributed Paper

2:50

2pAAa4. The acoustic design of the Centro Nacional de las Artes in Mexico City. Rusell Cooper (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854)

In this paper the acoustic design of the separate buildings housing the school of music, school of drama, and school of dance that opened in 1996 will be described. Spaces that JHA designed included practice rooms, studios, rehearsal rooms, black box, and concert hall. Details of room acoustic treatments, sound isolation measures, and venturi air flow will be illustrated. An overview of the entire project will also include the 500 seat multipurpose theater (with variable absorption systems) and the Alla Magna. Differences between the American and Mexican styles of consulting, importing of materials, installation, and commissioning will also be discussed.

TUESDAY AFTERNOON, 3 DECEMBER 2002 GRAND CORAL 2, 3:15 TO 5:00 P.M.

Session 2pAAb

Architectural Acoustics: Sound Recording and Studios in Mexico

Jose Negrete, Chair

Inst. Politecnico Nacional de Mexico, ESIME-Zacatenco, edif 1, Av. IPN, Mexico 07738, D.F. Mexico

Chair's Introduction—3:15

Invited Paper

3:20

2pAAb1. National Television of Chile—New headquarters building acoustic projects. Mario Huaquin (Natl. Television of Chile, Santiago, Chile, acustmhm@entelchile.net)

In the last 15 years TV stations in Chile have been incorporating in their facilities architectural acoustic and noise control approaches. This has been necessary as much for the technological advance, as for the necessity to achieve a better quality of sound that the listeners receive. In 1998, the National Television of Chile, with the sponsorship of the College of Architects of Chile, requested preliminary architectural designs in order to enlarge and to renovate its headquarters buildings in Santiago, Chile in stages. The Acoustic Project has been developed in an integral way, with three fundamental disciplines: noise and noise control; Machine rooms; vibrations and vibration control; Buildings, engines; architectural acoustics and acoustic comfort; TV studios and technical rooms. This presentation describes the Acoustic Project, phases I (1999), and II (2002), how it was possible to establish a common language with architects and engineers and the different specialties, to apply acoustic criteria and standards, the theoretical development and the projected acoustic solutions. (To be presented in Spanish.)

Contributed Papers

3:45

2pAAb2. Sound perception in the mixing and mastering processes of recorded music. Jose Javier Muedano Meneses (Acoust. Lab., ESIME, IPN, Mexico, jmued@hotmail.com)

This research work presents an analysis of the perceived musical sounds within the mixing process and computer mastering at the audio and video recording studio of the Acoustics Laboratory in ESIME, IPN, Mexico, working with the so-called plug-ins to process the audio signal. The analysis is made from a psychoacoustics standpoint. Samples analyzed include rock, tropical and classical music, and these experiences are part of the Audio course at the Acoustics option of the Electronics and Communications Engineers studies in the Mexican Polytechnic. (To be presented in Spanish.)

4:00

2pAAb3. Acoustic simulations of studio control rooms. Richard A. Moscoso (Seccion Fisica, Pontificia Universidad Catolica del Peru, Apartado 1761, Lima, Peru, rmoscoso@pucp.edu.pe) and Sylvio R. Bistafa (Univ. of Sao Paulo, Sao Paulo, 05508-900, SP, Brazil)

The results of studio-control-room computer simulations with a ray-tracing-type computer program will be presented. Although the validity of ray-tracing simulations of small rooms and at low frequencies may be questionable, the early time echogram at mid and high frequencies obtained by ray tracing provide essential information. Reflections up to 20 ms and frequencies above 500 Hz are responsible for the quality of the listening conditions in studio control rooms, particularly for the assessment of stereophony. Thus, ray-tracing simulations of small rooms for these ranges in the time and frequency domains are justifiable. Computer-

generated impulse responses were obtained for different studio-control-room design philosophies as applied to a basic rectangular room. From a simple scrutiny of the impulse responses the characteristics of the initial time delay region could be verified for each design. Traditional room acoustical parameters such as reverberation time and early decay time were analyzed as well as center time Ts, C5, and C20 for the assessment of the spatial impression, timbre, and transparency. The characteristics of each studio-control-room design philosophy will be discussed in light of these results.

4:15

2pAAb4. Brazilian professional recording studios: Analysis and diagnostics of their acoustical properties. Lineu Passeri, Jr. (Dept. of Technol., Faculty of Architecture and Urbanism, Univ. of Sao Paulo, 05424-970 Sao Paulo, SP, Brazil, nutmusic@uol.com.br) and Sylvio R. Bistafa (Univ. of Sao Paulo, Sao Paulo, SP, Brazil)

Acoustic measurements and design characteristics from a sample of Brazilian professional recording studios located in Sao Paulo and Rio de Janeiro will be presented and discussed. Noise levels, EDT (early decay time), RT20 and RT30 measured in the studios and control rooms will be presented and compared. From these results, the project solutions will be analyzed to check how each recording environment is compatible with its acoustical needs and with the contemporary recording technology. The main goal is to establish guidelines and objective acoustical parameters to be used in the design of new recording studios. (To be presented in Portuguese.)

4:30

2pAAb5. "A" weighting curve algorithm. Jose de Jesus Negrete Redondo, Pablo Roberto Lizana Paulin, Isabel Elena Romero Rizo, and Igmar Moreno Cervantes (Acoust. Lab., ESIME, IPN, Mexico, plizana98@hotmail.com)

The overall project was aimed to build a sound level meter, based on national and international standards. The instrument is being developed on a PC. For this section of the project, the sound signal is captured with a microphone, through an analog/digital converter to the computer. The weighted signal is obtained with a software which includes the equation for the "A" weighting curve to perform a convolution with the fast Fourier transfer of the sound signal. (To be presented in Spanish.)

4:45

2pAAb6. Indoor acoustic gain design. Justo Andrés Concha-Abarca (Universidad Tecnológica Vicente Pérez Rosales, Brown Norte 290, Ñuñoa, Santiago de Chile, Chile, justo.concha@upper.cl)

The design of sound reinforcement systems includes many variables and usually some of these variables are discussed. There are criteria to optimize the performance of the sound reinforcement systems under indoor conditions. The equivalent acoustic distance, the necessary acoustic gain, and the potential acoustic gain are parameters which must be adjusted with respect to the loudspeaker array, electric power and directionality of loudspeakers, the room acoustics conditions, the distance and distribution of the audience, and the type of the original sources. The design and installation of front of the house and monitoring systems have individual criteria. This article is about this criteria and it proposes general considerations for the indoor acoustic gain design.

TUESDAY AFTERNOON, 3 DECEMBER 2002
CORAL KINGDOM 2 AND 3, 1:00 TO 5:30 P.M.

Session 2pAB

Animal Bioacoustics: Amphibian Bioacoustics in Honor of Robert Capranica II

Albert S. Feng, Chair

Beckman Institute, University of Illinois, 405 North Mathews Avenue, Urbana, Illinois 61801

Invited Papers

1:00

2pAB1. Calling and plasticity in Pacific Treefrog choruses. Eliot A. Brenowitz (Depts. of Psychol. & Zool., Univ. of Washington, Box 351525, Seattle, WA 98195, eliotb@u.washington.edu) and Gary J. Rose (Univ. of Utah, Salt Lake City, UT 84112)

Male Pacific treefrogs (*Hyla regilla*) use advertisement and encounter calls to regulate intermale spacing within breeding choruses. Encounter calls are produced when a neighbor's advertisement calls exceed a threshold amplitude. These aggressive thresholds are plastic; males resume advertisement calling (i.e., accommodate) to repeated presentation of advertisement calls at amplitudes 4–8 dB above their aggressive threshold. Correspondingly, a male's aggressive threshold for the advertisement call is positively correlated with the maximum amplitude of neighbors advertisement calls measured at his position. The aggressive thresholds of males for the encounter call are also plastic but not correlated with the maximum call amplitude of their neighbors. This disjunction stems from the fact that males in stable choruses are not regularly exposed to suprathreshold encounter calls. Accommodation to one call type fails to significantly alter a male's aggressive threshold to the other call type, which suggests that the two call types are processed by discrete neural filters. Plasticity of aggressive thresholds and aggressive signalling appears to be important in balancing the costs and benefits of aggressive behaviors. In phonotaxic studies, females show a strong preference for the advertisement call over the encounter call. Prolonged encounter calling would therefore decrease a male's chance of mating.

1:25

2pAB2. Environmental influences on anuran sound communication. Mario Penna (Prog. of Physiol. and Biophys., Univ. of Chile, Casilla 70005, Correo 7, Santiago, Chile, mpenna@machi.med.uchile.cl)

Studies of anuran vocal behavior in the South American temperate forest may represent the southernmost influence of Robert Capranicas comparative approach to sound communication. Vocalizations of leptodactylid frogs in this region exhibit patterns of propagation characteristic for different microhabitats. Calls containing frequencies above about 1 kHz experience higher attenuations in bogs as compared to marshes, irrespective of the species native environment. A similar lack of optimal relationships between signal

structure and habitat properties for call propagation occurs in sound environments other than the temperate austral forest, as indicated by joint studies with colleagues from Cuba in Caribbean tropical forests and from Spain in European Mediterranean habitats. Anurans from southern Chile calling in bogs have peculiar acoustic adaptations: males call from burrows inside which the calls of neighboring cospecifics are amplified considerably. This effect potentially counteracts the constraints for signal propagation of sound-attenuating environments. The evidence presented indicates that anuran calls have not been subjected to environmental selection pressures, to the extent vocalizations of other vertebrates have. Dispersion of anurans is limited by availability of water sources, but males can choose locations favoring signal broadcast and reception in their relatively restricted breeding areas. [Work supported by FONDECYT Grant No. 1010569.]

1:50

2pAB3. Neural correlates of temporal pattern selectivity in anurans. H. Carl Gerhardt (Univ. of Missouri, Columbia, 215 Tucker Hall, Columbia, MO 65211)

Female anurans typically show strong phonotactic selectivity for synthetic signals having pulse-repetition (pulse) rates that are close to those in advertisement calls produced by conspecific males. Auditory neurons showing temporal selectivity for species-typical pulse rates (especially bandpass neurons) are commonly reported from the inferior colliculus and central nucleus of the auditory thalamus. However, a number of questions remain about the neural bases of temporal pattern selectivity. First, the proportion of neurons tuned to conspecific values varies enormously among species. Are these differences real or by-products of different methodologies? Second, behavioral studies indicate that some species do not evaluate the pulse rate *per se* but rather are selective to certain ranges of pulse duration and interpulse intervals. What is the role of bandpass (and other filter types) neurons in these species? Third, temporally selective neurons are often tuned to spectral peaks that are not emphasized in the advertisement call. Do these cells nevertheless play a role in temporal selectivity? Finally, recent studies indicate that extensive lesions of the auditory thalamus have little effect on phonotactic selectivity for conspecific pulse rates. What then is the role of temporally tuned neurons in this structure? [Work supported by NSF and NIH.]

2:15

2pAB4. Temporal call characters and vocal communication in the cricket frog (*Acris crepitans*). Walter Wilczynski (Dept. of Psych., Univ. of Texas, Austin, TX 78712) and Michael J. Ryan (Univ. of Texas, Austin, TX 78712)

During intermale vocal interactions, male cricket frogs change the temporal, frequency, and amplitude characteristics of their calls. In playback studies, males changed the temporal structure of their calls in the direction of higher aggression to a greater degree when presented with calls having a high aggression temporal structure compared to a low aggression structure (with frequency and amplitude held constant). Hearing a low aggression call from the same location prior to a high aggression call resulted in a smaller change in the male's call. Temporal structure degrades progressively between 4 and 16 m from the source over natural substrates. As distance-induced temporal degradation increases, males change temporal call characters progressively less. In sum, male cricket frogs respond to the temporal characteristics of other male calls by changing the temporal parameters of their own calls in a graded fashion. The magnitude of the change depends on the parameters of the challengers call, prior experience with calls from the challengers position, and temporal degradation (hence distance) of the challengers' call. Phonotaxis experiments show that female cricket frogs prefer the high aggression calls. The call changes therefore seem to represent males increasing call attractiveness in response to increased competition from neighboring males.

2:40

2pAB5. Dynamic bimodal signal evokes fighting behavior in a dart-poison frog. Peter M. Narins (Dept. of Physiological Sci., UCLA, Los Angeles, CA 90095-1606), Walter Hoedl, and Daniela S. Grabul (Univ. of Vienna, A 1090, Vienna, Austria)

As a neuroethologist, Bob Capranica strongly encouraged his students to study animals in their natural habitat in order to truly understand the behavioral rules underlying acoustic communication. This study was one of many inspired by those lessons. Male anuran vocalizations serve to attract conspecific females and to regulate male spacing through territorial interactions. In response to playback of a conspecific call, some male frogs have been reported to alternate their call with the perceived stimulus or shift their call-dominant frequency to avoid acoustic interference, or add call notes to signal an increased state of aggression. In some species, males orient toward the sound source followed by physically approaching the loudspeaker. Although natural fighting behavior between conspecific males has been observed in the field, it has not heretofore been possible to elicit with loudspeakers broadcasting sounds. In this playback study of the Brilliant-thighed dart-poison frog (*Epipedobates femoralis*) in French Guiana, we used an electromechanical model to provide realistic bimodal cues (acoustic and visual) to calling males. Our data suggest that agonistic behavior could be evoked in territorial males only when both acoustic and visual cues were presented simultaneously. [Work supported by grants from NIH (PMN) and Austrian Science Foundation (WH).]

3:05–3:30 Break

3:30

2pAB6. The role of call frequency modulation and the auditory papillae in phonotactic behavior in a dart-poison frog. Walter Hoedl (Inst. of Zoology, Univ. of Vienna, Althanstrasse 14, A-1090 Wien, Austria, walter.hoedl@univie.ac.at), Adolfo Amezquita (Universidade de Los Andes, Bogota, Colombia), and Peter M. Narins (Univ. of California, Los Angeles, Los Angeles, CA 90095)

Territorial males of the Brilliant-thighed dart-poison frog, *Epipedobates femoralis*, are known to present stereotypic phonotactic responses to the playback of conspecific and synthetic calls. Fixed site attachment and a long calling period within an environment of little microclimatic changes render this terrestrial and diurnal pan-Amazonian species a rewarding subject for frog bioacoustics. In field experiments at Aratai, French Guiana, we tested whether the prominent frequency modulation of the advertisement call notes is critical for eliciting phonotactic response. Substitution of the natural upward sweep by either a pure tone within the species frequency range or a reverse sweep did not alter the males' phonotactic behavior. Playbacks with advertisement calls embedded in high levels of either low-pass or high-pass masking noise designed to saturate either the amphibian (AP) or basilar papilla (BP) showed that male phonotactic behavior in this species is subserved by activation of the BP rather than the AP of the inner ear. [Work supported by grants from Austrian Science Foundation P 11565, P 15345 (WH), and NIH (PMN).]

3:45

2pAB7. Auditory grouping in the túngara frog: The roles of complex call components in what and where decisions. Hamilton Farris (Integrative Biol., C0930, Univ. of Texas, Austin, TX 78712), A. Stanley Rand (Smithsonian Tropical Res. Inst., Balboa, Panama), and Michael J. Ryan (Univ. of Texas, Austin, TX 78712)

Like humans, numerous animals across disparate taxa must identify and locate complex acoustic signals imbedded in multiple overlapping signals and ambient noise. A requirement of this task is the ability to group sounds into auditory streams in which sounds are perceived as emanating from the same source. Although comparatively few assays have demonstrated aspects of auditory grouping in nonhuman animals, Capranica and colleagues have revealed several excellent examples through their work with anuran bioacoustics. In this study, we build on their work by presenting evidence for auditory grouping in túngara frogs (*Physalaemus pustulosus*). The complex calls of *P. pustulosus* consist of two discrete components, which are commonly produced in multi-male choruses. By measuring the phonotactic responses of females to spatially segregated components, we show that, in contrast to humans, spatial cues play a limited role in grouping, as grouping occurs over wide angular separations. In addition, the presentation of spatially segregated call components allowed us to measure the behavioral significance of each component in the complex. We show that once grouped the separate call components are weighted differently in recognizing and locating the call, so-called "what" and "where" decisions, respectively.

4:00

2pAB8. Nonparallel coevolution of sender and receiver in the acoustic communication system of treefrogs. Johannes Schul and Sarah L. Bush (Biological Sci., Univ. of Missouri, 207 Tucker Hall, Columbia, MO 65211)

Advertisement calls of closely related species often differ in quantitative features such as the repetition rate of signal units. These differences are important in species recognition. Current models of signal/receiver co-evolution predict two possible patterns in the evolution of the mechanism used by receivers to recognize the call. (1) Classical sexual selection models (Fisher-Process, good-genes/indirect benefits, direct benefits models) predict that close relatives use qualitatively similar signal recognition mechanisms tuned to different values of a call parameter. (2) Receiver bias models (hidden preference, pre-existing bias models) predict that if different signal recognition mechanisms are used by sibling species, evidence of

an ancestral mechanism will persist in the derived species, and evidence of a pre-existing bias will be detectable in the ancestral species. We describe qualitatively different call recognition mechanisms in sibling species of treefrogs. Whereas Hyla chrysoscelis uses pulse rate to recognize male calls, H. versicolor uses absolute measurements of pulse duration and interval duration. We found no evidence of either hidden preferences or pre-existing biases. The results are compared with similar data from katydids (Tettigonia sp.). The data are discussed with regard to current models of signal/receiver co-evolution.

4:15

2pAB9. Forty years of solitude: Using laboratory lines of *D. melanogaster* to study behavioral isolation. Christine R. Boake (Dept. of Ecol. & Evol. Biol., Univ. of Tennessee, Knoxville, TN 37996, cboake@utk.edu)

Population genetic models of speciation show that the initial population divergence can be very rapid when sexual signals are involved, and that speciation through sexual selection has a high probability compared to speciation through viability selection. The models also suggest that the exact nature of the changes in a signal system can be arbitrary. This raises empirical questions of whether behavioral divergence can be detected sooner than postzygotic isolation, and whether, in a multimodal signal system, certain signals are more likely to diverge than others. The early stages of behavioral isolation are being investigated by using stocks of D. melanogaster that have been in captivity for 40–50 years. Two lines that were part of a 1950s study of DDT resistance have begun to evolve behavioral isolation; however, postzygotic isolation is not detectable. The courtship signals of recently diverged populations can be compared to published reports of behavioral isolation between populations of D. melanogaster and between D. melanogaster and its close relative to learn whether signal divergence always follows the same trajectory.

4:30

2pAB10. How long do females listen? Assessment time for female choice in the gray treefrog, *Hyla versicolor*. Joshua J. Schwartz (Dept. of Biological Sci., Pace Univ., 861 Bedford Rd., Pleasantville, NY 10570)

A satisfactory understanding of the process of mate choice by female frogs requires that we know how sensitive females are to the variation in males' calls under natural conditions and what is the time scale or "window" over which females compare males. In natural choruses, gray treefrog females may sit near calling males for many minutes before approaching a particular individual to mate, while in laboratory-based tests they may approach a speaker following less than 30 s of exposure to broadcast calls. Females prefer long to short calls. In order to estimate "assessment time" of females in nature, calls were broadcast from four pairs of 360-deg speakers surrounded by screen cages at a pond in Missouri. One speaker per pair presented calls of constant duration, while the other speaker shifted between calls longer or shorter than the constant duration call. The period over which this change in call duration occurred differed for each of the four pairs of speakers. The numbers of females captured at the speaker array over the breeding season indicated that the most likely assessment time was close to 2 min. This estimate is similar, but not identical, to that obtained from additional laboratory tests.

4:45

2pAB11. Phonotaxis and chorus organization in African frogs. Phillip J. Bishop, Robert R. Capranica, and Neville I. Passmore (Dept. of Zoology, Univ. of the Witwatersrand, Johannesburg, S. Africa)

Detailed studies of the phonotactic behavior and chorus organization of several species of African amphibians were conducted from 1982–1992. A range of phonotactic experiments conducted in two- and three-dimensional arenas, using between one and four loudspeakers and a variety of different stimuli were carried out on the painted reed frog

2p TUE. PM

(*Hyperolius marmoratus*). These experiments revealed the remarkable ability of very small anurans (<30 mm, SVL) to accurately localize a sound source, in both the horizontal and vertical planes. Furthermore, females were able to discriminate between two sound sources that differed in intensity by only 5 dB and their ability was significantly impaired by the introduction of further sound sources. The confounding effect of multiple sound sources on female choice may explain the presence of nonran-dom mating with respect to size in this species. The chorus organization of five species of African anurans was investigated by using playback experiments. These experiments revealed four distinct categories of chorus organization which can be plotted on a continuum, from random calling to very precise triggered responses. The type of chorus organization was found to be directly related to the length of the call and the delay in response to the playback stimulus.

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5:00–5:30

Questions and Comments

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TUESDAY AFTERNOON, 3 DECEMBER 2002 CORAL GARDEN 1, 1:00 TO 5:45 P.M.

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Session 2pAO

Acoustical Oceanography and Animal Bioacoustics: Using Ambient Sound in the Ocean

Jeffrey A. Nystuen, Chair

Applied Physics Laboratory, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105

Chair's Introduction—1:00

Invited Papers

1:05

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2pAO1. NOAA efforts in monitoring of low-frequency sound in the global ocean. Christopher G. Fox   (NOAA/PMEL, 2115 S.E. OSU Dr., Newport, OR 97365, fox@pmel.noaa.gov), Robert P. Dziak, and Haruyoshi Matsumoto   (Oregon State Univ., Newport, OR)

Since August 1991, NOAA/PMEL has collected continuous recordings from the U.S. Navy SOSUS arrays in the North Pacific. In May 1996, this effort was expanded through the use of PMEL-developed autonomous hydrophones deployed in the eastern equatorial Pacific, and later to the central North Atlantic between 15N and 35N (March 1999), the Gulf of Alaska (October 1999), and the North Atlantic between 40N and 50N (June 2002). Natural seismicity in the Pacific produces nearly 10 000 events per year with source levels exceeding 200 dB (*re*: 1 micro-Pa @ 1 m), with about 3500 events per year exceeding this level in the North Atlantic. Significant contributions from manmade sources are present throughout the data but have not been quantified. Recordings from North Atlantic arrays are dominated by noise from seismic airgun profilers working offshore Canada, Brazil, and west Africa. In September 2001, a cabled vertical hydrophone array was installed at Pioneer Seamount, offshore central California, which will provide continuous, unclassified acoustic data (in the range of 1–450 Hz) to the research community in real time. Future plans call for the expansion of the NOAA monitoring effort to other opportunities worldwide and making the raw data available to the community via the Internet.

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1:25

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2pAO2. Eight-year records of low-frequency ambient sound in the North Pacific. Rex K. Andrew, Charlotte Leigh, Bruce M. Howe, and James A. Mercer   (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

Spectra of omnidirectional ambient sound have been collected since 1994 at 13 locations around the North Pacific. Data were acquired for 3 minutes every 6 minutes and spectra calculated from 0–500 Hz in 1 Hz bands. With a million spectra per site, this database allows investigation into the statistical character of low-frequency ambient sound at multiple scales. At the shortest scales, the spectral levels in the shipping bands have a fluctuation spectrum similar to a $1/f$ process, with decorrelation times less than 20 minutes. At intermediate scales, the seasonal baleen whale component becomes the most dominant and repeatable feature. At the longest scales (averaging over the entire record) the ambient levels (at the Pt. Sur site) seem to have increased by up to 10 dB since the 1960s. The distribution of the levels (in decibels) generally indicates a short tail for quieter levels but a long tail for loud events. The Pt. Sur data set has also been used to validate the new dynamic ambient noise model (DANM), which shows good agreement in one-third octave bands to within a couple of decibels for January 1998. These and further results will be discussed. [Work supported by ONR and SPAWAR.]

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1:45

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2pAO3. Creating a web-based library of underwater biological sound. Jack W. Bradbury and Carol A. Bloomgarden   (Macaulay Library, Cornell Lab. of Ornithology, 159 Sapsucker Woods Rd., Ithaca, NY 14850)

Establishing an archive of fish and other underwater biological sounds will meet many of the long-standing challenges faced by marine acousticians—the restoration and preservation of deteriorating recordings, the ability to catalog their sounds and data in a way that fosters the exchange and sharing of data, easy access to the sounds for analysis and identification, and the capacity to search through passive recordings for sounds of particular interest. The Macaulay Library of Natural Sounds, with a long history of working toward these goals in ornithology and animal behavior, recently launched into the realm of underwater sounds with the help of over 80 individual recordists and institutions worldwide. Researchers will be able to annotate their sounds through an online database

2260   J. Acoust. Soc. Am., Vol. 112, No. 5, Pt. 2, November 2002 Pan-American/Iberian Meeting on Acoustics   **2260**

application, summarize search results in exportable tables and maps, and download copies of recordings for research, teaching, and conservation. MLNS is committed to dual goals of maintaining open access to allow other researchers to listen and help identify sounds, while protecting recordists copyrights and restricting access during the publication process. Detailed and extensive metadata are needed, however, to create the functionality such an archive requires.

2:05

2pAO4. Eavesdropping on marine mammals to monitor migration. Kathleen Stafford (Natl. Marine Mammal Lab., 7600 Sand Point Way NE, Seattle, WA 98115)

Many baleen whales produce low-frequency sounds that are sufficiently loud that they can be detected over long distances. Collectively these sounds contribute significantly to ambient noise levels in the ocean during certain times of the year [Curtis *et al.*, J. Acoust. Soc. Am. **106**, 3189–3200 (2000)]. By monitoring the occurrence of these sounds at different locations in the ocean, geographic and seasonal patterns begin to emerge that may be used to document the distribution and basin-scale movements of baleen whales. Each baleen whale species make distinctly different calls so distinguishing among them, and sometimes among populations within species, is possible. Although for many species the ecological function of sound production remains poorly understood, acoustic data can nevertheless be used to examine the seasonal occurrence and migratory behavior of large whales. One example of the utility of monitoring migrations by use of passive acoustics can be seen in the northeastern Pacific where sounds from blue whales are detected in the eastern tropical Pacific from November to May, off the west coast of the United States from July to January and in the Gulf of Alaska from September to December.

2:25

2pAO5. Toward a fisheries bioacoustics. David A. Mann (Univ. of South Florida, College of Marine Sci., 140 7th Ave. S., St. Petersburg, FL 33701, dmann@marine.usf.edu)

Fisheries bioacoustics is emerging as an application of passive acoustic detection of fish sound production. Many fishes produce sounds during reproductive activities that can be used to determine where and when they are spawning. Autonomous dataloggers and hard-wired hydrophones were used to record sound production by fishes in estuaries of southwest Florida. These data show that passive acoustics can be used to locate spawning sites and determine the timing of spawning by commercially important species. Ultimately fisheries bioacoustics should move the way of fisheries acoustics where the signal output is not the actual sound data, but the locations and intensity of fish spawning. A useful analogy is the development of SONAR systems for fish quantification. These systems do not deliver raw sound data to the researcher. They return processed data on fish location and abundance. One can envision the day when real-time fisheries bioacoustics systems will produce maps of the locations of sound-producing fishes that can provide managers with data on the temporal and spatial extent of spawning.

2:45

2pAO6. Expanding uses of ambient noise for imaging, detection, and communication. John R. Potter and Laurent Malod (ARL, TMSI, NUS, 12a Kent Ridge Rd., Singapore 119223)

The use of ambient noise to sense the marine environment has a human history of only two decades. Starting with incoherent processing inspired by optical analogies such as Acoustic Daylight (TM), the exploration of the potential for ambient noise to provide useful information about the environment has blossomed into many related areas and diverse algorithms with connections to multi-static active sonar, classic passive sonar, communications, matched field processing and others. This presentation will introduce some recent work in these areas and attempt to draw together how the use of ambient noise, both by mankind and marine animals, is beginning to form a more complete picture of the potential of this exciting area of research. [Work supported by the Defence Science and Technology Agency, Singapore.]

3:05

2pAO7. Estimating shallow water bottom geoacoustic parameters from ambient noise. Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Knowing bottom geoacoustic parameters is of great importance for using sonar systems effectively in shallow waters. In this paper, ambient noise data recorded on a vertical hydrophone array taken in the frequency range of 1000 to 3000 Hz were used. Forward modeling and model/data comparison show that the energy ratio of down-looking and up-looking beams, after a proper average over time and frequency, is the energy reflection coefficient of the bottom. From the reflection coefficient, critical parameters of the sediments, the sound speed, density, and attenuation coefficient, are obtained. Core data taken at the experimental site support the inversion results.

3:25–3:40   Break

3:40

2pAO8. Monitoring air entrainment with breaking wave noise. Grant B. Deane and Dale M. Stokes (Code 0238, Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238)

It is now known that the dominant component of wind-driven oceanic noise comes from breaking waves. Bubbles ranging in size from tens of microns to centimeters are forced into the water column during the first second or so of whitecap formation. At the moment of creation, each bubble emits a pulse of sound at a center frequency inversely proportional to its radius. The ensemble of such events amounts to a burst of noise that continues throughout the active phase of bubble creation within the whitecap. As the noise spectrum is related to the bubble size distribution within the whitecap, it is natural to explore the possibility that underwater oceanic

ambient noise could be used as tool to remotely monitor air entrainment rates across the ocean surface. One of the problems with developing such a tool is understanding the relationship between bubble formation processes occurring within whitecaps and the concurrent noise emission. Here we will report recent progress in our understanding of bubble formation mechanisms in whitecaps, their role in ambient noise generation, and the implications for monitoring air entrainment rates.

Contributed Papers

4:00

2pAO9. Passive acoustic detection and measurement of rainfall at sea.
Jeffrey A. Nystuen and Barry Ma (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, nystuen@apl.washington.edu)

It is well recognized that rainfall measurements are needed over the world's oceans. One method of providing these measurements is to passively listen for the underwater sound signal that is produced by rainfall striking the ocean surface. Since 1998, over 70 buoy-months of ambient sound data have been collected using Acoustic Rain Gauges (ARGs) deployed on deep ocean moorings that form part of the Tropical Atmosphere Ocean (TAO) array in the Pacific Ocean. These data demonstrate the acoustic measurement of oceanic wind and rain. Other ''noises'' are present in the ocean and need to be detected and rejected. This is accomplished by recognizing the unique spectral and temporal character of rain-generated sound. A quantitative relationship between absolute sound levels and rainfall rate is proposed. The probability of acoustic detection of rainfall events under different weather conditions will be discussed. A quantitative comparison of rainfall accumulation using the acoustic technique with co-located rainfall estimates from on-board R.M. Young rain gauges and from NASA TRMM satellite overpasses (Rainfall Product 3B42) shows promising agreement, but also points out problems associated with each measurement method. [Work supported by ONR—Ocean Acoustics, NSF—Physical Oceanography, and NOAA Office of Global Programs.]

4:15

2pAO10. Is anthropogenic ambient noise in the ocean increasing?
Elena McCarthy (Dept. of Marine Affairs, Univ. of Rhode Island, Washburn Hall, Kingston, RI 02881, emcc7035@postoffice.uri.edu) and James H. Miller (Univ. of Rhode Island, Kingston, RI 02881)

It is commonly accepted that the ocean's ambient noise levels are rising due to increased human activities in coastal and offshore areas. It has been estimated that low-frequency noise levels increased more than 10 dB in many parts of the world between 1950 and 1975. [Ross, Acoustics Bulletin, Jan/Feb (1993)]. Several other sources cite an increase in man-made, or anthropogenic, noise over the past few decades. [D. A. Croll *et al.*, Animal Conservation **4**(1) (2001); Marine Mammal Commission Report to Congress (1999); C. W. Turl, NOSC Tech. Report 776 (1982)]. However, there are few historical records of ambient noise data to substantiate these claims. This paper examines several sectors of anthropogenic activities to determine their contributions to ambient noise. These activities include shipping, oil and gas exploration, military sonar development, and academic research. A series of indices for each of these industries is developed to predict ambient noise trends in the sea. It is found that the amount of noise generated by individual activities may have decreased overall due to new technologies and improved efficiency even if the intensity of such activities has increased.

4:30

2pAO11. Geoacoustic inversion of noise coherence in shallow water.
David J. Thomson, Francine Desharnais (Defence Res. and Development Canada Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada), Matthew L. Drover (Dalhousie Univ., Halifax, NS B3H 4J1, Canada), and Chris A. Gillard (Defence Sci. and Technol. Organisation, Edinburgh, South Australia 5111, Australia)

It is known that the geoacoustic properties of a shallow-water sea-bed can be inferred from relatively simple measurements of the ambient noise coherence between a pair of vertically separated hydrophones [D. M. F. Chapman, ''Surface-generated noise in shallow water: A model,'' Proc. Inst. Acoust. **9**, 1–11 (1987)]. The design of an autonomous buoy package for acquiring geoacoustic information by this method is currently being considered by DRDC-Atlantic in support of matched-field localization efforts that are being developed for use with rapidly deployable arrays. Initially, vertical coherence estimates from a simple shallow water noise model were fit to measured coherences by adjusting geoacoustic parameters by a trial and error procedure. A more systematic approach involves combining noise coherence models with nonlinear global optimization methods based on matched-coherence processing concepts to search the space of possible sea-bed parameters more efficiently. In this paper, we report on recent efforts to use a hybrid simplex simulated annealing scheme [M. R. Fallat and S. E. Dosso, ''Geoacoustic inversion via local, global, and hybrid algorithms,'' J. Acoust. Soc. Am. **105**, 3219–3230 (1999)] to match an increasingly realistic suite of candidate geoacoustic parametrizations to some acoustic noise coherent data measured with modified sonobuoys deployed at several shallow water locations on the Scotian Shelf.

4:45

2pAO12. Model-based tracking of marine mammals near California using seismometers.
Christopher O. Tiemann, Michael B. Porter (Sci. Appl. Intl. Corp., La Jolla, CA 92037), and John Hildebrand (Univ. of California, San Diego, La Jolla, CA 92037)

An algorithm originally developed for tracking humpback whales around a deep-water hydrophone array near Hawaii has been proven capable of localizing another species of marine mammal in the shallow waters off California. The new sources of interest are blue whales, animals with markedly different call characteristics from humpback whales, and the data under examination is from four bottom-mounted seismometers in a 3 km square array. The algorithm uses a range-dependent acoustic model to predict time differences of arrival (time-lag) of blue whale calls as measured between sensor pairs, while real pairwise time-lags are measured through a phase-only correlation process. Comparison between modeled and measured time-lags forms an ambiguity surface identifying the most probable whale location in a horizontal plane around the sparse array. The robustness of the model-based localization technique is illustrated in its application to a different scenario than for which it was developed, and it is also suitable for continuous, real-time, unattended alert and tracking applications. Examples of tracking whales along their migratory path will be provided.

5:00

2pAO13. Breaking waves and ambient sound in the ocean.
W. Kendall Melville (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0213, kmelville@ucsd.edu)

It is now well accepted that breaking waves at the ocean surface are the primary source of sea-surface sound. However, the relationship between the kinematics and dynamics of breaking waves, their acoustic source strength and their statistical description has not been explored in any comprehensive way. In this paper, the components of such a description are reviewed and discussed in the context of laboratory and field studies from the literature, theoretical models of breaking statistics, and recent field measurements of the kinematics and statistics of wave breaking. The implications of the results for the use of ambient sound to quantify processes of air–sea interaction are discussed. [Work supported by ONR and NSF.]

2pAO14. Hurricane classification using a full-field ocean surface noise model. Joshua D. Wilson and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

Hurricanes generate noise in the ocean due to wind–wave interaction. The authors have previously discussed the possibility of using this noise to determine the size and strength of the hurricane with a modal model [Wilson and Makris, J. Acoust. Soc. Am. **108** (2000)]. Here the analysis is extended to include a full-field model for surface-generated ocean noise. Unlike previous surface noise models that contained far-field approximations, this full-field model can be used to calculate the acoustic field both inside and outside the hurricane. This full-field model is used to calculate the spatial covariance of the acoustic field generated by a hurricane. This spatial covariance is then used to determine the sound from a hurricane that would be detected by hydrophones and hydrophone arrays. Several examples are presented using single sensors and sensor arrays inside and outside the hurricane to determine the best method for classifying a hurricane. In addition shallow- and deep-water environments are compared to illustrate their effect on the propagation of surface-generated hurricane noise. Also simulations are shown for multiple frequencies to show the filtering effect of the waveguide on the propagation and to determine the optimal frequency for hurricane classification.

2pAO15. Laboratory measurements of noise generation by shoaling breakers. Steven L. Means and Paul J. Gendron (Naval Res. Lab., Code 7120, 4555 Overlook Ave. SW, Washington, DC 20375)

Simultaneous measurements of the surface gravity wave field, void fraction entrained during breaking, and the generated acoustic spectra were made in a sand beached wave tank at the Center for Applied Coastal Research at the University of Delaware during April 2002. The tank has the dimensions of 30 m (l)×2 m (w)×1.5 m (d) and a 0.2-mm sand beach. Four hydrophones were distributed along the sand beach beneath the region of active breaking. Wave gauges measured surface gravity wave properties along the beach cross-shore. Conductivity and temperature probes allowed for the measurement of the void fraction of the entrained bubble cloud. A video camera captured the evolution of the entrained bubble cloud through the tank's Plexiglas™ side panel. The paper presents generated acoustic power levels as a function of surface wave amplitude and period. Initial results in obtaining the relationships between void fractions of the entrained bubble clouds and the spectral components of the generated acoustic signal will also be discussed. [Work supported by ONR base funding at NRL.]

2p TUE. PM

TUESDAY AFTERNOON, 3 DECEMBER 2002   CORAL KINGDOM 1, 1:00 TO 1:50 P.M.

Session 2pEA

Engineering Acoustics: Honoring Per Brüel's Contributions

Leo L. Beranek, Chair
975 Memorial Drive, Suite 804, Cambridge, Massachusetts 02138-7555

Chair's Introduction—1:00

Invited Paper

1:05

2pEA1. Working with Dr. Per V. Brüel. Svend Gade (Brüel & Kjær Univ., Brüel & Kjær, Sound & Vib. Measurement A/S, Skodsborgvej 307, DK-2850 Naerum, Denmark)

For more than a decade I have had the pleasure to work as an application specialist together with—and for—Dr. Brüel, one of the founders of the Brüel & Kjær Company, famous for sound and vibration measurement instrumentation, often nicknamed "Green Boxes." It has been a great experience for me, and I recall this period in my life as one where I was much inspired by Dr. Brüel's methods, both as a private person and with his work as a director for the company and leader of both the sales and the innovation departments. In this presentation I will highlight some funny stories that are told about Dr. Brüel combined with the episodes that I have experienced myself. In short, the most simple way to characterize this rather complex person is maybe by repeating his vision statement for the company: "We shall have fun and we shall make money. On the other hand we shall not have so much fun that we do not make any money, and we shall not make so much money that we do not have any fun!" For Per Brüel, acoustics is one of his great hobbies. He has others such as cars, airplanes, motorbikes (he is the lucky owner of a Danish Nimbus) and wine.

Contributed Paper

1:35

2pEA2. Upcoming new international measurement standards in the field of building acoustics. Hans Goydke (Physikalisch-Technische Bundesanstalt (PTB), Bundesallee 100, D-38116 Braunschweig, Germany, hans.goydke@ptb.de)

The extensively completed revision of most of the ISO measurement standards in building acoustics mainly initiated by the European Commissions demand for harmonized standards emphasized the insight that the main goal to avoid trade barriers between the countries can only be reached when the standards sufficiently and comprehensively cover the field when they are related to the actual state of the art and when they are sufficiently related to practice. In modern architecture one can observe the rapid change in the use of building materials, for instance regarding the use of glass. Lightweight constructions as well as heavyweight building elements with additional linings are increasingly in common use and unquestionably there are consequences to be considered regarding the ascertainment of sound insulation properties. Besides others, International Standardization is unsatisfactory regarding the assessment of noise in buildings from waste water installations, in the low frequency area and in general regarding the expression of uncertainty of measurements. Intensity measurements in building acoustics, rainfall noise assessment, estimation of sound insulation, impulse response measurement methods, assessment of sound scattering are examples of upcoming standards.

Session 2pMUa

Musical Acoustics: General Topics in Musical Acoustics

James P. Cottingham, Chair

Physics Department, Coe College, Cedar Rapids, Iowa 52402

Contributed Papers

1:00

2pMUa1. Observation of the laryngeal movements for throat singing. Ken-Ichi Sakakibara (NTT Commun. Sci. Labs., 3-1, Morinosato Wakamiya, Atsugi-shi, Kanagawa 243-0198, Japan), Tomoko Konishi, Emi Z. Murano, Hiroshi Imagawa (The Univ. of Tokyo, Tokyo, Japan), Masanobu Kumada (Natl. Rehabilitation Ctr. for the Disabled, Saitama, Japan), Kazumasa Kondo (Saitama, Japan), and Seiji Niimi (Intl. Univ. of Health and Welfare, Tochigi, Japan)

Throat singing is a traditional singing style of people who live around the Altai Mountains. Khöömei in Tyva and Khöömij in Mongolia are representative styles of throat singing. The laryngeal voices of throat singing is classified into (i) a *drone voice* which is the basic laryngeal voice in throat singing and used as drone and (ii) a *kargyraa voice* which is very low pitched with the range outside the modal register. In throat singing, the special features of the laryngeal movements are observed by using simultaneous recording of high-speed digital images, EGG, and sound wave forms. In the drone voice, the ventricular folds (VTFs) vibrate in the same frequency as the vocal folds (VFs) but in opposite phases. In the kargyraa voice, the VTFs can be assumed to close once for every two periods of closure of the VFs, and this closing blocks airflow and contributes to the generation of the subharmonic tone of kargyraa. Results show that in throat singing the VTFs vibrate and contribute to producing the laryngeal voice, which generates the special timbre and whistle-like overtone.

1:15

2pMUa2. A human vocal utterance corpus for perceptual and acoustic analysis of speech, singing, and intermediate vocalizations. David Gerhard (Dept. of Computing Sci., Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada)

In this paper we present the collection and annotation process of a corpus of human utterance vocalizations used for speech and song research. The corpus was collected to fill a void in current research tools, since no corpus currently exists which is useful for the classification of intermediate utterances between speech and monophonic singing. Much work has been done in the domain of speech versus music discrimination, and several corpora exist which can be used for this research. A specific example is the work done by Eric Scheirer and Malcom Slaney [IEEE ICASSP, 1997, pp. 1331–1334]. The collection of the corpus is described including questionnaire design and intended and actual response characteristics, as well as the collection and annotation of pre-existing samples. The annotation of the corpus consisted of a survey tool for a subset of the corpus samples, including ratings of the clips based on a speech–song continuum, and questions on the perceptual qualities of speech and song, both generally and corresponding to particular clips in the corpus.

1:30

2pMUa3. Computer-animated illustrations of vibrations and waves. Donald E. Hall (Phys. Dept., California State Univ., 6000 J St., Sacramento, CA 95819, hallde@dgweb.com)

Under this same title, Bruce Richards presented at the 143rd ASA meeting [J. Acoust. Soc. Am. **111**, 2394 (2002)] an admirable set of class demonstrations implemented on a Macintosh computer with C++ and the Mac Toolbox. This has inspired the preparation of a similar package of animations written in BASIC for the PC. They include visualizations of motion and corresponding graphs for plucked and bowed strings, bars and membranes, standing and traveling waves in pipes, and normal modes of cylindrical and conical pipes. The animations correspond closely to illustrations in standard textbooks such as the author's [*Musical Acoustics*, Brooks–Cole, 3rd ed., 2002].

1:45

2pMUa4. Pitch jnd and the tritone paradox: The linguistic nexus. Kourosh Safari (Music Cognition and Acoust. Lab., Univ. of California, Los Angeles, Box 951657, Los Angeles, CA 90095-1657)

Previous research has shown a connection between absolute pitch (the ability to name a specific pitch in the absence of any reference) and native competence in a tone language (Deutsch, 1990). In tone languages, tone is one of the features which determines the lexical meaning of a word. This study investigates the relationship between native competence in a tone language and the just noticeable difference of pitch. Furthermore, the tritone paradox studies have shown that subjects hear two tritones (with bell-shaped spectral envelopes) as either ascending or descending depending on their linguistic backgrounds (Deutsch, 1987). It is hypothesized that the native speakers of tone languages have a higher JND for pitch, and hear the two tones of the tritone paradox as ascending, whereas, native speakers of nontone languages hear them as descending. This study will indicate the importance of early musical training for the development of acute tone sensitivity. It will also underline the importance of language and culture in the way it shapes our musical understanding. The significance of this study will be in the areas of music education and pedagogy.

2:00

2pMUa5. The effects of timbre on melody recognition are mediated by familiarity. J. Devin McAuley and Chris Ayala (Dept. of Psych., Bowling Green State Univ., Bowling Green, OH 43403)

Two experiments examined the role of timbre in music recognition. In both experiments, participants rated the familiarity of a set of novel and well-known musical excerpts during a study phase and then were given a surprise old/new recognition test after a retention interval. The recognition test was comprised of the target melodies and an equal number of distractors; participants were instructed to respond yes to the targets and no to the distractors. In experiment 1, the timbre of the melodies was held constant throughout the study and then either stayed the same or switched to a different instrument sound during the test. In experiment 2, timbre varied randomly from trial to trial between the same two instruments used in experiment 1, yielding target melodies that were either mismatched or matched in their timbre. Switching timbre between study and test in experiment 1 was found to hurt the recognition of the novel melodies, but

not the familiar melodies. The mediating effect of familiarity was eliminated in experiment 2 when timbre varied randomly from trial to trial rather than remaining constant. Possible reasons for the difference between studies will be discussed.

2:15

2pMUa6. Recreating the real, realizing the imaginary—a composer's preoccupation with acoustic space. Rob Godman (Univ. of Coventry and Univ. of Hertfordshire, c/o 4 Mill Close Wotton-under-Edge, Glos GL12 7LP, UK)

For centuries composers have been concerned with spatialization of sound and with the use of acoustic spaces to create feeling, atmosphere, and musical structure. This paper will explore Rob Godman's own use of sound in space, including (1) his treatment of ancient Vitruvian principles and how they are combined with new technologies; (2) an exploration of virtual journeys through real and imaginary acoustic spaces; (3) how sounds might be perceived in air, liquid, and solids; and (4) how technology has allowed composers to realize ideas that previously had only existed in the imagination. While focusing on artistic concerns, the paper will provide information on research carried out by the composer into acoustic spaces that are able to transform in real time with the aid of digital technology (Max/MSP software with sensor technology) and how these have been used in installation and pre-recorded work. It will also explore digital reconstructions of Vitruvian theatres and how we perceive resonance and ambience in the real and virtual world.

2:30

2pMUa7. Music-therapy analyzed through conceptual mapping. Rodolfo Martinez (CIICIR, IPN, Oaxaca, Mexico, Rodolfo_cm@yahoo.com) and Rebeca de la Fuente (IMA, Mexico)

Conceptual maps have been employed lately as a learning tool, as a modern study technique, and as a new way to understand intelligence, which allows for the development of a strong theoretical reference, in order to prove the research hypothesis. This paper presents a music-therapy analysis based on this tool to produce a conceptual mapping network, which ranges from magic through the rigor of the hard sciences.

2:45

2pMUa8. Some recent examples of fractal music. Cesar Guerra-Torres, Moises Hinojosa-Rivera, Juan Angel Garza-Garza, and Fernando J. Elizondo-Garza (Acoust. Lab., FIME, Univ. A. de Nuevo Leon, P.O. Box 28 "F," Cd. Universitaria, San Nicolas, 66450, N.L., Mexico, cguerratorres@hotmail.com)

Since the first published studies of fractal music by Mandelbrot and Voss, the relationship between music, mathematics, and fractal geometry has been a very active field of research. It has been found that the music of classical composers can be characterized by fractal or self-affine parameters, which in turn serve as the basis for synthetic fractal music. In this work is presented a brief discussion of the state of the art as well as some recent examples of fractal music, including a live demonstration.

TUESDAY AFTERNOON, 3 DECEMBER 2002   CORAL SEA 1 AND 2, 3:30 TO 5:10 P.M.

Session 2pMUb

Musical Acoustics: Musical Instruments of the South American Dance Tradition

Paul A. Wheeler, Chair
Utah State University, Logan, Utah 84322

Invited Papers

3:30

2pMUb1. An overview of musical instruments used in South American dance traditions. Paul A. Wheeler (Utah State Univ., Logan, UT 84322)

Musical instruments used in South American dances combine elements from Amerindian, African, and European musical traditions. The Amerindian influence can be seen in Andean instruments, such as the end-blown flute, panpipe, and charango (modified from the European guitar). The berimau, a musical bow used in the Brazilian capoeira dance, is an example of African influence. The bandoneon is a square-ended German concertina most famous for its use in the tango from Argentina. This paper provides an overview of the musical instruments commonly used in South American dance traditions in relationship to their origins. The acoustics of some of these instruments, such as the guitar, has been studied in detail, whereas others, like the Brazilian cuica, provide opportunity for new studies.

3:50

2pMUb2. Musical instruments of Brazilian capoeira: Historical roots, symbolism, and use. Beatriz Ilari (Faculty of Music, McGill Univ., 555 Sherbrooke St. W., Montreal, QC H3A 1E3, Canada, bilari@po-box.mcgill.ca)

This paper describes the historical roots, symbolism, and uses of musical instruments in capoeira. A martial art form of Afro-Brazilian origin, capoeira is rhythmically performed to music in a roda (i.e., circle). Capoeira is at times defined as a martial art form disguised as dance because it is rooted in the struggles of African slaves. Elements of music, dance, fight, and ritual are part of this unique martial art form, which has two main styles: Angola and Regional. Capoeira styles are important as they determine rhythmic patterns, chant, movement, and musical instrumentation in a roda. The leading instrument in all capoeira styles is the berimbau. The instrument dictates the rhythm and movement of capoeira players in a roda (Ilari, 2001). Made out of a wooden stick, a wire, and a gourd and played with a stick and a coin, the berimbau is considered a sacred instrument due to its association with the cry of the slaves. Other instruments used in capoeira are pandeiros, agogo bells, reco-recos, and atabaques. A discussion regarding the use of these instruments within the context of capoeira will be presented at the conference. The incorporation of these instruments into contemporary Brazilian music will also be considered.

4:10

2pMUb3. An acoustic study of the Brazilian cuica. Paul A. Wheeler (Utah State Univ., Logan, UT 84322)

The cuica is a friction drum of African origin played in the batucada (an ensemble of instruments used for the samba) during the Brazilian carnival. It is played by rubbing a bamboo rod which is connected to the center of a drum head, giving a rhythmic grunting sound. Pitch is changed by applying pressure to the membrane. This paper discusses several acoustic aspects of a folk cuica (made of a gourd) including the waveforms, spectra, and time envelopes produced. Rubbing the bamboo rod gives a primitive saw-toothed excitation, similar to a bowed violin string. This is connected to the center of a membrane which modifies and radiates the sound. The body of the cuica contributes little to the sound.

4:30

2pMUb4. Samba and the other sambas: Instrumentation in different forms of Brazil's main musical genre. Pablo Majlis (Faculte de Musique, Universite de Montreal, 109 Chestnut Ave., Pointe-Claire, QC H9R 3B2, Canada, pablomajlis@yahoo.com) and Beatriz Ilari (McGill Univ., Montreal, QC H3A 1E3, Canada)

The aim of the present paper is to describe the instruments of the different forms of samba, their origin, and their uses, focusing on percussion instruments. Samba is a Brazilian popular genre that developed mainly during the 20th century, though being deeply rooted in the precedent centuries of colonization and metissage between the Portuguese colonizers and the Africans that were brought as slaves. From its origins to the present day, samba has branched into multiple forms and instrumentation. Perhaps the most famous samba form is the samba enredo. This type of samba accompanies the Carnival parade in Rio de Janeiro, and features hundreds of percussionists. Another possible samba group instrumentation can be as simple as a single voice and a matchbox played by the singer. Between these two extremes there are several possible formations for a samba group, depending on the social context and function in which it occurs. Different group formations sometimes imply different song forms. Examples include samba de roda (i.e., circle samba), samba de gafieira (i.e., ballroom samba), and samba-cancao (i.e., samba ballad), among others. Some instruments will be available for attendees to try during the conference.

4:50–5:10
An opportunity will be provided for those in attendance to try some of the instruments.

TUESDAY AFTERNOON, 3 DECEMBER 2002   CORAL GARDEN 2 AND 3, 1:00 TO 5:30 P.M.

Session 2pNS

Noise and Architectural Acoustics: Predicting Noise in Indoor Industrial Spaces

Murray R. Hodgson, Cochair
Occupational Hygiene Program, University of British Columbia, 2206 East Mall, Vancouver, British Columbia V6T 1Z3, Canada

Frank H. Brittain, Cochair
Bechtel Corporation, 50 Beale Street, San Francisco, California 94105

Chair's Introduction—1:00

Invited Papers

1:05

2pNS1. Overview of predicting noise levels in indoor industrial spaces. Frank Brittain (Bechtel Corp., 50 Beale St., San Francisco, CA 94105, fhbritta@bechtel.com)

Predicting indoor noise in industrial facilities is a vital part of designing industrial plants. The predicted levels are used in the design process to determine compliance with occupation noise exposure limits, and to estimate levels inside the walls as starting point for predicting community noise radiated by buildings. Once levels are predicted, the noise controls needed can be developed. Special methodologies are needed, because normal room acoustics found in architectural acoustics texts is valid only for nearly empty rooms with limited absorption and ranges of room dimensions. The fittings inside industrial spaces can profoundly affect the propagation of noise and the resulting noise levels. In an industrial space, such as a power plant, there is no such thing as a reverberant field, except in isolated areas. In industrial spaces, including factories, predicting noise levels by summing free and reverberant fields gives erroneous results that are usually overly conservative. This paper discusses normal empty room acoustics, and problems typically encountered when it is applied to industrial spaces, particularly those with a high density of fittings or very large spaces. Also, alternative methodologies for predicting indoor noise levels in industrial spaces, which are based on standards and software, are identified and discussed.

1:35

2pNS2. Acoustics of industrial buildings. Michael Vorlaender (Institut fuer Technische Akustik, Aachen Univ. (RWTH), 52056 Aachen, Germany)

Industrial halls have significantly different shapes as rooms for other purposes. Typically these enclosures are very large and/or flat, which means that the room height is usually very much smaller than width and length. Sound fields in these types of enclosures cannot be expected to have an approximate diffuse sound field. Instead, the decay curves are bent, while sound level versus distance (sound propagation curve) is declining. This contribution summarizes physical properties of sound fields in flat and long rooms, gives some examples of predictions based on image sources, on Kuttruffs integral equation, results of scale model experiments and ray-tracing algorithms with two different levels of complexity. It can be shown that in many cases predictions can yield sufficiently accurate results. However, problems will occur in cases where rooms are heavily filled with scattering objects (fittings). On the other hand, the exact spatial distribution of sound in these nondiffuse cases has shown to have less influence on the perceived annoyance than expected. Accordingly, predictions of sound levels with reasonable accuracy should be sufficient to describe the subjective effect of listening, unpleasantness and annoyance in industrial halls.

2:05

2pNS3. Predicting noise in industrial workrooms using empirical models. Murray Hodgson (School of Occupational and Environ. Hygiene, Univ. of British Columbia, 3rd Fl., 2206 East Mall, Vancouver, BC V6T 1Z3, Canada)

Sound fields in complex industrial workrooms can be predicted well using numerical procedures such as the method of images and ray tracing. However, this requires acoustical expertise, as well as computational resources and times which result in prediction methods only being used in special cases. This paper discusses alternative empirical prediction methods which have the potential to be sufficiently accurate in "typical" cases, and more readily accessible to practitioners, making them more likely to be used in practice. The first method discussed is a hybrid approach, whereby characteristic workroom sound-propagation curves are predicted using ray tracing. These are then input into an empirical model which sums the energy contribution of all sources at a receiver position based on those curves and the applicable source/receiver distances. Next, the development of empirical models for predicting frequency-varying sound-propagation curves and reverberation times using regression techniques is discussed. These were developed from data measured in actual workrooms when empty or fitted, without and with sound-absorptive treatment. Empirical methods for estimating workroom fitting densities and multisource noise levels, and the integration of the empirical models into the PlantNoise prediction system, are also discussed.

2:35

2pNS4. A web-based noise control prediction model for rooms using the method of images. Stephen Dance (School of Eng., South Bank Univ., Borough Rd., London, UK, dances@sbu.ac.uk)

Previous simple models could only predict sound levels in untreated rooms. Now, using the method of images, it has become possible to accurately predict the sound level in fitted industrial rooms from any computer on the Internet. Thus, a powerful tool in an acoustician's armory is available to all, while requiring only the minimal amount of input data to construct the model. This is only achievable if the scope of the model is reduced to one or two acoustic parameters. Now, two common noise control techniques have been implemented into the image source model: acoustic barriers and absorptive patches. Predictions using the model with and without noise control techniques will be demonstrated, so the advantages can be clearly seen in typical industrial rooms. The models are now available on the web, running directly inside Netscape or Internet Explorer.

3:05–3:15 Break

3:15

2pNS5. Application of indoor noise prediction in the real world. David N. Lewis (SEAC, Unilever R&D, Colworth House, Sharnbrook, Bedfordshire MK44 1LQ, UK)

Predicting indoor noise in industrial workrooms is an important part of the process of designing industrial plants. Predicted levels are used in the design process to determine compliance with occupational-noise regulations, and to estimate levels inside the walls in order to predict community noise radiated from the building. Once predicted levels are known, noise-control strategies can be developed. In this paper an overview of over 20 years of experience is given with the use of various prediction approaches to manage noise in Unilever plants. This work has applied empirical and ray-tracing approaches separately, and in combination, to design various packaging and production plants and other facilities. The advantages of prediction methods in general, and of the various approaches in particular, will be discussed. A case-study application of prediction methods to the optimization of noise-control measures in a food-packaging plant will be presented. Plans to acquire a simplified prediction model for use as a company noise-screening tool will be discussed.

3:45

2pNS6. Validation–comparison of predicted and measured levels in industrial spaces. Wolfgang Probst (ACCON GmbH, Grfelfinger Str. 133A, 81375 Munich, Germany)

Financed by the German Federal Agency for Labor and Social Affairs, several methods of calculating the noise propagation in industrial halls were used for about 150 halls and compared with measurements. With all noise sources, such as machines and equipment stopped, a dodecahedron loudspeaker emitting broadband noise was used, and the octave-band levels were measured on different propagation paths. The room geometry and equipment in the room were entered into uniform datasets, the calculation methods were applied to each dataset, and the results in terms of deviations between calculated and measured values were evaluated

statistically. The prediction method with the smallest deviations was chosen for further evaluation. This method, which uses mirror images with approximations and takes into account diffraction with a method first developed by Kuttruff and extended by Jovicic, uses mean values for fittings and absorption at walls and ceilings. This method has been incorporated into VDI 3760, and will be extended in the future to take into account screening by single objects and the real distribution of absorptive materials on surfaces. Representative experimental results, calculation techniques, predicted levels, and deviations between measured and predicted levels are presented.

<center>**4:15–5:30**</center>

<center>**Panel Discussion**</center>

TUESDAY AFTERNOON, 3 DECEMBER 2002   CORAL ISLAND 1 AND 2, 1:00 TO 4:45 P.M.

<center>**Session 2pPA**</center>

<center>**Physical Acoustics: Bubbles, Drops and Foams II**</center>

<center>Joachim Holzfuss, Chair</center>

<center>*Nonlinear Physics Group, Institute of Applied Physics, Technical University of Darmstadt, Darmstadt D-62489, Germany*</center>

<center>***Invited Papers***</center>

<center>**1:00**</center>

2pPA1. Encapsulated bubble dynamics. John Allen III (510 Arthur St., Apt. # 116, Davis, CA 95616, johnallen312@hotmail.com)

The study of a bubble encapsulated by a fluid or an elastic shell is a subject of interest for a wide variety of applications. In particular, ultrasound contrast agents are encapsulated bubbles 1–5 microns in radius developed for diagnostic imaging and, more recently, therapeutic purposes involving drug delivery. The previous formulations of the equations for encapsulated bubbles have originated from a generalized Rayleigh–Plesset equation, which follows from the Eulerian form of the fluid dynamics equations. Dynamic formulations of empty spherical shells in an inviscid medium have been also developed independently of these gas bubble studies. Little effort has gone into unifying the two approaches. The equations for a gas-filled incompressible, isotropic elastic spherical shell are derived from the Lagrangian frame using the first Piola–Kirchoff stress tensor. Some previous results are obtained and compared in different limiting cases. Instabilities of the shell are determined for sufficiently flexible shell materials or high internal gas pressures. Also highlighted are the nonspherical instabilities associated with acoustically driven bubbles with fluid shells of different density and viscosity than that of the surrounding fluid.

<center>**1:25**</center>

2pPA2. Single bubble sonoluminescence: Unstable diffusion and the kinetics of chemical reactions. Joachim Holzfuss (Nonlinear Phys. Group, Inst. of Appl. Phys., Tech. Univ. of Darmstadt, Germany)

Sonoluminescence of a single bubble in water driven by ultrasound is accompanied by long-term stability of the radial bubble wall oscillation. However, in a certain parameter range instabilities of the phenomenon are observed. They are characterized by growing of the ambient (and maximum) radius of the bubble and a sudden microbubble split-off with spatial dislocation (recoil). Experimental results using images of shock waves shed into the surrounding water at bubble collapse are shown to visualize the effect. Numerical calculations show that the observed nonlinear dynamical effects can be interpreted by the influence of diffusional, chemical, and spatial instabilities of the bubble.

<center>**1:50**</center>

2pPA3. Acoustically-driven spherical implosions and the possibility of thermonuclear reactions. D. Felipe Gaitan, Ross A. Tessien (Impulse Devices, Inc., 12731-A, Grass Valley, CA 95945, gaitan@impulsedevices.com), and William C. Mead (Adaptive Network Solutions Res., Inc., Los Alamos, NM)

Acoustically driven, gas-filled cavities in liquids have been known to collapse violently, generating short flashes of light of ∼100 psec duration. This phenomenon has been known as Sonoluminescence (SL) and was first observed by Marinesco *et al.* in 1933. Ten years ago the author pioneered a technique for observing the oscillations of a single, gas-filled cavity (termed Single-Bubble Sonoluminescence) which has provided new insights into the phenomenon of sonoluminescence. More recently, the possibility of generating fusion reactions using acoustics has been considered. Results of computer simulations and preliminary experimental data will be presented. Back-of-the-envelope calculations in terms of the acoustical and thermodynamic parameters necessary to achieve Thermonuclear reactions will be presented in an effort to evaluate the feasibility of Sonofusion as an energy source.

2pPA4. Evidence for nuclear emissions during neutron seeded acoustic bubble cavitation. R. P. Taleyarkhan, C. D. West, J. S. Cho (Oak Ridge Natl. Lab., P.O. Box 2009, Bldg. 9204-1, Oak Ridge, TN 37831), R. T. Lahey, Jr., R. C. Block (Rensselaer Polytechnic Inst., Troy, NY), and R. Nigmatulin (Russian Acad. of Sci., Ufa, Russia)

In cavitation experiments with deuterated acetone, statistically significant tritium decay activity above background levels was detected. In addition, evidence for statistically significant neutron emissions near 2.5 MeV was also observed, as would be expected for deuterium–deuterium fusion. Control experiments with normal acetone did not result in tritium activity or neutron emissions. Hydrodynamic shock code simulations supported the observed data and indicated compressed, hot (10^6–10^7 K) bubble implosion conditions, as required for thermonuclear fusion reactions. Separate experiments with additional fluids are under way and results appear to support those observed with acetone. Scalability potential to higher yields, as well as evidence for neutron−tritium branching ratios are presented.

2:40−3:00   Break

Contributed Papers

3:00

2pPA5. Dispersion relation measurements of acoustic waves in bubbly water. Gregory J. Orris and Michael Nicholas (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375)

Recent theoretical work on the propagation of acoustic waves in bubbly media has highlighted the need for more precise and modern measurements of the relationship between the phase speed and attenuation in bubbly media. During the engineering tests of the new Salt-Water Tank Facility at the Naval Research Laboratory measurements of the dispersion of acoustic waves in fresh water were performed over a broad range of environmental conditions under semi-free field conditions. Large aquaculture aeration tubes were used to create bubble clouds completely filling the facility with bubbles whose radii ranged from a few tens of microns to 1 cm with total void fractions that reached to a few percent. We discuss these experimental results within the context of current theories and their implications on ocean acoustic experiments. [Work supported by ONR.]

3:15

2pPA6. Hydroacoustical interaction of bubble clouds. Stefan Luther and Detlef Lohse (Phys. of Fluids, Faculty of Appl. Phys., Univ. of Twente, The Netherlands)

Acoustically driven cavitation bubble fields consist of typically 10^4 micron-sized bubbles. Due to their nonlinear hydroacoustical interaction, these extended multiscale systems exhibit the phenomenon of spatiotemporal structure formation. Apart from its significance for the theory of self-organization, it plays a major role in design and control of many industrial and medical applications. Prominent examples are ultrasound cleaning, sono-chemistry and medical diagnostics. From a fundamental point of view the key question to ask is ''How does the fast dynamics on small length scales determine the global slow dynamics of the bubble field?'' To clarify the complex interplay of acoustical and hydrodynamical forces acting on the bubbles, we employ high-speed particle tracking velocimetry. This technique allows the three-dimensional reconstruction of the bubbles' trajectories on small and fast scales as well as the measurement of the bubble density on large and slow scales. A theoretical model is derived that describes the nonlinear radial and translational dynamics of the individual bubbles and their interaction. The numerical solution of this N body problem is presented.

3:30

2pPA7. Phase velocity measurements in a bubble swarm using a fiber optic sensor near the bubble resonant frequency. Stanley A. Cheyne (Dept. of Phys. and Astron., Hampden-Sydney College, Hampden-Sydney, VA 23943)

Acoustic phase velocity measurements of a bubble swarm in a cylindrical tube have been made with a fiber optic sensor. The fundamental design of this system is similar to one used in a previous experiment [S. A. Cheyne *et al.*, ''Phase velocity measurements in bubbly liquids using a fiber optic laser interferometer,'' J. Acoust. Soc. Am. **97**, 1621 (1995)].

This new system is more robust and is more easily constructed than the previous system. Results will be presented that will show conclusively that data have been obtained just after the bubble resonance in the regime where the attenuation of sound is very high. Other results will be presented at different air-to-water ratios (void fraction).

3:45

2pPA8. Statistical characteristics of cavitation noise. Karel Vokurka (Phys. Dept., Tech. Univ. of Liberec, Halkova 6, CZ-461 17 Liberec, Czech Republic, karel.vokurka@vslib.cz)

Cavitation noise originates as a superposition of pressure waves emitted during oscillations of individual cavitation bubbles. These pressure waves contain useful information on bubbles generating them and efforts are done to extract it. Unfortunately the pressure waves emitted by different bubbles usually overlap heavily and thus in experiments it makes sense to measure statistical characteristics only. Typical statistical characteristics determined experimentally encompass autospectral densities and instantaneous autospectra. To be able to extract information concerning the oscillating bubbles, suitable models of both cavitation bubbles and cavitation noise are necessary. It has been found out recently that a reasonable insight into the cavitation noise structure may be obtained by simulating cavitation noise on a computer and comparing statistical characteristics of simulated cavitation noise with those determined experimentally. By varying different parameters in theoretical models used to simulate the noise, a good agreement between the simulated and measured cavitation noise statistical characteristics can be obtained. The models parameters thus found may be then analyzed from a physical point of view and conclusions on behavior of cavitation bubbles can be drawn. [Work supported by the Ministry of Education of the Czech Republic as the research Project No. MSM 245100304.]

4:00

2pPA9. Time-scales for quenching single-bubble sonoluminescence in the presence of alcohols. Jingfeng Guan and Thomas Matula (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

A small amount of alcohol added to water dramatically decreases the light intensity from single-bubble sonoluminescence [Weninger *et al.*, J. Phys. Chem. **99**, 14195–14197 (1995)]. From an excess accumulation at the bubble surface [Ashokkumar *et al.*, J. Phys. Chem. **104**, 8462–8465 (2000)], the molecules evaporate into the bubble interior, reducing the effective adiabatic exponent of the gas, and decreasing the bubble temperature and light output [Toegel *et al.*, Phys. Rev. Lett. **84**, 2509–2512 (2000)]. There is a debate as to the rate at which alcohol is injected into the bubble interior. One camp favors the notion that molecules must be repetitively injected over many acoustic cycles. Another camp favors the notion that most quenching occurs during a single collapse. An experiment has been conducted in order to resolve the debate. Quenching rates were measured by recording the instantaneous bubble response and corresponding light emission during a sudden increase in pressure. It was found that

complete quenching in the presence of methanol requires over 8000 acoustic cycles, while quenching with butanol occurs in about 20 acoustic cycles. These observations are consistent with the view that quenching requires the repetitive injection of alcohol molecules over repetitive acoustic cycles.

4:15

2pPA10. A KC-135 experiment for studying single bubble sonoluminescence: Design, fabrication, and results. Ronald A. Roy and R. Glynn Holt (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

A detailed description of the design and fabrication of an apparatus to study SBSL on the NASA's KC-135 parabolic flight aircraft will be presented. The apparatus was used during two recent flights aboard the KC-135 (March and July 2002); data from these flights will be presented. Parameters measured during the flights include the acoustic pressure, ambient pressure, acceleration, bubble size, bubble location, water temperature, and light intensity. All measurements were made in a simultaneous

fashion, and time tagged using the time code from a VCR which was used to record video images of the bubble. A review of and comparison to data from other KC-135 SL experiments will also be given. [Work supported by NASA.]

4:30

2pPA11. High frequency acoustic scattering by a gas bubble. Nail A. Gumerov (UMIACS, Univ. of Maryland, 115 A. V. Williams Bldg., College Park, MD 20742)

A problem of acoustic scattering by a gas bubble when the length of the incident acoustic waves can be comparable with or smaller than the bubble size is considered. For such waves pressure and temperature distributions inside and outside the bubble are not spherically symmetrical even though the bubble is spherical in the absence of the acoustic field and the amplitude of the acoustic field is small. This case is considered in the present study. Genearal three-dimensional pressure and temperature distributions together with capillary effects are taken into account by expansion of solutions for coupled thermal and acoustic problems in series of spherical multipoles. The acoustic response of the bubble including volume and surface oscillations is analyzed. Limits for modeling of the bubble as a sound-soft sphere in acoustic scattering problems are discussed.

TUESDAY AFTERNOON, 3 DECEMBER 2002

GRAND CORAL 3, 1:30 TO 5:00 P.M.

Session 2pPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Potpourri (Poster Session)

Walt Jesteadt, Cochair
Boys Town National Research Hospital, 555 North 30th Street, Omaha, Nebraska 68131

Joel Flores, Cochair
3A Privada de Mimosas No. 617, Villa de las Flores, Coacalco, Edo. de Mexico CP 55710, Mexico

Contributed Papers

All posters will be on display from 1:30 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 p.m. to 2:45 p.m. and contributors of even-numbered papers will be at their posters from 2:45 p.m. to 5:00 p.m.

2pPP1. Computer-based method for the diagnostics and rehabilitation of tinnitus patients. Bozena Kostek, Andrzej Czyzewski, and Henryk Skarzynski (Inst. of Physiol. and Pathol. of Hearing, Pstrowskiego 1, Warsaw, Poland)

The proposed method is an electronic diagnostic and rehabilitation system for people suffering from internal ear noise, namely tinnitus patients and people with an abnormally high hearing sensitivity—hyperacusis patients. Thanks to the method employing multimedia personal computers to the fitting masking sounds to the patient's needs, tinnitus patients can be rehabilitated using the masking or habituation method. The subject is asked to answer detailed questions in the electronic questionnaire and next to identify those sounds which most strongly resemble those they perceive as ear noise. Following an algorithm analysis of the results and analysis of the selection of sounds made by the patient, the computer diagnoses the patient as free from ear noise and hyperacusis or classifies them into a risk group. Next, the patient is informed about the result and can then read about the causes of the ailment and the recom-

mended treatment and also can download compressed sounds applicable in therapy to the rehabilitation tool coupled with the personal computer which is a programmable ear noise masker. In the paper the algorithm of analysis of patients answers and acoustic characteristics of tinnitus maskers are discussed. Hitherto obtained results of this method application are demonstrated.

2pPP2. Principles and acoustical foundations of the computer-based hearing screening method. Henryk Skarzynski, Andrzej Czyzewski, and Bozena Kostek (Inst. of Physiol. and Pathol. of Hearing, Pstrowskiego 1, Warsaw, Poland)

The hearing impairment is one of the fastest growing diseases in modern societies. Therefore it is very important to organize screening tests allowing us to find people suffering from this kind of impairment. The computer-based system was designed to conduct hearing screening,

mainly in children and youth. The test uses automatic questionnaire analysis, audiometric tone test procedure, and testing speech intelligibility in noise. The starting point of the test is an automatic interview with the individual to be tested. Based on the interview, the electronic questionnaire is filled out. After the questionnaire has been filled out and the specially conceived three tone audiometric test is completed, the mode of the speech-in-noise based test might be selected as appropriate for the specific age. When all the testing is completed, the system "I CAN HEAR ..." automatically analyzes the results for every person examined. Based on the number of wrong answers those who may have hearing problems are referred to cooperating medical consulting centers. In the paper foundations and principles of the hearing tests are discussed and results of testing of more than 200 000 children with this method are demonstrated.

2pPP3. The effects of high intensity pure tone on visual field, eye fixation, pupil size, and visual false positive and negative errors. Hashir Aazh, Ali Nouraeinejad, Ali Asghar Peyvandi, and Latif Gachkar (P.O. Box 17445-177, Tehran, Iran, hash_aazh@hotmail.com)

Our aim in this study is to evaluate the effects of pure tone on visual field, pupil size, eye fixation, and visual false positive and negative errors. Thirty-two young adult subjects with normal hearing and normal visual acuity were tested in this study. Measurements were performed over two test sessions. In one session visual factors were measured in quiet, and another measurement was performed during presenting a pure tone (1000 Hz, 100-dB HL) binaurally via a headphone. The statistical program SPSS10 was used for all analyses. Fixation loss was significantly lower (better) in quiet condition compared with continuous pure tone condition, and other visual factors had no significant differences in the two conditions. As a conclusion it can be stated that changes in attentional focus which result from altered levels of arousal or autonomic system activity during presenting a high intensity pure tone affect on fixation of the eye. However, it is suggested that these effects are possibly related to stress caused by sound masking the hearing of speech and other wanted environmental sounds, and not from some direct autonomic system arousal by sound.

2pPP4. On the lateralization of the Huggins pitch. Peter Xinya Zhang and William M. Hartmann (Phys.-Astron., Michigan State Univ., East Lansing, MI 48824 and Biomed. Eng., Boston Univ., Boston, MA 02215)

The central activity pattern (CAP) model of Raatgever and Bilsen [J. Acoust. Soc. Am. **80**, 429–441 (1986)] correctly predicts that Huggins pitch (HP^+) is lateralized in the center whereas HP^- is lateralized to the left or the right. Experiments show that some listeners (left-eared listeners) always hear the pitch sensation on the left and others always hear it on the right. Still others can hear it on one side or the other. The CAP model also predicts that the laterality of HP^- should follow a hyperbolic function of the boundary frequency. To test this prediction, laterality was measured in careful laterality estimation experiments, wherein HP^- was combined with a set of interaural time differences (ITDs). Although laterality estimates followed predictions for finite (ITDs), on those trials where the ITD was zero the hyperbolic law was violated for five out of five listeners. Instead, the laterality of HP^- was very insensitive to the boundary frequency over the range tested, 200 to 1000 Hz. A search for a satisfactory variation on the CAP model continues. [Work supported by the NIDCD under Grants Nos. DC00181 and DC00100.]

2pPP5. Development of the positional presumption system of sound source which combines sound source information and picture image. Yamashita Yasuhiro (4-17-1 Wakasato, Nagano City, Japan)

The development of the positional presumption system of sound sources which combines sound information and picture images has been done by using five microphones. The calculation of the positional presumption of the sound source is obtained by calculating the time difference from the cross spectrum of the output of several microphones. There was a tendency that the direction presumption varies by enlarging the elevation of the position of the sound source and the receiving sound point. It is thought that this cause is influenced by the ground level reflection. We tried to separate the reflected sound and the removed method was shown. In addition, the methods of improving the calculation accuracy in consideration of the acoustic impedance of the ground surface were shown, and compared.

2pPP6. Functional segregation of segmental features, pitch-accents, and nondistinctive suprasegmental features in working memory. Akihiro Tanaka, Koichi Mori (Res. Inst. of Natl. Rehabilitation Ctr. for the Disabled, 4-1, Namiki, Tokorozawa-shi, Saitama 359-8555, Japan, tanaka@rehab.go.jp), and Yohtaro Takano (Univ. of Tokyo, Hongo, Bunkyo-ku, Tokyo 113-0033, Japan)

Japanese has pitch-accent which contribute to lexical distinctiveness just as segmental features. A dual-task experiment was conducted to examine the independence among segmental features, pitch-accents, and nondistinctive suprasegmental features in working memory. Japanese speakers of Tokyo dialect participated in the experiment. The primary task was a working memory task which required the retention of both segmental and suprasegmental features of nonsense words. The suprasegmental features were either regular pitch-accent of Tokyo dialect or pseudopitch changes that do not exist in a Japanese accent pattern. The secondary task was silent mouthing of irrelevant verbal material either without putative pitch change, with pitch accent, or with pseudopitch change, that was performed during the retention period of the primary task. The results revealed selective interference according to distinctivity of pitch patterns in native speakers of Tokyo dialect, suggesting functional segregation between distinctive and nondistinctive suprasegmental features. In contrast, functional segregation was not observed in "non-native" speakers of Tokyo dialect, suggesting that the prosody of the non-native dialect acts as "second language." Additionally, in both groups, suprasegmental processing did not interfere with the retention of segmental features. The results suggest that there are at least three substores for segmental features, pitch-accents, and nondistinctive suprasegmental features.

2pPP7. Electroacoustic verification of FM benefits in advanced hearing aid circuitry. Erin C. Schafer and Linda M. Thibodeau (Adv. Hearing Res. Ctr., Univ. of Texas, Dallas, 1966 Inwood Rd., Dallas, TX 75235, eschafer@utdallas.edu)

Miniature FM receivers attached to ear-level hearing aids can provide significant improvements in speech recognition in noisy environments. These receivers are designed to couple to the hearing aid via a boot connection with limited or no adjustment of hearing aid settings. Ideally, the circuitry allows the FM signal to be approximately 10 dB more intense than the typical signal from the hearing aid microphone, allowing for an FM advantage. Furthermore, for intense input levels, the output limiting should not differ for the hearing aid compared to the hearing aid and FM receiver combined, i.e., FM transparency. Following procedures recommended by the American Speech, Language, and Hearing Association, the electroacoustic responses of digital and conventional aids were measured with and without coupling to FM systems at typical conversational input levels and maximum input levels. The rms difference between the electroacoustic responses of the hearing aid alone and coupled to the FM was used to quantify the FM advantage at typical input levels and the FM

transparency at high input levels. The finding of great variability in both FM advantage and transparency supports the need for additional fitting controls or design modifications to obtain the maximum FM benefit.

2pPP8. A comparison of middle ear acoustic admittance in adults and 3-week-old infants based on multifrequency tympanometry. Linda Polka (School of Commun. Sci. and Disord., McGill Univ., 1266 Pine Ave. W., Montreal, QC H3G 1A8, Canada, linda.polka@McGill.ca), Navid Shahnaz (Univ. of British Columbia, BC, Canada), and Anthony Zeitouni (McGill Univ., Montreal, QC, Canada)

The assessment of newborn hearing requires information on middle ear status yet the interpretation of tympanometry in newborns is unclear. This study aims to further our understanding of acoustic admittance in the newborn middle ear. Multifrequency tympanograms were recorded from sixteen 3-week-old infants (30 ears) and sixteen young normal-hearing adults (30 ears). Tympanometry was conducted using the Virtual 310 middle ear analyzer using 9 probe tone frequencies between 226 and 1000 Hz at roughly 100 Hz intervals. All infants passed a hearing screening using automated ABR (Algo II) shortly after birth (within 24 h) and again at 3-weeks of age. At 226 Hz, admittance tympanograms had a single peak in all adult ears while 60% of infant ears had multiple peaks or irregular patterns. At 1000 Hz admittance tympanograms had a single peak for 74% of infant ears while 78% of adult ears showed multiple peak or irregular patterns. Analyses of tympanometric shape (using the Vanhuyse classification scheme), as well as static admittance, static susceptance, and static conductance also reveal differences in adult and infant middle ear function. Implications for the clinical application of tympanometry in the first month of life will be discussed.

2pPP9. Estimates of the strength of repetition pitch in infants. Marsha G. Clarkson (Dept. of Psych., Georgia State Univ., University Plaza, Atlanta, GA 30303-3083, mclarkson@gsu.edu), Cynthia M. Zettler, Michelle J. Follmer, and Michael J. Takagi (Georgia State Univ., Atlanta, GA 30303)

To measure the strength of the pitch of iterated rippled noise (IRN), 24 7- to 8-month-olds and 24 adults were tested in an operant conditioning procedure. To generate IRN, a 500-ms Gaussian noise was delayed by 5 or 6 ms (pitches of 200 and 166 Hz) and added to the original noise for 16 iterations. IRN stimuli having one delay were presented repeatedly, and on signal trials the delay changed for 6 s. Overall stimulus level roved from 63–67 dBA. Infants learned to turn their heads toward the sound, and adults learned to press a button when the delay of the stimulus changed. Testing started with IRN stimuli having 0 dB attenuation (i.e., maximal pitch strength). Then, stimuli having weaker pitches (i.e., progressively greater attenuation applied to the delayed noise) were presented. Strength of pitch can be quantified as the maximum attenuation for which pitch can be discerned. For each subject, threshold attenuation for pitch strength was extrapolated as the 71% point on a psychometric function depicting percent-correct performance as a function of attenuation. Mean thresholds revealed that the pitch percept was significantly weaker for infants (6.9 dB) than for adults (19.1 dB).

2pPP10. Noise addiction. Fernando J. Elizondo-Garza (Acoust. Lab., FIME, Univ. A. de Nuevo Leon, P.O. Box 28 "F," Cd. Universitaria, San Nicolas, 66450, N.L., Mexico, fjelizon@ccr.dsi.uanl.mx)

Progress in understanding the brain processes involved in addiction to chemicals has resulted in the recognition of other forms of addiction related to energy and information, i.e., TV and Internet. In that context noise becomes a cause of addiction and, therefore, it is necessary to develop more adequate approaches even to conceptualize noise addiction. In this presentation some of the main aspects to be considered to approach an operational definition of noise addiction are discussed.

2pPP11. Effect of noise on social relationships. Karla Panuszka (Acoust. Consulting, 2929 Beverly Ln., Lafayette, IN 47904) and Ryszard Panuszka (Staszic Univ., AGH Krakow, Poland)

Sociology is the scientific study of human society and social behavior. The effect of noise and vibrations under unwanted vibroacoustical fields shows important changes in social interactions. A new social question arises from these studies: What effect does unwanted noise pollution play on human environments and how are relationships changed? It has been observed in workers that noise and vibrations play an important factor in machinery safety. Music shows a positive effect on relationships by providing a person's mental attitude with a happy euphoria. Natural sounds of waterfalls, streams, and rivers also show an improved mood that enhances relationships in a positive way. Staszic University AGH has documented these interactions under the influence of acoustical fields. The human ear cannot hear sound waves below 20 Hz, but these waves have been shown to have effects on people's interactions. The main focus of sociology is interactions of large populations; thus, noise interactions need to be studied. Three areas of investigations could arise from these studies: family relationship responses, occupational safety, and domestic and business violence. Broad investigation results will be displayed.

2pPP12. Validation of audiometric method for measuring temporary threshold shifts. Rodrigo Ordoñez, Ioanna-Rigina Karariga, Brian L. Karlsen, Karen Reuter (Dept. of Acoust., Aalborg Univ., Denmark, rop @acoustics.auc.dk), and Dorte Hammershøi (Aalborg Univ., Denmark)

The proposed experiment aims at testing the audiometric method developed earlier [Ordoñez *et al.*, Acta Acustica (Beijing) **88**, 450–452 (2002)] by evaluating if it is able to reproduce some of the known aspects of temporary threshold shifts (TTS), such as: Immediate sensitization, the 2 min maximum of the recovery and the half-octave shift of the peak of TTS. The subjects will be exposed to pure tones of 500 Hz and 2 kHz for 2 to 5 min and the levels will cover the range between 40- to 100-dB SPL. The aim is to induce a maximum of 15 dB of TTS. For showing the half-octave shift, the threshold will be determined from one octave below the exposure frequency to one octave above in 1/2 octave steps. For the 2 min maximum, the threshold will be measured at the most affected frequency and it can be determined continuously for 4 min after the exposure. Sensitization at probe frequencies lower than the exposure frequencies will be tested by measuring the threshold at 1/2 and 1 octave below the exposure frequency.

2pPP13. Scientific discovery of the function of ear in the light of material property. Hari S. Paul and M. Kumaresan (Int. Res. Inst. for the Deaf—A component of Acoust. Foundation, 94/159 Avvai Sanmugam Salai, Chennai-600014, India, mirc@md3.vsnl.net.in)

Piezoelectricity is conversion of mechanical energy to electric energy and vice versa. This property exhibits in noncentrocymetric materials. Bone is well-known piezoelectric material in living body. Eardrum is connected with bones (malleus, incus, stapes, and bony cochlea). Cochlea is snail-shaped and filled with fluid and hair cells (human electrodes). Fluid is centrocymetric; hence, it is nonpiezoelectric material. Acoustic pressure on eardrum sets mechanical energy to these bones. Bones convert mechanical energy to electric polarization, which is direct piezoelectric property. Electric charges generated in bones are transmitted through fluid as $\pm$ ions (like a car battery charger) and picked up by $-/+$ hair cells and

auditory nerves for transmission to brain. Transmissions of ±charges through fluid generate movement of fluid. Cochlea replacement is required when hair cells in cochlea lose their power to transmit charges. Vocal cord surrounded by cricoid and arytenoid cartilages is known as vocal box/larynx. Larynx transforms pulse energy to sound energy (converse piezo-electric property). Vocal cord narrowing and opening its air passage from lungs can also produce sound. Present concept of ear function contradicts majority of doctor's view, that is, fluid in cochlea transforms sound (mechanical) energy to electric energy, which is untrue.

2pPP14. Tempo discrimination of isochronous tone sequences: The multiple-look model revisited.

Nathaniel Miller and J. Devin McAuley (Dept. of Psych., Bowling Green State Univ., Bowling Green, OH 43403)

Previous research has shown that increasing the number of intervals in an isochronous tone sequence reduces tempo discrimination thresholds [C. Drake and M. Botte, Percept. Psychophys. **54**, 277–286 (1993)]. One question that arises is whether increased tempo sensitivity in this instance is attributed to multiple looks at the standard interval, comparison interval, or both. The present study addressed this question by examining tempo discrimination using isochronous tone sequences that contained variable numbers of standard and comparison intervals. In all cases, participants judged the tempo of the comparison sequence relative to a standard sequence (responding faster or slower). Preliminary results suggest that in some cases increases in tempo sensitivity are more due to repetitions of the comparison interval than to repetitions of the standard. The implications of these findings for theories of auditory tempo discrimination will be discussed.

2pPP15. Listening strategies used by normal-hearing adults during loudness estimation.

Lori J. Leibold and Lynne A. Werner (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA 98105-6246)

This study describes individual differences in the weight normal-hearing adults give to the frequency components of a complex sound when performing loudness estimation. In addition, the relationship between listening strategy and loudness was examined. A multi-tone complex (1000, 2000, and 4000 Hz) was presented for 500 ms to the left ear of seven normal-hearing adults. The level of each tone was selected independently and randomly from a rectangular distribution ranging in 10 dB steps from 40 to 80 dB SPL on each of 500 presentations. In the nonselective condition, subjects provided a numerical estimate of the loudness of the complex. In selective conditions, listening strategy was controlled by instructing subjects to attend to a single target frequency. Relative weights for each subject were estimated by normalizing the raw correlations calculated from the level of each component and the subjects magnitude estimate for each trial. Results for the nonselective condition revealed individual differences in subjects' weighting functions. In contrast, similar weighting functions were observed in the selective conditions, with subjects giving the greatest weight to the target frequency. The loudness growth data can best be described by weighting the intensity of each frequency component prior to determining the loudness growth.

2pPP16. Melody recognition with temporal and spectral cues in normal-hearing and cochlear-implant listeners.

Ying-Yee Kong (Dept. of Cognit. Sci., Univ. of California, Irvine, CA 92697) and Fan-Gang Zeng (Univ. of California, Irvine, CA 92697)

Cochlear-implant users can achieve a high level of speech recognition, but their ability to appreciate music is severely limited. This study investigates the relative contribution of the temporal and spectral cues to

melody recognition. Two sets of 12 familiar songs were generated: one set contained both rhythm and melody information (the rhythm–melody condition), whereas the other set contained only melody information (the no-rhythm condition, where all notes had the same duration). Normal-hearing listeners achieved 95%–100% accuracy in both conditions. While cochlear-implant listeners achieved 42%–94% accuracy in the rhythm–melody condition, they performed essentially at chance level in the no-rhythm condition. To further identify the cues used in melody recognition, temporal envelopes from the original broadband signal were extracted and modulated with white noise. When the rhythm cue was available, the normal-hearing listeners achieved 60%–100% recognition, which was similar to the performance achieved by the cochlear-implant listeners. When the rhythm cue was not available, both normal-hearing and cochlear-implant listeners performed essentially at chance level. These results suggest that the present cochlear-implant listeners relied solely on temporal information to recognize familiar melodies. The fine structure information is certainly needed to allow true appreciation of music for cochlear-implant users.

2pPP17. The effects of aging on spatial hearing in listeners with normal hearing.

Janet Koehnke, Joan Besing, Ianthe Dunn-Murad, and Caryn Neuvirth (School of Grad. Medical Education, Seton Hall Univ., 400 S. Orange Ave., South Orange, NJ 07079)

This study was designed to examine the effects of aging on spatial hearing in young, middle-aged, and older adults with normal hearing. In order to determine the onset and progression of age-related changes in spatial hearing, performance was measured for three groups of listeners: 18 to 30 years, 38 to 50 years, and over 60 years. All listeners had hearing thresholds of 25 dB HL or better at octave frequencies from 250 through 8000 Hz. A series of virtual tests were administered including spatial localization in quiet, spatial localization in noise, spatial detection in noise, and speech intelligibility gain. For all of these tests, the source locations and listening environments were simulated using head-related-transfer-functions. Results indicate no clear or consistent differences between the young and middle-aged groups on any of the spatial tasks. In contrast, localization errors, spatial detection thresholds, and speech intelligibility thresholds are poorer for the older listeners than for the young and middle-aged listeners. It is noteworthy that all three groups obtain comparable gain in spatial detection thresholds and speech intelligibility thresholds in some listening conditions when the signal and noise sources are spatially separated. [Work supported by NIH/NIDCD Grant No. DC004402.]

2pPP18. Binaural phase masking experiments in stereo audio.

Alexander I. Iliev and Michael S. Scordilis (Dept. of Elec. and Computer. Eng., Univ. of Miami, 1251 Memorial Dr., Coral Gables, FL 33124-0640)

Researchers have established that in binaural hearing the smallest detectable angular separation between two sources, commonly referred to as the minimal audible angle (MAA), for a pair of sources on the horizontal plane depends on the frequency of the emitted pure tone and the azimuth angular separation between the sources. One interesting way is to view the sources' angular perturbation within the MAA limits as noise in the phase domain, and the listener inability to detect this perturbation as the result of a masking process. The present discussion focuses on experimental procedures for examining the perception of MAA and the corresponding interaural phase difference (IPD) when complex sound sources are located in the most sensitive region, which is directly in front of the observer (both azimuth and elevation angles at 0° degrees). Sound stimuli were viewed as the linear combination of pure tones, as provided by Fourier analysis. Results indicate that masking is achieved when the IPD is disturbed within

a threshold limit corresponding to the MAA for pure tone sources. Listening tests using stereo audio further validated our observations. [Work supported by Watermark Technologies.]

2pPP19. Distinguishing feature misperception from illusory conjunctions in spatially distributed musical tones. Michael D. Hall and Kimberly Wieberg (Psych. Dept., Univ. of Nevada, Las Vegas, 4505 Maryland Pkwy., Box 455030, Las Vegas, NV 89154, hallm@unlv.edu)

Recent questions have been raised in the visual search literature concerning whether illusory conjunctions of correctly registered features occur (indicating a feature integration process) or are an artifact of feature misperception. The current investigation raised and addressed similar questions for findings of auditory illusory conjunctions using simultaneous, spatially distributed musical tones. Two experiments were conducted where musically trained listeners identified pairs of tones that reflected possible combinations of two instrument timbres with two pitches. Experiments differed in the spatial separation between simultaneous tones to potentially evaluate the effects of distance on feature perception/integration. In Experiment 1 tones were presented to opposing ears. In Experiment 2 tones were only slightly lateralized by a manipulation of interaural time disparities. Conjunction responses, reflecting the incorrect combination of features, frequently occurred, and were more common for slightly lateralized tones. To evaluate the perceptual event(s) responsible for conjunction responses, data were submitted to multinomial models that differed with respect to whether or not they allowed for illusory conjunctions, the misperception of features, or both errors. Across experiments, data fitting by models was improved by feature misperception, but was not further improved by illusory conjunctions. Implications for models of search performance and feature binding are discussed.

2pPP20. Dichotic pitch and the missing fundamental. Joseph Hall III, Emily Buss, and John Grose (Dept. of Otolaryngol., Univ. of North Carolina at Chapel Hill, 610 Burnett-Womack Bldg., CB 7070, Chapel Hill, NC 27599-7070)

A dichotic pitch can be heard by introducing interaural phase shifts in one or more low-frequency spectral regions of an otherwise diotic noise. The present study examined the perceived pitch associated with phase disparities in spectral regions corresponding to harmonics of a low-frequency fundamental. Of particular interest was whether the pitch of the complex was related to the frequency region of the harmonics or whether it was associated with the missing fundamental. Discrimination thresholds for this dichotic virtual pitch were estimated in a 3AFC paradigm, where listeners were asked to identify the interval with the lower virtual pitch. After modest training, listeners were able to perform this task well even when a randomized subset of harmonics above $F0$ were presented on each trial, a manipulation designed to make spectral region an unreliable cue. In another manipulation, frequency regions of the phase disparity were chosen such that although the frequency increased, the missing fundamental decreased (or vice versa). In these cases listeners reported that the perceived pitch shifted according to the change in $F0$, not according to the spectral region of the harmonics. Results were consistent with the interpretation that the perceived pitch correspond to the missing fundamental.

2pPP21. Auditory performance in an open field. Kim Fluitt, Tomasz Letowski, and Timothy Mermagen (U.S. Army Res. Lab., Auditory Res. Team, Bldg. 520, Aberdeen Proving Ground, MD 21005, kfluitt@arl.army.mil)

Detection and recognition of acoustic sources in an open field are important elements of situational awareness on the battlefield. They are affected by many technical and environmental conditions such as the type of sound, the distance to a sound source, terrain configuration, meteorological conditions, hearing capabilities of the listener, the level of background noise, and the listeners' familiarity with the sound source. A limited body of knowledge about auditory perception of sources located over long distances makes it difficult to develop models predicting auditory behavior on the battlefield. Our purpose in the present study was to determine the listeners' abilities to detect, recognize, localize, and estimate distances to sound sources from 25 to 800 meters from the listening position. Data were also collected for meteorological conditions (wind direction and strength, temperature, atmospheric pressure, humidity) and background noise level for each experimental trial. Forty subjects (men and women, ages 18–25) participated in the study. Nine types of sounds were presented from six loudspeakers in random order; each series was presented four times. Partial results indicate that both detection and recognition declined at distances greater than approximately 200 m and distance estimation was grossly underestimated by listeners. Specific results will be presented.

2pPP22. Comparing some convolution-based methods for creation of surround sound. Andrzej Czyzewski and Bozena Kostek (Sound & Vision Eng. Dept., Gdansk Univ. of Technol., Narutowicza 11/12, 80-952 Gdansk, Poland)

Spatialization of the sound using the multichannel techniques is now getting widespread. One can derive many rules for surround sound recording and reproduction. However, there exists only few methods suitable for recording sound in large auditoria ensuring its proper subsequent reproduction in small reproduction rooms, preserving spatial properties of sound acquired in the original recording location. Some experiments presented in the paper were devoted to the simulation of acoustics of the recording hall using the convolution of a monophonic audio signal with the multichannel impulse response of the hall. A special microphone setup was created for this task and an original method of recording multichannel impulse response of auditory halls was conceived and implemented. In this method the acoustical signal recorded quasianechoically was convolved with five impulse responses of the simulated room measured in the room corners and on the stage. Another examined method which is more standard employed convolution of monophonic signals with long-term averaged HRTFs (Head-Related Transfer Funtions). Surround recordings made with both mentioned convolution techniques were then compared on the basis of subjective testing results. The details of the examined surround recording methods and results of their assessments will be discussed in the paper. [Work supported by KBN, Grant No. 8 T11D 00218.]

2pPP23. Auscultation in noise: A program to develop a stethoscope capable of functioning during aeromedical transport. Paul A. Cain, William A. Ahroon (US Army Aeromedical Res. Lab., Bldg. 6901, Ft. Rucker, AL 36362), John M. Sewell, and William N. Bernhard (Active Signal Technologies, Linthicum Heights, MD)

U.S. Army helicopters used to provide aeromedical transport are extremely noisy (105 dB). This prevents auscultation with current stethoscopes. Ten subjects used four different stethoscopes in noise at 70–100 dB in order to determine the detection threshold of a body sound. The stethoscopes differed in performance ($p = 0.001$); the best being the acoustic, followed by the three electronic stethoscopes. The threshold of noise for the detection of heart and breath sounds was 80 dB and 70–75 dB, respectively. A comparison of a standard stethoscope and one modified with Communications Ear Plugs revealed a difference ($p = 0.05$) at 70 and 80 dB that disappeared at higher noise levels, implying that it was not simple masking, but that noise was amplified after entering the sensing head. Despite the poorer performance of the electronic stethoscopes, the need for a gain of 30–35 dB indicated that this was the preferred route for development. A sensor with an acoustic impedance match close to body tissue, but poorly matched to air-coupled noise, was developed, and trials indicate that heart sounds can be heard at 105 dB. Development continues with the aim of reducing or eliminating the need for signal processing.

Session 2pSAa

Structural Acoustics and Vibration: Vibration of Floors of Buildings

Eric E. Ungar, Chair

Acentech, Incorporated, 33 Moulton Street, Cambridge, Massachusetts 02138-1118

Chair's Introduction—1:00

Invited Papers

1:05

2pSAa1. The art of building floor vibration evaluation. Thomas M. Murray (Dept. of Civil and Environ. Eng., Virginia Tech, Blacksburg, VA 24061)

Annoying floor vibration caused by building occupant activity is an increasingly common occurrence. Optimized floor systems and the advent of the electronic office are the main causes. Because humans are very sensitive to vertical movement, especially in quiet environments, careful floor design is required. When this is not done or when the occupancy does not meet the design criteria, complaints are commonly received. The complex nature of human excitations (walking, jumping, running, exercising) makes the evaluation of problem floors as much art as science. Typical floor motions include accelerations caused by frequencies above the human threshold limit, which must be considered in any evaluation. Further, most floor systems exhibit several closely spaced modes, which make remedial measures difficult to implement. Structural modifications, passive control in the form of tuned mass dampers, and active control are all potential remedies but a choice is highly dependent on the particulars of the problem floor and the occupancy. This paper discusses techniques for evaluating human induced floor motion and gives examples of successful and unsuccessful retrofits.

1:35

2pSAa2. Diagnosing a case of occupant-induced whole building vibration. Linda M. Hanagan (Dept. of Architectural Eng., Penn State Univ., 104 Eng. Unit A, University Park, PA 16802)

People in the tenth floor office suite of a 10-story building were complaining of annoying floor vibrations. These vibrations were worse on some days than on others and seemed to be emanating from a dance studio on the floor below. The building owners wanted the problem fixed; however, the exact mode of transmission to the tenth floor was, as yet, unknown. Understanding how the vibration was being transmitted was essential to developing a repair solution. Among the possibilities for transmission were a full height partition on the ninth floor, the curtain wall, column flexure, and column shortening. Through vibration testing, it was determined that a whole building mode, with an estimated equivalent mass of almost 1 000 000 kg, was excited to cause disturbing levels of vibration at the tenth floor with as few as six people jumping on the floor below. Details of the vibration testing are provided.

2:05

2pSAa3. Estimation of vibrations due to walking on floors that support sensitive equipment. Eric E. Ungar and Jeffrey A. Zapfe (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138-1118)

The development of the extensively used simple method for predicting footfall-induced vibrations of floors of buildings (American Institute of Steel Construction Design Guide 11) is reviewed. The empirical basis and underlying analytical assumptions of the method are delineated and critiqued. Its limitations are discussed and suggestions for its extension are presented. Results of some recent related measurements are summarized.

2:35

2pSAa4. Vibrations of raised access floors. Hal Amick, Michael Gendreau, and Colin G. Gordon (Colin Gordon and Assoc., 883 Sneath Ln., Ste. 150, San Bruno, CA 94066)

Raised access floors play a critical role in modern cleanroom design. They have unique mechanical properties that make them respond to dynamic loading in a manner quite different from conventional floors. For example, an unbraced floor is much more flexible horizontally than in the vertical direction. Horizontal vibration amplitudes with walker excitation may exceed 100 μm/s in an unbraced floor, exceeding the sensitivity of 1000$\times$ inspection microscopes by as much as an order of magnitude. Issues such as these become important when moderately vibration-sensitive instruments, such as optical microscopes, are supported on access floors, typically the case in cleanrooms. This paper presents results of experimental studies involving a 3 m$\times$3 m segment of floor and a large floor installed in a cleanroom, both of which were subjected to dynamic loads using a shaker. Both drive-point and propagation properties were examined. In addition, data are presented for variations in bracing and bolting using the 3 m$\times$3 m segment.

2p TUE. PM

2pSAa5. Vibration sensitivity of a laboratory bench microscope. Hal Amick (Colin Gordon and Assoc., 883 Sneath Ln., Ste. 150, San Bruno, CA 94066) and Matthew Stead (Bassett Acoustics, Kent Town SA 5067, Australia)

Bench-mounted optical microscopes have a wide variety of applications in science and technology. The vibration sensitivity is a function of both magnification and support conditions. In this paper we present the results of experimental studies addressing vibration sensitivity as well as the amplification and attenuation provided by typical laboratory casework. The benchtop vibration amplitudes at which the effects of motion first become perceptible were found for magnifications of $40\times$, $100\times$, $400\times$, and $1000\times$ using sinusoidal excitation. Frequency response functions were determined for benchtop motion with respect to floor motion, using both conventional casework and a popular pneumatic isolation bench. Floor vibration criteria were developed for microscopes with the two types of support.

TUESDAY AFTERNOON, 3 DECEMBER 2002　　　　　　　CORAL GALLERY FOYER, 3:50 TO 4:50 P.M.

Session 2pSAb

Structural Acoustics and Vibration: Vibration Abatement

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

Contributed Papers

3:50

2pSAb1. Vibration analysis and building vibration isolation design for an automated people mover system in airport terminal structures. James E. Phillips (Wilson, Ihrig & Assoc., Inc., 5776 Broadway, Oakland, CA 94618)

Detailed Finite Element Analysis (FEA) models were developed for a proposed airport terminal expansion project. An Automated People Mover (APM) system is incorporated into the airport structures for shuttling passengers quickly between terminals. The dynamic forces imparted onto the structures by the moving APM vehicles and the analysis approach were based upon established techniques developed for addressing ground-borne and structure-borne vibrations from rail systems. Measurements were conducted at two other major airports with existing rubber tire APM systems on aerial structures. These measurements provided baseline vibration levels for the analysis as well as the forces imparted to the structure by the APM vehicles. The results of the analyses were utilized in the design of a vibration isolation system included in the structural design of a guideway within a terminal building in order to mitigate structure-borne vibration from APM operations.

4:05

2pSAb2. Helmholtz design for noise transmission attenuation on a chamber core composite cylinder. Deyu Li and Jeffrey S. Vipperman (Dept. of Mech. Eng., Univ. of Pittsburgh, 531 Benedum Hall, Pittsburgh, PA 15261)

This work explores the feasibility of using Helmholtz resonators to attenuate a subscale ChamberCore cylinder noise transmission. The ChamberCore cylindrical composite is an innovative new sandwich-type structure. It consists of an outer skin, an inner skin, and linking ribs. There are wedge-cross-section chambers along the axis direction between the outer and inner skins. These chambers provide a potential for the acoustic Helmholtz resonator design in order to reduce the noise transmission, which is dominated by the internal acoustic cavity. In this experimental work, the sound transmission behavior of the ChamberCore fairing is investigated and divided into four interesting frequency regions: the stiffness-controlled zone, cavity resonance-controlled zone, coincidence-controlled zone, and mass-controlled zone. It is found that the noise transmission in the low-frequency band is controlled by the structural stiffness and cavity resonances, where the acoustic Helmholtz design method has the potential to improve the noise transmission.

4:20

2pSAb3. Evidence of the existence of phononic band gaps: A practical example of a tunable sound insulation by a periodic device of rods. Cecile Goffaux, Philippe Lambin, Jean-Pol Vigneron (LPS, Facultés Universitaires Notre-Dame de la Paix, 61 rue de Bruxelles, B-5000 Namur, Belgium, cecile.goffaux@scf.fundp.ac.be), and Fabrizio Maseri (Ctr. de Conception Solutions Acier pour la Construction, B-4000 Liege, Belgium)

Based on the principle of phononic band gap materials, the control of acoustic frequency gaps by altering the geometry of the system is analyzed in the particular case of a set of parallel solid square-section columns distributed in air on a square lattice. This system is shown to be sensitive enough to the rotation of the columns to be considered for a practical sonic band gap width engineering. For different geometric configurations, experimental and theoretical results are presented and a discussion about the application of such structures as sound insulators is discussed. We acknowledge the use of the Namur Scientific Computing Facility (Namur-SCF), a common project between FNRS, IBM Belgium, and the Facultés Universitaires Notre-Dame de la Paix (FUNDP). [C. Goffaux acknowledges the financial support of the FIRST program of the Walloon Region Government and the Research and Development Centre of Cockerill-Sambre (ARCELOR Group).]

4:35

2pSAb4. Physics of a pneumatic vibration isolator revisited. Vyacheslav M. Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA 92606, vryaboy@newport.com)

This paper gives detailed consideration to two phenomena that affect the performance of a pneumatic isolator but have not received adequate attention in the past. The first one is the thermal conductivity of gas. It is usually assumed that the compression of gas in the pneumatic chamber is an adiabatic process. Thermal conductivity can, nevertheless, affect the performance, especially in the low-frequency domain and for small-size isolators. A simple explicit expression for the acoustic compliance is derived from exact solution of conductive gas equations in the cylindrical domain. Another factor affecting the isolator response is the stiffness introduced by the diaphragm. A formula for the stiffness is derived based on the mathematical model of the diaphragm as an elastic membrane. Comparison to experimental data shows that adequate representation of these two factors results in accurate predictions of the isolator performance as a function of the supported load.

Session 2pSP

Signal Processing in Acoustics and Speech Communication: Feature Extraction and Models for Speech

Jose A. Diaz, Cochair
Universidad de Carobobo, Valencia, Venezuela

Shrikanth Narayanan, Cochair
Signal and Image Processing Institute, University of Southern California, 3740 McClintock Avenue, Los Angeles, California 90089-2564

Chair's Introduction—1:00

Invited Papers

1:05

2pSP1. Feature extraction and models for speech: An overview. Manfred Schroeder (Univ. of Goettingen, Buergerstrasse 42–44, 37073 Goettingen, Germany)

Modeling of speech has a long history, beginning with Count von Kempelens 1770 mechanical speaking machine. Even then human vowel production was seen as resulting from a source (the vocal chords) driving a physically separate resonator (the vocal tract). Homer Dudley's 1928 frequency-channel vocoder and many of its descendants are based on the same successful source-filter paradigm. For linguistic studies as well as practical applications in speech recognition, compression, and synthesis (see M. R. Schroeder, Computer Speech), the extant models require the (often difficult) extraction of numerous parameters such as the fundamental and formant frequencies and various linguistic distinctive features. Some of these difficulties were obviated by the introduction of linear predictive coding (LPC) in 1967 in which the filter part is an all-pole filter, reflecting the fact that for non-nasalized vowels the vocal tract is well approximated by an all-pole transfer function. In the now ubiquitous code-excited linear prediction (CELP), the source-part is replaced by a code book which (together with a perceptual error criterion) permits speech compression to very low bit rates at high speech quality for the Internet and cell phones.

1:30

2pSP2. New pitch detection system for concurrent speech on a parallel processing system. Yoshifumi Chisaki (Dept. of Computer Sci., Faculty of Eng., Kumamoto Univ., 2-39-1 Kurokami, Kumamoto 860-8555, Japan), Akira Nakagawa (Kumamoto Univ., Kumamoto, Japan), Hidetoshi Nakashima (Kumamoto Natl. College of Technol.), Tsuyoshi Usagawa (Kumamoto Univ., Kumamoto, Japan), and Masanao Ebata (Kumamoto Natl. College of Technol., Japan)

Pitch detection system based on the harmonic wavelet transform (HWT) algorithm for concurrent speech [Y. Chisaki, 6C.15, Proc. ICA2001] has been proposed. The HWT algorithm has two major advantages. One is to detect a pitch from a frame signal, which is a short duration, such as 20 ms. The other one is robustness against noise; namely, all pitches can be detected from concurrent speech in a short delay, simultaneously. However, processing time for the modified correlation method, as a conventional method is 8.6 times faster than that for the HWT based system. Due to the development of computers, processing time can be reduced easily by using PCs. This paper proposes a new pitch detection system for concurrent speech with a new parallel algorithm. The system is based on the HWT method, and implemented on a parallel processing system. Simulations are performed with some node arrangements. As a result, a processing speed for a parallelized block of the system is 6.3 times faster as that for the original one without being accompanied by degradation of pitch detection accuracy. [Part of this work is supported by The Sagawa Foundation for Promotion of Frontier Science, the Cooperative Research Project Program of the RIEC, Tohoku Univ., and the Ono Acoustics Research Fund (2001).]

1:55

2pSP3. Methodology for analyzing spectral differences. Howard Rothman (Dept. of Commun. Sci. and Disord., Univ. of Florida, P.O. Box 117420, Gainesville, FL 32611) and Jose Diaz (Universidad de Carabobo, Valencia, Venezuela)

The human voice can be an exceptional instrument. The larynx contributes significantly to making up the exceptional voice: it provides the speaking/singing fundamental frequency (SFF) and the resultant harmonic structure. The vocal tract modulates energy and enhances the contributions of the larynx. There is evidence that the singers/speakers formant provides brilliance and increased amplitude to the voice. Several papers have examined the effects of the spectrum on the perception of the aging voice, the charismatic voice and the professional voice. Spectral differences can be seen in voices that have been identified as good/bad, young/old, brilliant/ordinary. One of the authors presented data differentiating between good and bad vibrato samples that aid in identifying singers that are experiencing vocal difficulty. However, there are singers, identified by vocal pedagogues, critics and other singers as

2p TUE. PM

experiencing vocal difficulty, whose vibrato remained unchanged over time. While changes in vibrato are most probably laryngeal in nature, there are no data available as to the changes in the spectra of these singers that provide the perceptual cue for their vocal problems. This paper will present methods detailing the spectral differences and their contributions to perceptual judgments.

2:20

2pSP4. Novel features for robust speech recognition. Alexandros Potamianos   (Bell Labs, Lucent Technologies, 600 Mountain Ave., Murray Hill, NJ 07974)

Recently there has been much research in the area of robust front-ends and new features for automatic speech recongition (ASR). These efforts have had limited success for certain databases and recording conditions. In this work, we review some recent work on features for ASR: the articulatory front-end of Li *et al.* (2000), nonlinear modulation features [Quatieri (2002); Dimitriadis and Maragos (2002)], chaotic features [Pitsikalis and Maragos (2002)], short-time spectral moments [Paliwal *et al.* (2000)], etc. We extend the work of Potamianos and Maragos (2001) to show how some of these features relate to the standard front-end of short-time smooth spectral envelope. We also analyze some of these new features using classification and regression trees to show the relevance of the features for phone-classification task.

2:45−3:00   Break

3:00

2pSP5. Convolutive mixture separation in time−frequency domain for robust automatic speech recognition. Shubha L. Kadambe   (HRL Labs., LLC, 3011 Malibu Canyon Rd., Malibu, CA 90265)

In a mobile environment, automatic speech recognition (ASR) systems are being used for information retrieval. Due to the presence of multiple speakers, noise sources and reverberations in such environments, the ASR performance degrades. Here, the problem of improvement of ASR performance by separating the convolutively mixed speech signals that predominantly exist in mobile environments is addressed. For the separation, an extension of the algorithm published in A. Ossadtchi and S. Kadambe ["Over-complete blind source separation by applying sparse decomposition and information theoretic based probabilistic approach," ICASSP, 2001] is applied in the time–frequency domain. In the extended algorithm, the dual update algorithm that minimizes L1 and L2 norms simultaneously is applied in every frequency band. The problem of channel swapping is also addressed. The experimental results of separation of convolutively mixed signals indicate about 6-dB SNR improvement. The enhanced speech signals are then used in GMM based continuous speech recognizer. The recognition experiments are performed using the real speech data collected inside a vehicle. During the presentation, complete ASR performance improvement results will be provided.

3:25

2pSP6. Linguistically informed automatic speech recognition. Carol Y. Espy-Wilson   (Dept. of Elec. Eng., Univ. of Maryland, A. V. Williams Bldg., Cambridge, MA 20742)

The development of an automatic event-based speech recognition system (EBS) that relies heavily on acoustic phonetics (to guide the recognition process and to extract relevant information) and combines a phonetic-feature hierarchy with a uniform statistical framework (at present, Support Vector Machines) to provide adaptability and flexibility is currently under way. This recognition framework allows for easy assessment and distinction of the performances of the acoustic parameters versus that of the pattern recognizer. The overall structure of EBS involves (1) landmark detection based on acoustic parameters that are related to the source and manner-of-articulation phonetic features and (2) use of the landmarks in the extraction of other acoustic parameters related to the place-of-articulation phonetic features. This talk will focus on the development of the acoustic parameters and the need for relative parameters for speaker independence, multi-time-scale processing to capture the dynamics of phonetic segments and extensive evaluation of the parameters to hone in on direct measures of the relevant acoustic properties. [Work supported by NSF and NIH.]

3:50

2pSP7. Graphical models and automatic speech recognition. Jeff A. Bilmes   (Univ. of Washington, Seattle, WA 98195-2500)

Graphical models (GMs) are a flexible statistical abstraction that has been successfully used to describe problems in a variety of different domains. Commonly used for ASR, hidden Markov models are only one example of the large space of models constituting GMs. Therefore, GMs are useful to understand existing ASR approaches and also offer a promising path towards novel techniques. In this work, several such ways are described including (1) using both directed and undirected GMs to represent sparse Gaussian and conditional Gaussian distributions, (2) GMs for representing information fusion and classifier combination, (3) GMs for representing hidden articulatory information in a speech signal, (4) structural discriminability where the graph structure itself is discriminative, and

the difficulties that arise when learning discriminative structure (5) switching graph structures, where the graph may change dynamically, and (6) language modeling. The graphical model toolkit (GMTK), a software system for general graphical-model based speech recognition and time series analysis, will also be described, including a number of GMTK's features that are specifically geared to ASR.

Contributed Papers

4:15

2pSP8. Fundamental frequency estimation using signal embedding in state space. Dmitry Terez (SoundMath Technologies LLC, 6 N. 9th St., Millville, NJ 08332, terez@soundmathtech.com)

A new robust nonlinear method for determination of fundamental frequency ($F0$) was recently proposed with application to speech pitch detection [Terez, Proc. ICASSP **1**, 345–348 (2002)]. The method uses state-space embedding technique originally introduced for analyzing chaotic signals. The new method has been generalized and tested on different types of speech signals, as well as on a variety of other acoustic signals. In addition, some artificially generated nonstationary and complex wave forms have been used to test the limits of the method in comparison with other known (short-term) $F0$-estimation techniques (e.g., correlation, spectrum or cepstrum-based methods). Evaluation results demonstrate a unique combination of properties distinguishing the new method from conventional techniques. In particular, reliable and accurate $F0$ estimates can be obtained for clean periodic signals using signal segments slightly longer than one complete fundamental period. Other properties include immunity to speech formants and robust performance on noisy and band-limited speech signals. Some improvements are introduced to reduce the number of required computations and to achieve higher (subsample) accuracy. The method waused to implement a robust pitch-tracking algorithm for speech processing applications. Further information and demo software can be found at http://www.soundmathtech.com/pitch.

4:30

2pSP9. Human factor cepstral coefficients. Mark D. Skowronski and John G. Harris (Computational Neuro-Eng. Lab., Univ. of Florida, Gainesville, FL 32611, markskow@cnel.ufl.edu)

Automatic speech recognition (ASR) is an emerging field with the goal of creating a more natural man/machine interface. The single largest obstacle to widespread use of ASR technology is robustness to noise. Since human speech recognition greatly outperforms current ASR systems in noisy environments, ASR systems seek to improve noise robustness by drawing on biological inspiration. Most ASR front ends employ mel frequency cepstral coefficients (mfcc) which is a filter bank-based algorithm whose filters are spaced on a linear-log frequency scale. Although center frequency is based on a perceptually motivated frequency scale, *filter bandwidth is set by filter spacing and not through biological motivation*. The coupling of filter bandwidth to other filter bank parameters (frequency range, number of filters) has led to variations of the original algorithm with different filter bandwidths. In this work, a novel extension to mfcc is introduced which decouples filter bandwidth from the rest of the filter bank parameters by employing the relationship between filter center frequency and critical bandwidth of the human auditory system. The new algorithm, called human factor cepstral coefficients (hfcc), is shown to outperform the original mfcc and two popular variations in several ASR experiments and noise sources.

4:45

2pSP10. Vowel landmark detection. Andrew W. Howitt (Res. Lab. of Electron., MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

Landmark based speech processing is a component of Lexical Access From Features (LAFF), a novel paradigm for feature based speech recognition. Detection and classification of landmarks is a crucial first step in a LAFF system. Vowel landmarks are detected using an existing syllabic segmentation algorithm with several novel extensions that incorporate du-

rational information, absolute energy level, and $F1$ track information. The detector is scored against the TIMIT database, using a novel algorithm to convert the segmental transcriptions to a landmark representation for scoring. Previous experiments have validated the predictions of acoustic theory, specifically the presence of $F1$ peak in vowels, and demonstrated that amplitude peak in a fixed frequency band is practically as good as a formant tracker. Substantial improvements in performance are achieved by optimizing the fixed frequency band to the $F1$ range, and use of a trainable neural network to combine the multiple acoustic cues. The neural network can also be used to generate confidence scores for detected landmarks, which provide vital information for later stages of processing.

5:00

2pSP11. A Gaussian-selection-based preclassifier for speaker identification. Marie A. Roch (Dept. of Computer Sci., San Diego State Univ., San Diego, CA 92182-7720)

For practical reasons driven by the need to periodically adapt speaker models or enroll new members of the population, most speaker identification systems train individual models for each speaker. When classifying a speech token, the token is scored against each model and the maximum *a posteriori* decision rule is used to decide the classification label. Consequently, the cost of classification grows linearly for each token as the population size grows. When considering that the number of tokens to classify is also likely to grow linearly with the population, the total work load increases exponentially. In this work, a new system is presented which builds upon the so-called "Gaussian selection" techniques. The system uses the speaker-specific models as source data and constructs N-best hypotheses of speaker identity. The N-best hypothesis set is then evaluated using individual speaker models. This process results in an overall reduction of workload. The cost of the model generation is low enough to permit enrollment and adaptation, and the accuracy of the preclassifier is such that there is minimal impact on the recognition rate.

5:15

2pSP12. Quality of service (QoS) on public telephonic networks for multimedia transmission systems. Salvador Alvarez-Ballesteros and Miguel Alvarez-Rangel (IPN–ESIME, Zacatenco Edif. #1 Col., Lindavista, Mexico)

The object of this paper is to determine which QoS maximum can be obtained by applying commercial telecommunications technology in the transmission of audio and video on public telephone lines and the Internet, procuring to achieve the necessary quality to reach the conditions required on H323. It is certain that the top technology that allows to assure that the QoS is most demanding, are a reality. However, these technologies alone pay off practically under conditions that are not very feasible for formalizing would be of the ambient control of the laboratory. In this work we analyzed the technical problems of the telecommunications that affect the yield of the links for applications of multimedia on the Internet.

5:30

2pSP13. The challenges of archiving networked-based multimedia performances (Performance cryogenics). Elizabeth Cohen (Cohen Acoustical, Inc., 132 S. Lucerne Blvd., Los Angeles, CA 90004-3725 and CCRMA/Dept. of Elec. Eng., Information Systems Lab., Stanford Univ., CA, akustik@attbi.com), Jeremy Cooperstock (McGill Univ., Montreal, Canada), and Chris Kyriakakis (Univ. of Southern California)

Music archives and libraries have cultural preservation at the core of their charters. New forms of art often race ahead of the preservation infrastructure. The ability to stream multiple synchronized ultra-low latency streams of audio and video across a continent for a distributed interactive performance such as music and dance with high-definition video and multichannel audio raises a series of challenges for the architects of digital libraries and those responsible for cultural preservation. The archiving of such performances presents numerous challenges that go beyond simply recording each stream. Case studies of storage and subsequent retrieval issues for Internet2 collaborative performances are discussed. The development of shared reality and immersive environments generate issues about, What constitutes an archived performance that occurs across a network (in multiple spaces over time)? What are the families of necessary metadata to reconstruct this virtual world in another venue or era? For example, if the network exhibited changes in latency the performers most likely adapted. In a future recreation, the latency will most likely be completely different. We discuss the parameters of immersive environment acquisition and rendering, network architectures, software architecture, musical/choreographic scores, and environmental acoustics that must be considered to address this problem.

TUESDAY AFTERNOON, 3 DECEMBER 2002

GRAND CORAL 1, 1:00 TO 5:35 P.M.

Session 2pUW

Underwater Acoustics: Geoclutter and Boundary Characterization II

Purnima Ratilal, Cochair

Department of Ocean Engineering, Massachusetts Institute of Technology, 77 Massachusetts Avenue, Room 5-435, Cambridge, Massachusetts 02139

Nicholas C. Makris, Cochair

Department of Ocean Engineering, Massachusetts Institute of Technology, 77 Massachusetts Avenue, Room 5-204, Cambridge, Massachusetts 02139

Chair's Introduction—1:00

Contributed Papers

1:05

2pUW1. The Geoclutter Experiment 2001: Remote acoustic imaging of sub-bottom and seafloor geomorphology in continental shelf waters. Nicholas C. Makris, Purnima Ratilal, Yisan Lai, Deanelle T. Symonds (MIT, 77 Massachusetts Ave., Cambridge, MA 02139, makris@mit.edu), Lilimar A. Ruhlmann, and Edward K. Scheer (MIT, Cambridge, MA 02139)

In the Geoclutter experiment of April–May 2001, an active sonar system was used to remotely and rapidly image geomorphology over wide areas in continental shelf waters by long-range echo sounding. The bistatic system, deployed in the strataform area south of Long Island, imaged extensive networks of buried river channels and inclined subseafloor strata over tens of kilometers in near real time. Bathymetric relief in the strataform area is extremely benign. The vast majority of features imaged apparently correspond to sub-bottom geomorphology that sound waves reach after tunneling as well as propagating through the overlying sediment. Returns from buried river channels were often found to be as discrete and strong as those from calibrated targets placed in the water column. Since buried river channels are expected to be ubiquitous in continental shelf environments, sub-seafloor geomorphology will play a major role in producing "false alarms" or clutter in long-range sonar systems that search for submerged objects such as underwater vehicles or marine mammals. Wave guide scattering and propagation are inherent to this new remote sensing technology because source signals are transmitted over hundreds of water-column depths in range to image sub-seafloor and seafloor geomorphology.

1:20

2pUW2. Coherent versus diffuse surface and volume Reverberation in an ocean wave guide: Reverberation rings, modal decoupling, and possible fish scattering in Geoclutter 2001. Purnima Ratilal and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139, makris@mit.edu)

Conditions necessary for the field scattered from volume inhomogeneities and surface roughness in a wave guide to become diffuse in nonforward azimuths are derived from Green's theorem. Diffuse scattering leads to a decorrelation of the wave guide modes that simplifies the expression for the intensity of the total scattered field. When diffuse scattering conditions are not satisfied, seafloor scattering becomes coherent and leads to the formation of "reverberation rings" sometimes observed in high-resolution sonar systems. The formation of these rings will be investigated both analytically and with simulations. The possibility that some prominent clutter events in the Geoclutter 2001 experiment are due to diffuse scattering from fish schools is investigated theoretically and experimentally.

1:35

2pUW3. Fish schools as potential clutter and false targets: Observations on the New Jersey shelf. Redwood W. Nero, Charles H. Thompson (Naval Res. Lab., Stennis Space Center, MS 39529-5004), and Richard H. Love (BayouAcoust., Pass Christian, MS 39571-2111)

Fish schools can appear as clutter or false targets on search sonars and can confuse the interpretation of scattering from the sea floor. During Boundary Characterization 2001, a high-frequency echosounder was used

2280   J. Acoust. Soc. Am., Vol. 112, No. 5, Pt. 2, November 2002

Pan-American/Iberian Meeting on Acoustics   **2280**

to quantify fish schools in an effort to provide estimates of false targets and clutter at low frequency. Schools were quantified from the echo-sounder data using an image processing algorithm designed to provide estimates of school size, acoustic intensity, and a variety of diagnostic features. The number of schools that had the potential to be low-frequency false targets was estimated using information on fish species obtained from fisheries research trawls and an NRL swimbladder scattering model. A few pelagic fish schools of intermediate size and high intensity, with low-frequency target strengths estimated at $+12$ dB, occurred near the sea surface at night in the northern corner of the study site, at a density of about one per km^2. These schools were most likely to appear as false targets. Demersal fish, those near the sea floor, although abundant along the 80-m contour, were not likely to be strong false targets at low frequency. [Work supported by ONR.]

1:50

2pUW4. Model for coherent scattering from a network of buried river channels in a stratified ocean waveguide. Sunwoong Lee, Purnima Ratilal, and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139, makris@mit.edu)

A theoretical model of three-dimensional bistatic scattering from a network of buried river channels in a continental shelf waveguide is presented. The scattered field on the river channel walls is estimated to be locally specular using the Kirchhoff approximation. This field is then propagated out to a distant receiving array using Green's theorem and the waveguide Green function, so that it is also valid for scattering in the near field of the river channel network. This model is applied to calculate the scattered and beamformed field from a network of buried river channels in the East Coast strataform area, whose geomorphology was mapped out by numerous geophysical surveys. The modeled scattered field is found to be dependent upon the bistatic orientation and projected area of the river channel walls relative to the source and receiver. To verify the modeling, the result of the simulation is compared with data from the Geoclutter 2001 acoustic experiment where images of buried river channel networks were acquired from long-range using a bistatic sonar system.

2:05

2pUW5. Evidence for single-mode propagation and its implications for range-dependent geoacoustic inversion in the SWAT experiment. George V. Frisk (Appl. Ocean Phys. & Eng. Dept., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, gfrisk@whoi.edu) and Kyle M. Becker (Penn State Univ., State College, PA 16804)

During the Shallow Water Acoustic Technology (SWAT) experiments, which were conducted in October 2000 on the New Jersey Shelf, a 20 Hz pure tone was transmitted from a source in 73 m of water to two drifting receivers. The behavior of the pressure magnitude and phase out to ranges of about 8 km suggest that the field is dominated by a single normal mode. In particular, an adiabatic modal interpretation of the phase variation with range is consistent with an independent autoregressive spectral estimate of the wave number content of the field. It is then shown, within the context of an isovelocity bottom model with varying water depth, that the measured range-varying bathymetry does not account for the range dependence in the single modal eigenvalue. On the other hand, an alternative interpretation of the data in terms of a wave guide with constant depth and range-varying sound speed in the bottom yields reasonable estimates of the lateral seabed variability. [Work supported by ONR.]

2:20

2pUW6. Effects of measured bathymetry and subbottom variability on low-frequency shallow-water reverberation. Altan Turgut and Roger Gauss (Naval Res. Lab., Acoust. Div., Washington, DC 20375)

A pseudo-spectral numerical method is used to study the effects of bathymetry and subbottom variability on shallow-water reverberation at low frequencies (less than 500 Hz). Bathymetry and subbottom variability at New Jersey Shelf Geoclutter and Boundary Characterization 2001 experimental sites were measured by using chirp sonar and sediment core data at different wavelength scales. A stochastic bottom simulator was incorporated with the measured deterministic bottom/subbottom features, including dipping layers, erosional channels, and buried river channels. Three-dimensional simulations were also performed to study the effects of anisotropic bottom/subbottom variability on the shallow-water reverberation. It has been shown that, for the Geoclutter and Boundary Characterization 2001 experimental sites, predicted anisotropy in the acoustic scattering field is mainly due to the deterministic features rather than due to anistropic small-scale variabilities in the sediment. [Work supported by ONR.]

2:35

2pUW7. Analysis of the effects of sea floor and sub-sea floor geology on underwater sound propagation estimation and performance prediction. Richard Katz and Michael Sundvik (NUWC Newport, 1176 Howell St., Newport, RI 02841)

In this study, we investigate shallow water sound propagation measurements and interactions with the seabed as a function of geologic composition and waveform characterization. We have analyzed sound transmission loss measurements in the shallow acoustic channel from the geoclutter experiments conducted in 2001 and 2002 and found them to be in good agreement with standard modeling results. The sound propagation through the acoustic channel and interactions with the boundaries lead to energy spreading losses, which make acoustic performance prediction and modeling uncertain. We quantify these losses to the extent possible based on the signals transmitted during the experiments. The contributions to and effects on sound propagation variability from sound speed profile (degree of downward refraction) are discussed with emphasis placed on interactions with the bottom and sub-bottom geology. Analysis is conducted for signals in the 400 Hz to 4 kHz range.

2:50

2pUW8. Range-dependent modal eigenvalue estimation in the SWAT experiment. Luiz L. Souza (MIT/WHOI Joint Prog. in Oceanogr./Appl. Ocean Sci. and Eng., 77 Massachusetts Ave., Rm. 5-435, Cambridge, MA 02139), George V. Frisk (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), and Kyle M. Becker (Penn State Univ., State College, PA 16804)

Three modal mapping experiments (MOMAX) have been conducted in shallow water environments: two in the East coast STRATAFORM site off the New Jersey coast; and one in the Gulf of Mexico off the Florida coast. A low-frequency source emits a set of CW signals in the range 20–500 Hz, and the field is measured by drifting buoys. The ultimate goal of these experiments is to invert the acoustic field data in water for the sound speed profile in the seabed. The analysis has focused on measuring the modal characteristic wavenumbers (eigenvalues) as a function of position. In general, one has to resort to range-dependent spectral analysis. Techniques based on both the short-term Fourier transform and high-resolution parametric spectrum estimation have been used for measuring eigenvalues. New results from the year 2000 SWAT (MOMAX III) experiment at the STRATAFORM site will be discussed. These include a comparison of data obtained from tracks along and across the New Jersey shelf. [Work supported by ONR and WHOI Academic Programs Office.]

be seen. Simulations using the parabolic equation to mimic spatial variations of bottom properties confirm that the bottom loss inferred from this type of measurement is indeed a local one rather than a regional average.

3:20

2pUW9. Geoacoustic measurements on the New Jersey shelf. Charles Holland (Appl. Res. Lab., Penn State Univ., State College, PA 16804)

Two key acoustic parameters for predicting shallow water reverberation are bottom reflection and bottom scattering strength. Historically, seabed reflection and scattering have been treated in a disconnected fashion both in measurement and analysis. A coupled approach to reflection and scattering was conducted during the GeoClutter and Boundary2001 Experiments. By coupled, it is meant that the reflection and scattering measurements are not only colocated, but sample commensurate spatial scales of the seabed. It is also meant that the processing and analyses employ commensurate assumptions and modeling. The coupled nature of the measurements provides the opportunity for determining the physical mechanisms responsible for the reflection and scattering. Deterministic and stochastic geoacoustic properties obtained from an analysis of these mechanisms show very high spatial variability across the measurement area. [Work supported by the NATO SACLANT Undersea Research Center and ONR.]

3:35

2pUW10. Statistical characterization of geologic clutter observed on the STRATAFORM. John R. Preston and Douglas A. Abraham (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804)

Together with SACLANTCEN, the authors recently participated in the 2001 Geoclutter Experiment to study shallow water bottom reverberation in the STRATAFORM off New Jersey. Sources were bistatic and monostatic coherent pulses near 400 Hz. The receivers were horizontal arrays. The STRATAFORM is known to have benign surface morphology but contains many buried river channels and other subsurface horizons. Some highlights of the reverberant returns are discussed that include the correlation of returns with the buried river channels and some other subbottom features. The main objective of this study is the statistical characterization of the geologic clutter. The K distribution has been shown to be useful in describing non-Rayleigh behavior. The clutter from STRATAFORM is described by its K-distribution parameters as a function of location and sediment type. [Work supported by ONR Code 32, Grant N00014-97-1-1034.]

3:50

2pUW11. Geoacoustic inversion of ambient noise. Chris H. Harrison (SACLANT Undersea Res. Ctr., Viale San Bartolomeo, 400, 19138 La Spezia, Italy, harrison@saclantc.nato.int)

The vertical directionality of ambient noise is strongly influenced by seabed reflections. Therefore, geoacoustic parameters can be inferred by inversion of the noise. In this approach, using vertical array measurements (16 m aperture), the reflection loss is found directly by comparing the upward with the downward going noise. Theory suggests that this simple ratio is, in fact, the power reflection coefficient (a function of angle and frequency). A layer model and a search is required to find geoacoustic parameters, but no such model or search is required for reflection loss alone. Experimental data have been gathered at 11 sites, 10 in the Mediterranean, and 1 on the New Jersey Shelf during BOUNDARY2001. Site to site variations are discussed, and comparisons are made with simple layer models. Usually the vertical array is bottom-moored, but at two of the Mediterranean sites the array was allowed to drift over a few miles so that bottom properties could be surveyed. Bottom variations could indeed

4:05

2pUW12. Geoacoustic inversion using genetic algorithm. Itaru Morishita, Hirotaka Murakami (Oki Electric Industry Co., Ltd., 4-10-3, Shibaura, Minato-ku, Tokyo 108-8551, Japan, morishita555@oki.com), Kazuhiko Ohta, Kouki Okabe, and Masamichi Oikawa (TRDI, Yokosuka 239-0826, Japan)

Geoacoustic parameters of bottom sediment have a great influence on acoustic propagation in a water column. In order to estimate the parameters, the genetic algorithm (GA) was applied to the low frequency acoustic data obtained at the SWAT (Shallow Water Acoustic Technology) experiments which were conducted at the East China Sea and the New Jersey Shelf. In the experiments, complex acoustic pressure were measured while distance between the source, which transmitted the cw signal, and the receiver was changed. Horizontal wave number spectrum were obtained from the measured data by synthetic aperture process and the Hankel transform. For estimating geoacoustic parameters, an error function was defined as the difference between the measured spectrum and a calculated one which is obtained by applying the Hankel transform to calculated complex pressure, and GA searched geoacoustic parameters which minimize the error. To evaluate the estimated parameters, transmission loss was calculated using the parameters and was compared with measured data.

4:20

2pUW13. Geoacoustic inversion of broadband acoustic recordings of a surface ship on an horizontal line array in the Gulf of Mexico. Robert A. Koch and David P. Knobles (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

As part of a comprehensive experiment to support research on geoacoustic inversion techniques, acoustic signals from a surface ship were recorded on a bottom mounted, horizontal line array at a soft sediment, shallow-water site in the Gulf of Mexico off Port Aransas, TX. Along with the surface ship recordings, water sound-speed profiles, cw TL data at 53, 103, 153, 503, and 953 Hz, and recordings of light bulb implosions were obtained. In this paper a method for performing inversions from broadband beam data is demonstrated and compared with corresponding inversions from broadband element data and with previously reported inversions of the light bulb implosions. The performance in a simulated annealing algorithm of cost functions that involve coherent or incoherent sums over frequency and multiple time segments is presented, and the uncertainties in the geoacoustic parameter values are examined. Comparisons of measured TL with predicted TL for the bottom descriptions resulting from the inversions are used to judge validity of the inverted profiles and the efficacy of the various cost functions. The applications of this work to ocean environments for which the bottom parameter values are unknown or uncertain will be discussed. [Work supported by ONR.]

4:35

2pUW14. Finite-element technique for multi-fluid 3-D two-way propagation in ocean waveguides. Mario Zampolli, Finn B. Jensen, David S. Burnett, and Carlo M. Ferla (SACLANT Undersea Res. Ctr., Viale S. Bartolomeo 400, 19138 La Spezia, Italy)

A finite-element technique is presented which makes it possible to treat 3-D forward- and backward scattering in underwater waveguides with inhomogeneous layers of fluid and sediment of arbitrary geometry. The

finite-element multi-fluid scattering model was derived from the linear wave equation via the Galerkin residual formulation. From the numerical model, software was developed by customizing an hp-adaptive finite-element package developed by COMCO/Altair Engineering in Austin, Texas. Based on automatic error estimation, it is possible to adapt the user-supplied inhomogeneous computational mesh by making the element size smaller, so-called h-refinement, and/or by increasing the order of the approximating polynomials, so-called p-enrichment, in selected regions of the domain. This process is repeated until a desired level of accuracy in the numerical solution is eventually achieved. The code is validated against a coupled normal-mode model for two-dimensional geometries, and results for shallow-water two-way propagation and scattering with strongly three-dimensional features are presented. The finite-element software is particularly useful for the prediction of scattering from geometries with high local complexity, and as a benchmarking tool for possible future three-dimensional underwater-propagation models.

4:50

2pUW15. Seafloor characterization from inversion of high-frequency backscatter. Frank W. Bentrem, John Sample, and Will Avera (Marine Geosciences Div., Naval Res. Lab., Stennis Space Center, MS 39529, frank.bentrem@nrlssc.navy.mil)

Geoacoustic inversion of high-frequency backscatter data is presented for characterization of the seafloor. The APL-UW (Applied Physics Laboratory at the University of Washington) backscattering model is used to model the grazing-angle dependence of the backscattering strength for a number of seafloor parameters. Backscattering strength versus grazing angle data sets (with frequencies 12–35 kHz) are provided for three sites along with sediment ground truth. Roughness measurements are also available for two of the sites. Inversion of these data sets is performed via simulated annealing, with some of the parameters constrained by empirical relationships with mean grain size. From the inversion, estimates are obtained for mean grain size, roughness, and volume interation. Inversion of data from a smooth, silty site in Arafura Sea yields estimates in good agreement with ground truth. Results also compare well with ground truth in a rough, sandy site (Quinault) and in Onslow Bay (sand). [Work supported by SPAWAR.]

5:05

2pUW16. Modal inversion results for geoacoustic properties in the SWAT experiments. Kazuhiko Ohta, Kouki Okabe, Masanichi Oikawa (5th Res. Ctr., TRDI, JDA 3-13-1 Nagase, Yokosuka 239-0826, Japan), Itaru Morishita, Hirotaka Murakami (Oki Electric Industry Co., Ltd., Tokyo 108-8551, Japan), and George V. Frisk (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Bottom sediment properties at sites on the New Jersey Shelf and in the East China Sea were studied in the Shallow Water Acoustic Technology (SWAT) experiments. In these experiments, a source towed at constant depth transmitted low-frequency cw signals, which were measured on a bottom-moored vertical line array. The Hankel transform was applied to the acoustic field measured on the resulting synthetic aperture horizontal array created at each receiver depth. The horizontal wave number spectra, with peak positions corresponding to the modal eigenvalues, were observed to be slightly different among the different receiver depths, partially due to noise and range dependency. Thus stochastic mode inversion was exploited by using all of the observed peak positions for estimation of the geoacoustic properties. The sound field simulated using the inversion results agrees well with the measured one for each receiver depth.

5:20

2pUW17. Source tow geoacoustic inversions from the 2001 ASIAEX East China Sea experiment. Chen-Fen Huang and William S. Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA 92093-0238, chenfen@mpl.ucsd.edu)

During the 2001 ASIAEX East China Sea experiment, source tow data were collected by a 16-element, 75-m aperture, autonomously recording vertical line array in 105-m-deep water. Transmissions from two similar 6-km-long tracks have been analyzed. In the first, CW tonals at 95, 195, 295, 395, 805, 850, and 905 Hz were transmitted from a J-15 transducer at a nominal depth of 46 m. In the second, CW tonals at 1.6, 2.4, 3.5, and 4.4 kHz were transmitted from an ITC-2015 transducer at a nominal depth of 49 m. Inversion results for seafloor geoacoustic parameters from these transmissions will be presented. [Work supported by ONR.]

2p TUE. PM