waves. The estimates of the order of magnitude of sound intensities were made indicating that the main part of the infrasound generated by the surface waves is absorbed in the upper layers of the atmosphere, resulting in the heating of these layers. The other part of the sound energy can be trapped by the atmospheric acoustic waveguide and then returned to earth at distances of hundreds of kilometers, producing the voice of the sea.

11:40

4aUWb5. Tangent-plane approximation by L. M. Brekhovskikh and connected methods in the theory of wave scattering from rough surfaces. Alexander G. Voronovich (NOAA, Earth System Res. Lab., Physical Sci. Div., 325 Broadway, Boulder, CO 80305, alexander.voronovich@noaa.gov)

Starting from pioneering work by Rayleigh in 1907, scattering of waves from rough surfaces was restricted by the case of small Rayleigh parameter. In this case perturbation analysis describing the process of Bragg scattering applies. Apparently, smallness of the roughness is too restrictive for many applications. In 1952 L. M. Brekhovskikh suggested a tangent-plane approximation (TPA). For ideal boundary conditions it represents the first iteration of the appropriate boundary integral equation. However, for more complex situations (e.g., dielectric or solid-fluid interfaces) appropriate boundary integral equations are rather complicated and, even worse, they cannot be readily iterated. The TPA allows bypassing this step providing the answer in closed form for arbitrary boundary conditions and for scalar or vector waves in terms of the local reflection coefficient. Unfortunately, the TPA does not correctly describe the Bragg scattering. However, later it was realized that the TPA allows simple generalization, which treats both low- and high-frequency limits within single theoretical scheme. This is achieved by considering the local reflection coefficient as an operator rather than a factor. New methods going beyond the two classical ones with much wider regions of validity were developed based on this idea. Some of them will be reviewed in this talk.

FRIDAY AFTERNOON, 1 DECEMBER 2006 LANAI ROOM, 1:00 TO 4:45 P.M.

Session 4pAA

Architectural Acoustics: Measurement of Room Acoustics II

Boaz Rafaely, Cochair
Ben Gurion Univ., Electrical and Computer Engineering Dept., 84105 Beer Sheva, Israel

Hideo Miyazaki, Cochair
Yamaha Corp., Ctr. for Advanced Sound Technologies, 203 Matsunokijima, Iwata, Shizuoka 438-0192, Japan

Contributed Papers

1:00

4pAA1. Impulse response measurements based on music and speech signals. Wolfgang Ahnert, Stefan Feistel, Alexandru Miron, and Enno Finder (Ahnert Feistel Media Group, Berlin, Germany)

All known software based measurement systems, including TEF, MLSSA, SMAART, and EASERA, derive results using predetermined excitation signals like Sweep, MLS, or Noise. This work extends the range of excitations to natural signals like speech and music. In this context selected parameters like frequency range, dynamic range, and fluctuation of the signal and the signal duration are investigated in order to reach conclusions about the conditions required to obtain results comparable with standard excitation signals. Also the limitations of the standard stimuli and the proposed natural stimuli are discussed.

1:15

4pAA2. Assessment of reverberation time in halls through analysis of running music. David Conant (McKay Conant Brook Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, dconant@mcbinc.com)

The source signal to excite a room's reverberant field sufficient for detailed measurement of reverberation time (RT60) and other measures has been the subject of considerable investigation over several decades. It is generally acknowledged that the best sources are (depending on the researcher) swept tones, MLS, MLS variations, stopped noise, cannon shots, etc. All can be characterized as highly audience unfriendly. In the interest of obtaining useful approximations of measured midfrequency RT60 in the presence of live audiences, this paper discusses several approaches that may be fruitful while being entirely unobtrusive to the concert experience.

1:30

4pAA3. Comparison of measurement techniques for speech intelligibility. Bruce C. Olson (Olson Sound Design, 8717 Humboldt Ave N, Brooklyn Park, MN 55444, bco@olsonsound.com)

A comparison of measurement techniques for speech intelligibility between two recently released measurement systems is made. EASERA (Electronic and Acoustic System Evaluation and Response Analysis) uses a standard PC and an EASERA Gateway interface attached via Firewire. The software postprocesses a variety of stimuli in order to derive the impulse response for the room under test. This impulse response is then further processed and the results are presented to the user in both graphical and textual presentations. The Ivie Technologies IE-35 is based on a Pocket PC system and uses an external modulated noise source as stimulus to produce an intelligibility score as a single number or average of a series of measurements. This paper will explore a variety of measurements made in the same locations in a room by both systems. Results will also be shown for a variety of other acoustic measures that quantify the acoustical parameters of the room.

1:45

4pAA4. Under-balcony acoustics in concert halls: Single source versus an array of multiple sources. Youngmin Kwon and Gary W. Siebein (Architecture Technol. Res. Ctr., Univ. of Florida, 134 ARCH, P.O. Box 115702, Gainesville, FL 32611, ymkwon@ufl.edu)

The conventional measurement protocol using a single omnidirectional sound source may have limits or uncertainty in objective acoustical analysis of a performance hall. This study conducted monaural and binaural impulse response measurements with an array of 16 directional loudspeakers for quantitative acoustical assessment of specifically under-balcony area in a concert hall. The measurements were executed in a real performance hall. The measured time- and frequency-domain responses as well as the results of room acoustical parameters including binaural parameters were compared to the ones measured with a single omnidirectional source. The results were also compared to the ones taken at the main orchestra seating area. The time-domain responses showed a clear distinction particularly in early responses between single source and multiple sources. On the other hand, the magnitude of frequency response showed significantly lower at frequencies above 1 kHz than the one measured at the main area. The results of a binaural parameter, IACC, were found to be marginal between single source and multiple sources but critically different between under-balcony area and main area. Variations were also observed in the results of other room acoustical parameters when compared either between single source and multiple sources or between under-balcony area and main area.

2:00

4pAA5. Alternative metrics for the directivity of acoustic sources. Timothy W. Leishman (Acoust. Res. Group, Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT 84602)

While the directivity of an acoustic source at a given frequency is thoroughly characterized by a directivity function over the angular coordinates, it may also be characterized to a lesser degree by a single-number directivity factor. The directivity index (i.e., the logarithmic version of the directivity factor) is a related figure of merit. Recent efforts to quantify the directivity of sources for architectural acoustics measurements have led to several alternatives to these values. One example is the area-weighted spatial standard deviation of radiated levels over a free-field measurement sphere. This paper presents and compares this and other directivity metrics for several types of sources, and discusses their benefits.

2:15

4pAA6. Room volume estimation from diffuse field theory. Martin Kuster and Maarten van Walstijn (Sonic Arts Res. Ctr., Queen's Univ. Belfast, BT7 1NN Belfast, Northern Ireland, m.kuster@qub.ac.uk)

Among the parameters relevant in room acoustics, the room volume is one of the most important. The general course in room acoustics research is to use the room volume in the prediction of room acoustic parameters such as reverberation time or total relative sound pressure level. Contrary to this, it has been investigated to what extent the room volume can be retrieved from a measured room impulse response. The approach followed is based on room acoustic diffuse field theory and requires correctly measured room impulse responses with the initial time delay corresponding to the source to receiver distance. A total of ten rooms of varying size and acoustic characteristics have been included. The results in three rooms were unreliable, which was explained by the particular acoustic characteristics. In the remaining rooms the results were numerically useful and consistent between different positions within the same room (relative standard deviation around 20%). The influence of source and receiver directivity is also considered.

2:30

4pAA7. *In situ* measurements for evaluating the scattering surfaces in a concert hall. Jin Yong Jeon and Shin-ichi Sato (School of Architectural Eng., Hanyang Univ., Seoul 133-791, Korea, jyjeon@hanyang.ac.kr)

Sound diffusion by a wall structure is one of the main concerns with respect to the sound quality of concert halls. There is a need to develop measurement and evaluation methods for determining the performance of scattering wall surfaces not only in a laboratory but also in actual halls. In this study, the acoustical measurements were conducted in a concert hall which has diffusers with ceramic cubic tiles on the side walls of the stage and the audience area. Binaural impulse responses were measured at all of the seats under two conditions, that is, with and without diffusers. The area which was affected by the diffusive wall was determined and quantified. The condition without diffusers was produced by covering them with the movable reflectors. From the binaural impulse responses, the temporal diffusion [H. Kuttruff, *Room Acoustics*, (Elsevier Science, London, 1991)], which is calculated from the autocorrelation of the impulse response, and other acoustical parameters were analyzed. From the relationship between the scattering coefficient and the acoustical parameters, sound scattering index for real halls, which represents the degree of the diffusion of a hall, was proposed.

2:45

4pAA8. Further investigations on acoustically coupled spaces using scale-model technique. Zuhre Su, Ning Xiang (Grad. Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY 12180), and Jason E. Summers (Naval Res. Lab., Washington, DC 20024)

Recently, architectural acousticians have been increasingly interested in halls that incorporate coupled-volume systems because of their potential for creating nonexponential sound energy decay. Effects of coupling-aperture configuration and source and receiver locations on energy decay are essential aspects of acoustically coupled spaces that have not yet been extensively investigated. In order to further understand these effects on sound fields in coupled rooms, a systematic experimental study is carried out. An acoustic scale model technique is used in collecting room impulse responses of a two-room coupled system for varying aperture configurations and surface-scattering conditions. Baseline behavior is established by varying aperture area for a fixed aperture shape and analyzing relevant energy-decay parameters at different locations. Effects of aperture shape and number are systematically investigated by varying these parameters while holding coupling area fixed. Similarly, effects of receiver location are systematically investigated by varying the distance of the receiver from the coupling aperture for a fixed aperture configuration. Schroeder decay-function decompositions by Bayesian analysis reveal sensitivities to receiver location and aperture configuration across different frequency bands.

3:00–3:15 **Break**

3:15

4pAA9. Virtual microphone control: A comparison of measured to created impulse responses of various microphone techniques. Daniel Valente and Jonas Braasch (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, danvprod@yahoo.com)

A method of rendering sound sources in 3-D space has been developed using virtual microphone control (ViMiC) [J. Acoust. Soc. Am. **117**, 2391]. This method has been used to create a flexible architecture for the creation and rendering of a virtual auditory environment based on microphone techniques. One of the advantages of ViMiC is the ability to simulate coincident, near-coincident, and spaced microphone recording techniques. This allows the user active spatial control over the recorded environment and the ability to shape the final rendering based on his or her specific auditory needs. In order to determine the accuracy of simulating the virtual microphone techniques, measurements of several acoustic

spaces in Troy, NY will be compared to the measurements of generated impulse responses of the same modeled spaces within the ViMiC environment. The data from the measured impulse responses will be used to adapt the ViMiC system in order to create a more realistic auditory rendering. Moreover, the ViMiC system can be improved for use as an educational tool for teaching recording engineers to hear the subtle differences between various microphone techniques.

3:30

4pAA10. The estimation of the room acoustic characteristic using the acoustic intensity method. Yong Tang, Hideo Shibayama, and Takumi Yosida (Dept. of Commun. Eng., Shibaura Inst. of Technol., 3-7-5 Toyosu Koutou-ku, Tokyo, 135-8548 Japan, m603101@sic.shibaura-it.ac.jp)

When a sound radiates in rooms, a lot of reflection sounds are generated. From estimation of the direction where the room reflection sound comes from, we can understand the diffusion situation in the room acoustic field. By using the acoustic intensity method, we can measure the strength and the direction of the sound. In this paper, we estimate the direction of the reflection sound in the time-space by the acoustic intensity method and show the acoustic characteristic of the room.

3:45

4pAA11. Binaural simulation in an enclosure using the phased beam tracing. Cheol-Ho Jeong and Jeong-Guon Ih (NOVIC, Dept. of Mech. Eng., KAIST, Sci. Town, Daejeon 305-701, Korea, chjeong@kaist.ac.kr)

Binaural simulation in an enclosed space is important in the subjective evaluation of the enclosure acoustics in the design or refinement stage. A time domain scheme using the geometrical acoustics technique has been usually used in the binaural processing. However, one can calculate a pressure impulse response by using the phased beam-tracing method, which incorporates the phase information in the beam tracing process. Such phased method employs reflection coefficient and wave number, whereas the conventional method uses absorption coefficient and air attenuation factor. Impulse response can be obtained by the inverse Fourier transformation of the frequency domain result. This feature facilitates the binaural simulation because the convolution with the HRTF can be accomplished by a simple multiplication in frequency domain. Convolutions were conducted for all reflections one by one, and the convolved transfer functions were summed into one transfer function. Consequently binaural room impulse responses at receivers' ear positions can be simulated. The measured binaural room impulse responses in the conference room were compared with the predicted results for octave bands of 125 Hz to 4 kHz. A good agreement with measurement was found, especially in the early part of impulse responses. [Work supported by BK21.]

4:00

4pAA12. Visualization methods of direct and early reflection sounds in small enclosures. Chiaki Koga, Akira Omoto (Omoto Lab., Dept. of Acoust. Design, Faculty of Design, Kyushu Univ., Shiobaru 4-9-1, Minami, Fukuoka 815-8540, Japan), Atsuro Ikeda, Masataka Nakahara (SONA Corp., Nakno-ku, Tokyo, 164-0013, Japan), Natsu Tanaka, and Hiroshi Nakagawa (Nittobo Acoust. Eng. Co., Ltd. Sumida-ku, Tokyo 130-0021, Japan)

Many parameters exist for evaluating large sound fields such as concert halls. However, it is difficult to apply those parameters for evaluation of a small room such as a recording studio because of their different sound fields. Widely useful common parameters have not been established. Moreover, early reflections are important in small rooms for determining spatial acoustic impressions. Therefore, various methods that visualize spatial acoustic information obtained by early reflection in rooms are proposed. For this study, sound fields (a music studio and a filmmaking studio) were measured using three kinds of different techniques: instantaneous intensity, mean intensity, and a sphere-baffled microphone array. This report compares the information of sound source directions obtained using these methods. Results show that every method can estimate the position of sound sources and important reflections with high accuracy. In the future, we shall propose a method that visualizes spatial acoustic information more precisely by combining the methods and establishing acoustic parameters that are available for evaluating and designing small rooms.

4:15

4pAA13. Acoustic evaluation of worship spaces in the city of Curitiba, Brazil. Cristiane Pulsides, David Q. de Sant'Ana, Samuel Ansay (LAAICA/UFPR, Bloco 4 sala PG-05 81531-990 Curitiba, PR, Brasil, pulsides@gmail.com), Paulo Henrique T. Zannin, and Suzana Damico (LAAICA/UFPR, 81531-990 Curitiba, PR, Brasil)

This article searches acoustic parameters in religious buildings located in the city of Curitiba intending to study its behavior in this kind of facilities. The temples were analyzed according to type of ceremony, architectonic style, and construction date. The research was made through the impulsive response integration method for three energetic parameters: (1) reverberation time (RT); (2) clarity (C80); and (3) definition (D50) according recommendations of the ISO/3382:1997 Standard. Performed in between were six and eight impulsive responses in each room using sweep signals and omnidirectional microphones. The results were than compared with referential values already existing [W. Fasold and E. Veres, *Schallschutz + Raumakustik in der Praxis*, 136 (1998)] for acoustic characterizations. It is possible to observe in the measurements the direct connection between reverberation time and the parameters clarity or definition. Moreover, it is possible also to observe the influence of the geometric ratios and architectural elements of the rooms, getting itself for equivalent volumes and rays of removal of the source, different levels of definition.

4:30

4pAA14. A consideration of the measurement time interval for obtaining a reliable equivalent level of noise from expressway. Mitsunobu Maruyama (Salesian Polytechnic, Oyamagaoka 4-6-8, Machida, Tokyo 194-0215, Japan) and Toshio Sone (Akita Prepectural Univ., Honjo, Akita 015-0055, Japan)

The level of road traffic noise $LA_{eq,T}$ greatly depends on the maximum level during the measurement time interval tau, and the maximum level often appears at the moment when two consecutive heavy vehicles pass through the point adjacent to the observation point. A mathematical model is proposed for simulating the variation in traffic noise, especially from the point of heavy vehicles with passing. The mean time interval between a pair of two consecutive heavy vehicles with the minimum allowable distance is obtained from time-series data and the mean recurrence time h_{ij} which can be calculated from the transition matrix $P[p_{ij}]$. The comparative study is made among the numbers of heavy vehicles from 25 to 300 [vehicles/hour] in traffic flow and the observation distances of 40 to 200 m from the road. The result shows that the measurement time interval required for the acquisition of reliable data is three to four times as long as tau or h_{ij}.

Session 4pABa

Animal Bioacoustics: Marine Mammal Acoustics II

David K. Mellinger, Chair
Oregon State Univ., Hatfield Marine Science Ctr., Newport, OR 97365

Contributed Papers

1:15

4pABa1. Great ears: Functional comparisons of land and marine leviathan ears. D. R. Ketten (Harvard Med. School, Boston, MA; Woods Hole Oceanograph. Inst., Woods Hole, MA), J. Arruda, S. Cramer, M. Yamato (Woods Hole Oceanograph. Inst., Woods Hole, MA), J. O'Malley (Massachusetts Eye and Ear Infirmary, Boston, MA), D. Manoussaki (Vanderbilt Univ., Nashville, TN), E. K. Dimitriadis (NIH/NIDCD, Bethesda, MD), J. Shoshani (Univ. of Asmara, Asmara, Eritrea), and J. Meng (American Museum of Natural History, New York, NY)

Elephants and baleen whales are massive creatures that respond to exceptionally low frequency signals. Although we have many elephant and whale vocalization recordings, little is known about their hearing. Playback experiments suggest hearing in both proboscideans and mysticetes is tuned similarly to low or even infrasonic signals. This raises several interesting issues. First, they emit and perceive signals in two media, air and water, with radically different physical acoustic properties: 4.5-fold differences in sound speed, three-fold magnitude difference in acoustic impedance, and, for common percepts, whales must accommodate 60-fold acoustic pressures. Also, a commonly held tenet is that upper hearing limit is inversely correlated with body mass, implying there should be virtually no whale-elephant hearing overlap given body mass differences. This study analyzed how inner ears in these groups are structured and specialized for low-frequency hearing. Computerized tomography and celloidin histology sections were analyzed in six baleen whale ($n=15$) and two elephant species ($n=7$). The data show mysticetes have a substantially greater hearing range than elephants but that coiling and apical cochlear structures are similar, suggesting common mechanical underpinnings for LF hearing, including cochlear radii consistent with the Whispering Gallery propagation effect. [Work supported by ONR, NIH, WHOI OLI, Seaver Foundation.]

1:30

4pABa2. Social context of the behavior and vocalizations of the gray whale *Eschrichtius robustus*. Sarah M. Rohrkasse (School for Field Studies, Ctr. for Coastal Studies, Apartado Postal 15, Puerto San Carlos, BCS, CP 23740 Mexico, sarro101@hotmail.com) and Margaret M. Meserve (Guilford College, Greensboro, NC 27410)

Sound production and surface behavior of the gray whale were investigated at Bahia Magdalena, Mexico to determine if vocalizations have behavioral correlations or are used in specific social contexts. Fifteen-minute sessions of behavioral observations and acoustic recordings of gray whales in various social contexts were collected from February to April 2006 ($n=30$). Analysis of sound production included proportional use of different call types and acoustic variables of each sound type. Preliminary acoustic analysis found no correlation with social contexts or behaviors, but proportional use of different vocalizations is similar to past studies in Baja [Dahlheim *et al*, *The Gray Whale*, pp. 511–541 (1984), F. J. Ollervides, dissertation, Texas A&M University (2001)]. Initial results indicate significant differences in frequencies of high surface behaviors ($p=0.0477$) of groups that include mother-calf pairs. As analysis continues, possible correlations between social context and use of sounds could allow for acoustics to be an indicator of group composition, seasonal movements, and social patterns and to help determine the functions of sounds. [Work supported by SFS and NFWF.]

1:45

4pABa3. Ambient noise and gray whale *Eschrichtius robustus* behavior. Francisco Ollervides, Kristin Kuester, Hannah Plekon, Sarah Rohrkasse (School for Field Studies—Ctr. for Coastal Studies, Apartado Postal 15, Puerto San Carlos, BCS, CP 23740 Mexico, follervides@hotmail.com),Kristin Kuester (Univ. of Wisconsin−Madison, Madison, WI 53706), HannahPlekon (Davidson College, Davidson, NC), andSarahRohrkasse (Texas A and M Univ., College Station, TX 77843)

Between 14 February and 13, April 2006, we conducted 31 recording sessions of ambient noise and behavioral sampling of gray whales within Magdalena Bay, Mexico. This breeding lagoon does not have the same Marine Protected Area status compared to the other breeding lagoons of San Ignacio and Guerrero Negro in the Mexican Pacific coast. Poorly monitored guidelines and increasing boat traffic from whale−watching tourism in this area have the potential to affect the surface behavior of these animals and increase average ambient noise levels. Relative ambient noise levelswererecordedandcomparedtoapreviousstudy[Ollervides,2001]todetermine similarities or differences in the 5−year interval between both data sets. Although results are not comparable in decibel levels, probably due to equipment calibration problems, there was a significant difference between the different regions of the bay Kruskal−Wallis ($p=0.0067$). Activity levels ranged from 0.005–0.196 behaviors/whale/minute. Ambient noise levels ranged from 35.70–64.32 dB Re: 1 Pa. No correlation was found between the ambient noise levels in the bay and the activity level of gray whales (correlation value$=0.0126$; log correlation value$=0.172$). Further acoustic processing is currently underway.

2:00

4pABa4. Look who's talking; social communication in migrating humpback whales. Rebecca A. Dunlop, Michael J. Noad (School of Veterinary Sci., Univ. of Queensland, St. Lucia, Qld 4072, Australia. r.dunlop@uq.edu.au), Douglas H. Cato (Defence Sci. and Tech Org., Pyrmont, NSW 2009, Australia), and Dale Stokes (Scripps Inst. of Oceanogr., La Jolla, CA 92037)

A neglected area of humpback acoustics concerns nonsong vocalizations and surface behaviors known collectively as social sounds. This study describes a portion of the nonsong vocal repertoire and explores the social relevance of individual sound types. A total of 622 different sounds were catalogued and measured from whales migrating along the east coast of Australia. Aural and spectral categorization found 35 different sound types, and discriminate functions supported 33 of these. Vocalizations were analyzed from 60 pods that were tracked visually from land and acoustically using a static hydrophone array. Nonsong vocalizations occurred in all pod compositions: lone whales, adult pairs, mother/calf pairs, mother/calf/escorts, and multiple-adult pods. Thwops and wops were likely to be sex-differentiated calls with wops from females and thwops from males. Sounds similar to song-units were almost all from joining pods and yaps were only heard in splitting pods. Other low-frequency calls (less than 60 Hz) were thought to be within-pod contact calls. Higher-frequency cries (fundamental 450–700 Hz) and other calls (above 700 Hz) and presumed underwater blows were heard more frequently in joining

4p FRI. PM

pods displaying agonistic behaviors. This work demonstrates that humpbacks produce a great range of contextually different communication signals. [Work supported by ONR and DSTO.]

2:15

4pABa5. Seasonal ambient noise levels and impacts on communication in the North Atlantic right whale. Susan E. Parks, Christopher W. Clark, Kathryn A. Cortopassi, and Dimitri Ponirakis (Bioacoustics Res. Program, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, sep6@cornell.edu)

The North Atlantic right whale is a highly endangered species of baleen whale. Acoustic communication plays an important role in the social behavior of these whales. Right whales are found in coastal waters along the east coast of the United States, an area characterized by high levels of human activity. Most of these activities generate noise that is propagated into the coastal marine environment. The goals of this project are to characterize the noise, both natural and anthropogenic, in right whale habitat areas to determine what levels of noise the whales are regularly exposed to, and whether the acoustic behavior of right whales changes in response to increased noise. Continuous recordings were made from autonomous bottom-mounted recorders in three major habitat areas in 2004 and 2005; Cape Cod Bay (December–May), Great South Channel (May), and the Bay of Fundy, Canada (August) to passively detect right whales by recording their vocalizations. Here, we describe the ambient noise levels in these recordings to describe the daily acoustic environment of right whales, how noise varied over diel, weekly, and seasonal time scales, and whether noise levels correlated with any observed changes in acoustic behavior of the whales.

2:30

4pABa6. Blue whale calling in Australian waters. Robert D. McCauley, Chandra P. Salgado Kent (Curtin Univ. of Technol., G.P.O. Box U 1987, Perth 6845, Australia), Christopher L.K. Burton (Western Whale Res. Hillarys 6923, WA Australia), and Curt Jenner (Ctr. for Whale Res. (WA Inc.), Fremantle WA, 6959 Australia)

Calling from the Antarctic true blue whale (*Balaenoptera musculus intermedia*) and the tropical subspecies (brevicauda, or pygmy blue) have been recorded across southern Australia with the pygmy blue calls also recorded along the Western Australian (WA) coast. The subspecies have a believed common downsweep and markedly different longer, tonal calls. The frequency of most energy in the tonal calls is offset between the subspecies suggesting sound-space partitioning. The pygmy blue three-part tonal call is typically 120 s long repeated every 200 s, has several variants, and includes a complex two-source component. The nature of the pygmy blue call allows counts of instantaneous calling individuals, giving relative abundance. These estimates in the Perth Canyon, a localized seasonal feeding area, show patterns in usage of space and through time within and between seasons, such as the sudden departure of animals at a season end, which varies by approximately 2 weeks between years. Sea noise records along the WA coast indicate south-traveling animals arrive midway along the coast in October to November, animals fan out across southern Australian over December through May, then move north in the Austral winter. We have begun converting abundance estimates from relative to absolute for pygmy blue calling rates.

2:45

4pABa7. Acoustical monitoring of finback whale movements on the New Jersey Shelf. Altan Turgut (Naval Res. Lab., Acoust. Div., Washington, DC 20375) and Christopher Lefler (Univ. of California Santa Barbara, Santa Barbara, CA 93106)

Acoustical monitoring of finback whales is performed by using a data set collected over a 3-week period in December of 2003 on the New Jersey Shelf. One-second-duration 20-Hz signals of finback whales were recorded on three vertical line arrays (VLAs) and a bottomed horizontal line array (HLA.). One-second-duration pulses are separated by about 10 s and there is an approximately 2-min-long silent period between 10- to 18-min-long pulse trains. A 30- to 60-min silent period after 5 to 10 pulse trains is also common. Modal analysis of individual pulses indicated that most signals contained two acoustic modes. Arrival-time and group-speed differences of these modes are used for remote acoustic ranging. These modal characteristics are also exploited in a broadband matched-field algorithm for depth discrimination. Bearing estimation of individual whales is obtained by performing horizontal beamforming on the HLA data. Range estimation results are verified by time-of-flight triangulation using single hydrophone data from each VLA location. Acoustic monitoring results indicated that most finback whales traveled near the shelf break front where food might be abundant. Relations between silent periods and acoustic range/depth monitoring results are also investigated. [This work was supported by the ONR.]

3:00–3:15 Break

3:15

4pABa8. Analysis of melon-headed whale aggregation in Hanalei Bay, July 2004. David M. Fromm (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5350), Joseph R. Mobley, Jr. (Univ. of Hawaii at M_noa, Honolulu, HI 96822), Stephen W. Martin (Space and Naval Warfare Systems Ctr. San Diego, San Diego, CA 92152-5001), and Paul E. Nachtigall (Univ. of Hawaii at M_noa, Kailua, HI 96734)

On 3 July 2004, an aggregation of ca. 150–200 melon-headed whales (*Peponocephala electra*) appeared in the shallow waters of Hanalei Bay, Kauai and congregated there for over 27 h. Preceding the whales' appearance and partially coincident with their time in the Bay, midrange (3.5–5 kHz) tactical sonars were intermittently deployed during the Rim of the Pacific 2004 (RIMPAC) joint military exercises being conducted in waters near Kauai by the U.S., Japan, and Australia Navies. An NOAA report (Southall *et al.*, 2006) attributed the active sonar usage as a plausible, if not likely, contributing factor. A detailed timeline and reconstruction of the RIMPAC activities is presented showing the worst-case estimates of the sonar sound levels in the waters surrounding Kauai. A re-examination of available evidence combined with a new report of a simultaneous and similar aggregation in Sasanhaya Bay, Rota, Commonwealth of the Northern Mariana Islands, brings the plausibility conclusion into question. [This work was sponsored by multiple sources. D. Fromm and S. Martin conducted acoustic analyses with funds provided by the U.S. Pacific Fleet. J. Mobley received funding from the U.S. Geological Survey. P. Nachtigall is sponsored by the Office of Naval Research for marine mammal audiometric studies.]

3:30

4pABa9. Midfrequency sound propagation in beaked whale environments. Eryn M. Wezensky, Thomas R. Stottlemyer, Glenn H. Mitchell (Naval Undersea Warfare Ctr., Newport Div., Newport, RI 02841), and Colin D. MacLeod (Univ. of Aberdeen, Aberdeen, U.K.)

Recent mass strandings of beaked whales (*Ziphiidae, Cetacea*) coinciding with the use of midfrequency range (1–10 kHz) active sonar have caused speculation about the potentially adverse effects of these sound sources. Particular questions of the research and regulatory communities concern whether beaked whale sensitivity to midfrequency sound exposure is influenced by oceanographic characteristics present at the time of the mass stranding events. This study investigated the interaction between beaked whale habitat characteristics and the nature of a midfrequency signal by analyzing the oceanographic factors affecting underwater acoustic propagation. Three types of model sites were selected from five specific geographical locations where beaked whales have been regularly recorded or where a mass stranding event has been reported. A ray-trace acoustic propagation model was used to generate transmission loss for a 3-kHz signal over a representative 60-km transect at each locality. Model outputs visually demonstrated how the combination of site/event-specific oceanographic characteristics affects the sound propagation of a moving source. A parametric sensitivity comparison and statistical analysis were conducted to identify influential factors between environmental parameters, source depth, and the resulting transmission loss. Major findings of this study as well as future research direction are discussed. [Research supported by NAVSEA.]

3:45

4pABa10. Examination and evaluation of the effects of fast rise-time signals on aquatic animals. Michael Stocker (Seaflow, Inc., 1062 Ft. Cronkhite, Sausalito, CA 94965)

Increasingly human enterprise is subjecting the ocean environment to acoustic signals to which marine animals are not biologically adapted. This is evidenced by a marked rise in marine mammal strandings, as well as hearing and other physiological damage to fish and other marine organisms as a result of, or coincident to, human-generated noise events. Determining phonotoxic thresholds of marine organisms is complicated by the fact that various marine animals are adapted to sense either pressure gradient or particle motion acoustic energy, or some combination or gradient between the two. This has been addressed to some degree by exposure metrics that consider either net or accumulated acoustical flux densities from various noise sources. This paper examines the role and effects of signal rise time both in terms of physiological impulse response of the exposed organisms, as well as broadband saturation flux densities of fast rise-time signals on animal sense organs. Case studies from the literature will be presented to demonstrate the effects of fast rise time signals on fish. Acoustical signals with high crest factors and fast rise-time components will be compared to signals with dominantly sinusoidal components to illustrate the perceptual effects of these signals on human hearing.

4:00

4pABa11. Noise separation of underwater acoustic vocalization using auditory filter bank and Poisson rate estimation. Owen P. Kenny and Craig R. McPherson (Dept. of Elec. and Comput. Eng., James Cook Univ., Douglas 4811, Queensland, Australia)

Formant vocalization tracking has been achieved using a mammalian periphery model and a Poisson rate estimator. This approach used a set of linear bandpass filters to simulate the mechanical displacement of the basilar membrane. The auditory model simulated neural firing by producing a spike at the positive going zero crossing for each filter output. Poisson intensity of the neural firing rate is controlled by the dominant frequency components of the signal present in the filter. This approach is extended by incorporating neural synchronization information to separate the formant structure from that of noise. The filter structure is designed to overlap the frequency range of adjacent filters. The presence of a formant structure in adjacent filters controls the interspike intervals of neural firing for both filters, which results in the neural firing from both filters being synchronized. If a noise-only component is present in either filter, then the spiking outputs from the adjacent filters are unsynchronized. Experimental results have shown that incorporating neural synchronization information between adjacent filters has enabled separation of signal components from noise. This technique enables easier signal and noise separation than allowed by traditional methods.

4:15

4pABa12. Using vocalizations of Antarctic seals to determine pupping habitats. T. L. Rogers, C. J. Hogg, M. B. Ciaglia (Australian Marine Mammal Res. Ctr., Zoological Parks Board of NSW/Faculty of Veterinary Sci., Univ. of Sydney, Mosman Australia), and D. H. Cato (Defence Sci. & Technol. Organisation, Pyrmont, Australia)

The Ross and Leopard seal use the floes of the Antarctic pack ice to whelp and raise their pups, But both species are rarely seen in summer throughout the pack ice. We now realize that this is because they are under the water "calling" during the austral summer as part of their breeding display, and so their presence is underestimated in traditional visual surveys. The period of "calling" overlaps with the time that females give birth, so their vocalizations can be used to determine seal distributions during this time. Acoustic recordings were made using sonobuoys deployed during ship based surveys in the pack ice and analyzed to determine the seal distributions. This was used to predict habitat preference of seals by relating their distributions to remotely sensed indices: ice cover, ice floe type, ice thickness, distance to ice edge, distance to shelf break, distance to land, sea surface temperature, and chlorophyll a.

FRIDAY AFTERNOON, 1 DECEMBER 2006 KOHALA/KONA ROOM, 4:30 TO 5:15 P.M.

Session 4pABb

Animal Bioacoustics: Avian Acoustics

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

Contributed Papers

4:30

4pABb1. Effective area of acoustic lure surveys for Mexican spotted owls (*Strix occidentalis lucida*). Samuel L. Denes, Ann E. Bowles (Hubbs-SeaWorld Res. Inst., 2595 Ingraham St., San Diego, CA 92109, sdenes@hswri.org), Kenneth Plotkin, Chris Hobbs (Wyle Labs., Arlington, VA 22202), John Kern (Kern Statistical Services, Sauk Rapids, MN 56379), and Elizabeth Pruitt (GeoMarine, Inc., Hampton, VA 23666)

During acoustic lure surveys for birds, topography and ambient noise are likely to be important determinants of detectability. Examinations of propagation were conducted for acoustic lures (human-made calls) and owl responses recorded during acoustic surveys for Mexican spotted owls in the Gila National Forest (2005). Lure surveys were designed based on formal agency protocols, which assumed a 0.43-km detection range under typical conditions. A total of 558 points was called over a heavily forested, topographically complex 20×24-km area. Real-time measurements of owl calls and lures were made with a calibrated recording system. Ambient noise was collected using an array of 39 Larson-Davis 820 and 824 sound-level meters. The NMSIM (Wyle Laboratories) single-event propagation simulator was used to model propagation of both owl and human calls. The resulting model of survey effort was compared with a simple two-dimensional statistical model. Probability of detecting owls did not fit the expectations of the agency protocol, suggesting that acoustic propagation should be considered during owl surveys. [Work supported by U.S. Air Force ACC/CEVP; USFWS Permit No. TE024429]

4:45

4pABb2. Automated localization of antbirds and their interactions in a Mexican rainforest. Alexander N. G. Kirschel, Travis C. Collier, Kung Yao, and Charles E. Taylor (Univ. of California, Los Angeles, 621 Charles E. Young Dr. South, Los Angeles, CA 90095)

Tropical rainforests contain diverse avian communities incorporating species that compete vocally to propagate their signals to intended receivers. In order to effectively communicate with birds of the same species, birds need to organize their song performance temporally and spatially. An automated identification and localization system can provide information on the spatial and temporal arrangement of songs. Acoustic sensor arrays were tested for the ability to localize the source of songs of antbirds recorded in a Mexican rainforest. Pilot studies with a five-node array arranged in a rough circle with a 20-m diameter located the song of Dusky Antbird (*Cercomacra tyrannina*) with an error of 73 cm and Mexican Antthrush (*Formicarius moniliger*) with an error of 65 cm from the location of a source loudspeaker within the array. An additional source 21 m outside was also localized. Results will be presented for experiments and recordings of individuals at the Mexican rainforest site in October 2006. Locations of birds of the same and different species during vocal performance will provide a greater understanding of how individuals interact spatially with each other based on their vocal performance, from which the role of song in ecological interactions can be inferred.

5:00

4pABb3. Nonintrusive acoustic identification of hermit thrush (*Catharus guttatus*) individuals. Dennis F. Jones (Defence R&D Canada—Atlantic, P.O. Box 1012, Dartmouth, NS, Canada B2Y 3Z7, dennis.jones@drdc-rddc.gc.ca)

From mid-April well into the summer, the secretive hermit thrush (*Catharus guttatus*) can be heard singing throughout the woodlands of Nova Scotia. Its song is distinctive, beginning with a clear introductory note followed by a flurry of flutelike body notes, often cascading and reverberant in character. Despite this fine display of avian virtuosity, few studies have been reported that probe the differences between the calls, songs, and repertoires of individuals. From April 2003 to May 2006, over 3000 songs from several birds were recorded using digital video cameras at study sites in and around the city of Halifax, Nova Scotia. The only birds recorded were those in close proximity to roads and trails. None of the birds were marked, banded, or deliberately disturbed in any way. Although the study birds remained hidden from view most of the time, in the few instances where the birds perched in the open, their behaviors while singing were captured on videotape. All of the birds were readily distinguishable from each other as no two individuals had a single song in common. The most significant finding was that individuals could be reidentified acoustically after 1 week, 3 months, and 1 year had elapsed.

FRIDAY AFTERNOON, 1 DECEMBER 2006 KAHUKU ROOM, 1:00 TO 4:50 P.M.

Session 4pBB

Biomedical Ultrasound/Bioresponse to Vibration and Signal Processing in Acoustics: Elastic Imaging

Peter J. Kaczkowski, Cochair
Univ. of Washington, Applied Physics Lab., 1013 NE 40th Street, Seattle, WA 98105-6698

Tsuyoshi Shiina, Cochair
Univ. of Tsukuba, Graduate School of Systems and Information Engineering, 1-1-1 Tennodai, Tsukuba 305-8573, Japan

Invited Papers

1:00

4pBB1. Present and future of elasticity imaging technology. Tsuyoshi Shiina (Grad. School of Systems and Information Eng., Univ. of Tsukuba, 1-1-1 Tennodai Tsukuba, Japan) and Ei Ueno (Univ. of Tsukuba, Tsukuba, Japan)

Elastic properties of tissues are expected to provide us novel diagnostic information since they are based on tissue characteristics and sensitively reflect its pathological state. So far, various techniques for tissue elasticity imaging have been proposed. However, it was not so easy to satisfy real-time operation and freehand manipulation of probe, which was required for practical equipment. To satisfy these conditions, we developed the combined autocorrelation method (CAM) and recently manufactured a commercial ultrasound scanner, for real-time tissue elasticity imaging by implementing the CAM algorithm, By slightly compressing or relaxing the body through freehand operation, the strain images are obtained with real-time and superimposed on B-mode images with a translucent color scale. In addition, we proposed elasticity scores of malignancy by categorizing patterns of elasticity images of breast tumors into five classes from malignant to benign. As a result of diagnosis based on the elasticity score, it was revealed that even nonexperts could attain precise diagnosis of breast cancer based on elasticity score as well as experts since the criterion on elasticity score is much simpler than conventional B-mode images. Finally, some prospects for the next stages of elasticity imaging technology well be surveyed.

1:20

4pBB2. Real-time tissue elasticity system—Development and clinical application. Takeshi Matsumura, Tsuyoshi Mitake (Hitachi Medical Corp. 2-1, Toyofuta, Hashiwa-Shi, Chiba-Ken, Japan), Tsuyishi Tsuyishi, Makoto Yamakawa, Ei Ueno (Tsukuba Univ.), Nobuhiro Fukunari (Shouwa Univ.), and Kumi Tanaka (Nippon Medical Univ.)

The progress of recent semiconductor technology has a remarkable thing. Thanks to progress of this semiconductor technology, the ultrasound scanner in medicine could come to hold enormousness computing power and has come to realize various complicated processing. At the same time, hardness of human tissue which, as you know, is used by palpation is already the information that is important in a diagnosis. But, we think that it does not have enough objectivity. To increase objectivity by visualizing hardness of tissue, we adopted ECAM (extended combined autocorrelation method), which was developed by Professor Shiina at Tsukuba University in Japan, and succeeded in developing the commercial ultrasound scanner, which could display a strain image in real time. From a clinical point of view, in the breast region, mammography examination is effective in a diagnosis, but a judgment of permeation degree is not superior in ultrasound image. And in a thyroid gland region, we begin to get experience with availability from a diagnosis of papillary cancer and follicular cancer. So, we would like to have the presentation about the development of a strain imaging function and some of our clinical experiences by using the developed system.

1:40

4pBB3. Elasticity of perfused tissue. Kirk W. Beach (Dept. of Surgery, Univ. of Washington, Box 356410, Seattle, WA 98195-6410), Barbrina Dunmire, and John C. Kucewicz (Univ. of Washington, Seattle, WA 98105-6698)

Elastic imaging intends to measure Young's modulus (tissue stiffness) or bulk modulus (tissue compressibility) of tissue subjected to an applied strain of several percent. Underlying elastic imaging is the assumption of a linear stress/strain relationship without hysteresis or other time-dependent behavior. Perfused tissue is a composite material comprised of a solid matrix of cells, fibers, interstitial fluid (occupying up to 50% of the tissue volume and varying slowly with time), arterioles (pulsating high-pressure spaces that occupy 0.1% of the tissue volume), capillaries, and venules (low-pressure spaces that occupy up to 3% of the tissue volume varying with respiration). This talk will speculate on the nonlinear, nonstationary stress/strain relationships expected from dependent tissues (legs), pressurized tissues (breast tumors), and other living, perfused tissues. The pressure versus strain curve from each tissue voxel allows the measurement of arteriolar and venular volumes and pressures, and interstitial pressure within the tissues. These volumes and pressures may be key to classifying pathologies.

2:00

4pBB4. New developments in transient elastography. Mathias Fink, Mickael Tanter, Ralph Sinkus, and Gabriel Montaldo (LOA, ESPCI, 10 rue Vauquelin, 75005, Paris, France)

An ultra-high-rate ultrasonic scanner has been developed that can give 5000 ultrasonic images per second of the body. With such a high frame rate, the propagation of transient shear waves can be followed, and from the spatio-temporal evolution of the displacement fields, various inversion algorithms allow us to recover the shear modulus map. A discussion on the various inversion algorithms will be presented. In order to obtain unbiased shear elasticity map, different configurations of shear sources induced by radiation pressure of focused transducer arrays are used. Both 2-D and 3-D imaging can obtained with this technique. *In vitro* and *in vivo* results on breast will be presented that demonstrate the interest of elasticity imaging with transient elastography.

Contributed Papers

2:20

4pBB5. Spectral characteristics of breast vibro-acoustography images. Azra Alizad, Dana H. Whaley, Mathew Urban, Randall R. Kinnick, James F. Greenleaf, and Mostafa Fatemi (Mayo Clinic College of Medicine, Rochester, MN 55905 aza@mayo.edu)

Vibro-acoustography image is a function of the dynamic characteristics of the object at the vibration (difference) frequency (df). The dynamic characteristic of tissue is closely related to pathology. Therefore, it is important to evaluate image features versus df. Here, the influence of df on breast vibro-acoustography images is studied by scanning human breast at various df values ranging from 20 to 90 kHz. The subjects were chosen from a group of volunteers with different breast abnormalities. Images were compared subjectively to study image features and the appearances of breast lesions versus df. It is demonstrated that having a collection of images of the same tissue at different df values generally provides a better perception of the tissue structure and improves lesion identification. In most cases, higher df resulted in a higher signal-to-noise ratio and thus a higher image quality. Finally, a frequency-compounded images was obtained by calculating the weighted sum of images at different df values. It is demonstrated that image compounding normally improves visualization of breast tissue and abnormalities. [Work supported by NIH Grant EB-00535 and Grant BCTR0504550 from the Susan G. Komen Breast Cancer Foundation. Disclosure: Parts of the techniques used here are patented by MF and JFG.]

2:35

4pBB6. Tissue pulsatility imaging: Ultrasonic measurement of strain due to perfusion. John C. Kucewicz, Barbrina Dunmire, Lingyun Huang, Marla Paun (Univ. of Washington Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105-6698), and Kirk W. Beach (Univ. of Washington, Seattle, WA 98195-6410)

Over each cardiac cycle perfused tissues expand and relax by a fraction of a percent as blood rapidly accumulates in the arterial vasculature during systole and then slowly drains through the venous vasculature during diastole. Tissue pulsatility imaging (TPI) is a variation on ultrasonic tissue strain imaging that estimates tissue perfusion from this natural, cyclic tissue expansion and relaxation. TPI is derived in principle from plethysmography, a century-old technology for measuring gross tissue volume change from a whole limb or other isolatable body part. With TPI, the plethysmographic signal is measured from hundreds or thousands of sample volumes within an ultrasound image plane to characterize the local perfusion throughout a body part. TPI measures tissue strain over the cardiac cycle and parametrizes the signal in terms of its amplitude and shape. The amplitude of the strain waveform is correlated with perfusion, and the shape of the waveform is correlated with vascular resistance. Results will be presented from the leg showing the change in the TPI signals as the muscles recover from exercise, from breast tumors, and from the brain as blood flow changes in response to visual stimulation. [Work supported in part by NIH 1-R01EB002198-01 and NIH N01-CO-07118.]

2:50

4pBB7. Using human body shear wave noise for passive elastography. Karim G. Sabra, Stephane Conti, Philippe Roux, and William A. Kuperman (Scripps Inst. of Ocean., Univ. of California—San Diego, 9500 Gilman Dr., San Diego, CA 92093-0238)

An elastography imaging technique based on passive measurement of shear wave ambient noise generated in the human body (e.g., due to the heart, muscles twitches, and blood flow system) has been developed. This technique merges two recent research developments in medical imaging and physics: (1) recent work on the efficacy of elastographic imaging demonstrating that shear waves are excellent candidates to image tissue elasticity in the human body and (2) theory and experimental verification in ultrasonics, underwater acoustics, and seismology of the concept of extracting coherent Green's function from random noise cross correlations. These results provide a means for coherent passive imaging using only the human body noise field, without the use of external active sources. Coherent arrivals of the cross correlations of recordings of human body noise in the frequency band 2–50 Hz using skin-mounted accelerometers allows us to estimate the local shear velocity of the tissues. The coherent arrivals emerge from a correlation process that accumulates contributions over time from noise sources whose propagation paths pass through both sensors. The application of this passive elastography technique for constructing biomechanical models of *in vivo* muscles' properties will be discussed.

3:05–3:20 Break

3:20

4pBB8. Dynamic radiation force of acoustic waves on solid elastic spheres. Glauber T. Silva (Instituto de Computação, Universidade Federal de Algoas, Maceió, AL, 57072-970, Brazil)

The present study concerns the dynamic radiation force on solid elastic spheres exerted by a plane wave with two frequencies (bichromatic wave) considering the nonlinearity of the fluid. Our approach is based on solving the wave scattering for the sphere in the quasilinear approximation within the preshock wave range. The dynamic radiation force is then obtained by integrating the component of the momentum flux tensor at the difference of the primary frequencies over the boundary of the sphere. Effects of the fluid nonlinearity play a major role in dynamic radiation force, leading it to a regime of parametric amplification. The developed theory is used to calculate the dynamic radiation force on three different solid spheres (aluminum, silver, and tungsten). The obtained spectrum of dynamic radiation force presents resonances with larger amplitude and better shape than those exhibited in static radiation force. Applications of the results to some elasticity imaging techniques based on dynamic radiation force will be presented.

3:35

4pBB9. Ultrasonic measurement of displacement distribution inside an object caused by dual acoustic radiation force for evaluation of muscular relax property due to acupuncture therapy. Yoshitaka Odagiri, Hideyuki Hasegawa, and Hiroshi Kanai (Grad. School of Eng., Tohoku Univ., Sendai 980-8579, Japan, odagiri@us.ecei.tohoku.ac.jp)

Many studies have been carried out on the measurement of mechanical properties of tissues by applying an ultrasound-induced acoustic radiation force. To assess mechanical properties, strain of an object must be generated. However, one radiation force is not sufficient because it also causes translational motion when the object is much harder than surrounding medium. In this study, two cyclic radiation forces are applied to a muscle phantom from two opposite horizontal directions so that the object is cyclically compressed in the horizontal direction. As a result, the object is vertically expanded due to the incompressibility. The resultant vertical displacement is measured using ultrasound. Two concave ultrasonic transducers for actuation were both driven by sums of two continuous sinusoidal signals at two slightly different frequencies of 1 MHz and (1M +5) Hz. Displacement, which fluctuates at 5 Hz, was measured by the ultrasonic *phased tracking method* proposed by our group. Results indicated that the surface of the phantom was cyclically actuated with an amplitude of a tenth of a few micrometers, which well coincided with that measured with laser vibrometer. In addition, upward and downward displacements at the surface and deeper region were found during the increase phase of radiation forces. Such displacements correspond to the horizontal compression.

3:50

4pBB10. A phantom study on ultrasonic measurement of arterial wall strain combined with tracking of translational motion. Hideyuki Hasegawa and Hiroshi Kanai (Grad. School of Eng., Tohoku Univ., Aramaki-aza-Aoba 6-6-05, Sendai 980-8579, Japan, hasegawa@us.ecei.tohoku.ac.jp)

Correlation-based techniques are often applied to ultrasonic rf echoes to obtain the arterial wall deformation (strain). In such methods, the displacement estimates are biased due to changes in center frequency of echoes. One of the reasons for the change in center frequency is the interference of echoes from scatterers within the wall. In the phased tracking method previously proposed for strain estimation by our group, the estimated displacement contains both the components due to the translational motion and strain. The translational motion is larger than strain by a factor of 10 and, thus, the error in the estimated displacement due to the change in center frequency mainly depends on translational motion and is often larger than the minute displacement due to strain. To reduce this error, in this study, a method is proposed in which the translational motion is compensated using the displacement of the luminal boundary estimated by the phased tracking method before correlating echoes between the frame before deformation and that at the maximum deformation to estimate the strain distribution within the wall. In basic experiments using phantoms made of silicone rubber, the estimation error was much reduced to 15.6% in comparison with 36.4% obtained by the previous method.

4:05

4pBB11. Wave biomechanics of skeletal muscle. Oleg Rudenko (Dept. of. Blekinge Inst. of Technol., 371 79 Karlskrona, Sweden) and Armen Sarvazyan (Artann Labs., Inc., West Trenton, NJ 08618)

Physiological functions of skeletal muscle, such as voluntary contraction and force development, are accompanied by dramatic changes of its mechanical and acoustical properties. Experimental data show that during contraction, the muscle's Young's modulus, shear viscosity, and anisotropy parameter are changed by over an order of magnitude. None of the existing models of muscle contraction and muscle biomechanics can adequately explain the phenomena observed. A new mathematical model [O. Rudenko and A. Sarvazyan, Acoust. Phys. (6), (2006)], has been developed relating the shear wave propagation parameters to molecular structure of the muscle and to the kinetics of the mechanochemical crossbridges between the actin and myosin filaments. New analytical solutions describing waves in muscle including nonlinear phenomena are found. A molecular mechanism for the dependence of acoustical characteristics of muscle on its fiber orientation and the contractile state is proposed. It is shown that although the anisotropy connected with the preferential direction along the muscle fibers is characterized by five elastic moduli, only two of these moduli have independent values in the muscle. The potential implications of the proposed model in terms of the acoustical assessment of muscle function are explored.

4:20

4pBB12. Phase aberration correction for a linear array transducer using ultrasound radiation force and vibrometry optimization: Simulation study. Matthew W. Urban and James F. Greenleaf (Dept. of Physiol. and Biomed. Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu)

Diagnostic ultrasound images suffer from degradation due to tissues with sound speed inhomogeneities causing phase shifts of propagating waves. These phase shifts defocus the ultrasound beam, reducing spatial resolution and image contrast in the resulting image. We describe a phase aberration correction method that uses dynamic ultrasound radiation force to harmonically excite a medium using amplitude-modulated continuous wave ultrasound created by summing two ultrasound frequencies at $f_0 = 3.0$ MHz and $f_0 + \Delta f = 3.0005$ MHz. The phase of each element of a linear array transducer is sequentially adjusted to maximize the radiation force and obtain optimal focus of the ultrasound beam. The optimization is performed by monitoring the harmonic amplitude of the scatterer velocity in the desired focal region using Doppler techniques. Simulation results show the ability to regain a 3.0-MHz focused field after applying a phase screen with an rms time delay of 95.4 ns. The radiation force magnitude increased by 22 dB and the resolution of the field was regained. Simulation results show that the focus of the beam can be qualitatively and quantitatively improved with this method. [This study was supported in part by Grants EB002640 and EB002167 from the NIH.]

4:35

4pBB13. Application of the optoacoustic technique to visualization of lesions induced by high-intensity focused ultrasound. Tatiana Khokhlova, Ivan Pelivanov, Vladimir Solomatin, Alexander Karabutov (Intl. Laser Ctr., Moscow State Univ., 119992, Moscow, Russia t_khokhlova@ilc.edu.ru), and Oleg Sapozhnikov (Moscow State Univ., 119992, Moscow, Russia)

Today several techniques are being applied to monitoring of high-intensity focused ultrasound (HIFU) therapy, including MRI, conventional ultrasound, and elastography. In this work a new method for noninvasive monitoring of HIFU therapy is proposed: the optoacoustic method. The optoacoustic technique is based on the excitation of wideband ultrasonic pulses through the absorption of pulsed laser radiation in tissue and subsequent expansion of the heated volume. The excited optoacoustic (OA) pulse contains information on the distribution of optical properties within the tissue—light scattering and absorption coefficients. Therefore, if thermal lesions have different optical properties than the untreated tissue, they will be detectable on the OA waveform. The considerable change in light scattering and absorption coefficients after tissue coagulation was measured using techniques previously developed by our group. Heating induced by HIFU also influences the OA signal waveform due to the rise of thermal expansion coefficient of tissue with temperature. This dependence was measured in order to evaluate the feasibility of the OA technique in temperature monitoring. An OA image of HIFU lesion induced by a 1.1 MHz focused transducer in a liver sample was reconstructed using a 64-element wideband array transducer for OA signal detection.

FRIDAY AFTERNOON, 1 DECEMBER 2006 OAHU ROOM, 1:00 TO 3:00 P.M.

Session 4pEAa

Engineering Acoustics: New Electroacoustic Transducers Utilizing Advanced Technolgies and Materials

Juro Ohga, Cochair

Shibaura Inst. of Technology, 3-9-14 Shibaura, Minato-ku, Tokyo 108-8548, Japan

James E. West, Cochair

Johns Hopkins Univ., Dept. of Electrical and Computer Engineering, Barton 105, 3400 N. Charles St., Baltimore, MD 21218-2686

Invited Papers

1:00

4pEAa1. Solid-state photo-microphones or pressure sensors by total reflection. Yasushi Suzuki (Gunma Natl. College 08 Tech., 580, Toriba-cho, Maebashi-shi, Gunma, 371-8530 Japan., suzuki@elc.gunma-ct.ac.jp) and Ken'iti Kido (Tohoku Univ., Yokohama-shi, Kanagawa, 226-0017 Japan)

Solid-state photo-microphones or pressure sensors are proposed. These sensors use a new principle, involving the optical total reflection at the boundary surface between glass and air. The critical angle for total reflection changes by the refractive index of air, which depends on the air density. Sound pressure changes the air density. Therefore, the sound pressure is measurable by detecting the intensity of the reflected light from the total reflection area. The sensitivity of the sensor is investigated theoretically. It is expected that the sensor has sufficient sensitivity for practical use, employing laser light and a curved boundary surface with a large radius of curvature. Some experiments are carried out to verify the theoretical investigations. A He-Ne laser or a laser diode is employed as a light source in the experiments. Experimental results show that the sensor has equivalent sensitivity to that which was theoretically estimated, but that sensitivity is very low. The sensor is useful as a pressure sensor, but it is difficult to realize a microphone for general use at the present. The microphones have no diaphragm and the upper limit in the frequency range is extremely high in principle.

1:20

4pEAa2. Micromachined microphones with diffraction-based optical interferometric readout. F. Levent Degertekin (G.W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, levent@gatech.edu), Neal A. Hall (Sandia Natl. Labs, Albuquerque, NM 87185-5800), and Baris Bicen (Georgia Inst. of Technol., Atlanta, GA 30332)

A diffraction-based optical method for integrated interferometric detection of micromachined microphone diaphragm displacement is described. With multichip optoelectronics integration, this approach yields highly sensitive optical microphones in mm-cube volumes. Since the microphone sensitivity does not depend on capacitance, this method changes the paradigm for the backplate and gap structure design. As a result, one can use millimeter size diaphragms to achieve wide frequency response and low thermal mechanical noise levels characteristic of precision measurement microphones. Furthermore, the electrical port of the device, which is freed by optical detection, is used for electrostatic actuation of the microphone diaphragm to tune microphone sensitivity and to generate self-characterization signals. Prototype optical microphone structures have been fabricated using Sandia National Laboratories' silicon based SwIFT-Lite^{TM} process. Measurements on these diaphragms show an A-weighted diaphragm displacement noise of 2.4 pm and flat electrostatic response up to 20 kHz. These results indicate the feasibility of realizing measurement microphones with 1.5-mm-diam diaphragms, 15-dBA internal noise, and 40-kHz bandwidth. Application of the detection method in a bio-inspired directional microphone for hearing aids is also discussed. [Work partially supported by NIH Grant 5R01DC005762-03, Sensing and Processing for Hearing Aids.]

1:40

4pEAa3. Hardware and software technologies for improvement of hearing characteristics of headphone reproduction. Kiyofumi Inanaga and Yuji Yamada (Audio Codec Development Dept., Technol. Development Group, SONY Corp., Shinagawa Tec., 12-15-3, Tokyo, 108-6201 Japan)

This report specifically describes commercialization technology of a headphone system with out-of-head localization applying dynamic head-related transfer functions (HRTFs) that can localize sound easily over a full 360 deg. A source image by output of conventional headphones is localized inside the listener's head. However, the image can be localized outside the listener's head by wearing headphones over a full 360 deg through accurate simulation of the listener's HRTFs. Developments of headphone systems using signal processing technology for data correction have given rise to the static binaural reproduction system (SBRS). The first part of this speech describes its psychoacoustic characteristics and challenges. A rotating dummy-head that is synchronized with the listener's head movement was produced experimentally to create the dynamic binaural reproduction system (DBRS). Using the DBRS, HRTFs synchronize with the listener's head movement. Psychoacoustic characteristics and advantages of the system are also discussed in this report. Further developments were made to realize the commercialization of the DBRS in areas including piezoelectric gyroscope head-tracking technology, headphone technologies that can reproduce real sound characteristics, and simplification of HRTF signal processing employing a simulator with electronic circuits. Finally, future visions for these technologies will be touched upon.

2:00

4pEAa4. Piezoelectret microphones: A new and promising group of transducers. Gerhard M. Sessler and Joachim Hillenbrand (Darmstadt Univ. of Technol., Merckstrasse 25, 64283 Darmstadt, Germany, g.sessler@nt.tu-darmstadt.de)

Piezoelectret microphones, first described a few years ago, are transducers based on the strong longitudinal piezoelectric effect of charged cellular polymers. Such microphones have recently been improved in two respects: Firstly, an expansion process was used to increase the piezoelectric d_{33} coefficients of cellular polypropylene (PP) films in the audio frequency range up to 600 pC/N and, secondly, stacking of several films was applied to increase the microphone sensitivity. Transducers with six films now show open-circuit sensitivities of up to 15 mV/Pa, comparable to that of electret microphones. Other characteristics of piezoelectret microphones are their low equivalent noise level of about 26 dB(A) and the very small total harmonic distortion of less than 0.1% at 140 dB SPL. The piezoelectric activity of the PP films and the microphone sensitivities are stable at room temperature but start to decay above 50 °C. Recently, directional piezoelectret microphones with various directional characteristics have been designed. Major advantages of piezoelectret microphones are their simple design, their low harmonic distortion, and their wide frequency range extending into the ultrasonic region.

2:20

4pEAa5. Expansion of frequency range for piezoelectric loudspeakers by new transducer construction. Juro Ohga (Shibaura Inst. of Technol., 3-7-5, Toyosu, Koto-ku, Tokyo 135-8548, Japan)

Although simple construction of piezoelectric loudspeakers engenders various merits, expansion of its working frequency range to the very low region is difficult because the mechanically stiff characteristics of conventional piezoelectric ceramic diaphragms prevent their large amplitude operation. This paper proposes two sorts of new piezoelectric loudspeaker construction that are suitable for low-frequency signal radiation. One idea is the use of a tuck-shape diaphragm by a PVDF polymer film bimorph. It has large surface area with a very low resonant frequency. Resonant frequencies and sensitivity frequency characteristics are examined, and control methods of local diaphragm bending are discussed. The other idea is the use of continuous revolution of a piezoelectric ultrasonic motor. It produces a completely controlled large output force because its output mechanical impedance is much greater than that of any conventional transducer or motor. An ultrasonic motor, whose stator is connected to a direct-radiator loudspeaker cone by a rod and whose rotor is burdened by a heavy metal ring, rotates with a constant velocity. Modulation of the velocity by using an audio signal imparts a driving force to the diaphragm because the heavy ring tends to keep a constant velocity. Experimental models suggest that this construction is useful.

2:40

4pEAa6. Modal array signal processing using circular microphone arrays applied to acoustic source detection and localization problems. Heinz Teutsch (Avaya Labs, 233 Mt. Airy Rd., Basking Ridge, NJ 07920, teutsch@avaya.com) and Walter Kellermann (Univ. of Erlangen-Nuremberg, Erlangen, Germany)

Many applications of acoustic signal processing rely on estimates of several parameters present in the observed acoustic scene such as the number and location of acoustic sources. These parameters have been traditionally estimated by means of classical array signal processing (CASP) algorithms using microphone arrays. Algorithms for parameter estimation solely based on the paradigm of CASP often suffer from the narrowband assumption underlying the signal model. This restriction limits their usability when wideband signals, such as speech, are present in the wave field under observation. We investigate the parameter estimation problem by applying the notion of wave field decomposition using baffled circular microphone arrays. The obtained wave field representation is used as the basis for "modal array signal processing algorithms." It is shown that by applying the notion of modal array signal processing, novel algorithms can be derived that have the potential to unambiguously detect and localize multiple simultaneously active wideband sources in the array's full field-of-view. Performance evaluations by means of simulations, measurements, and real-time case studies are presented.

FRIDAY AFTERNOON, 1 DECEMBER 2006 OAHU ROOM, 3:15 TO 6:00 P.M.

Session 4pEAb

Engineering Acoustics: Special Topics in Engineering Acoustics

Timothy W. Leishman, Cochair
Brigham Young Univ., Dept. of Physics and Astronomy, N247 ESC, Provo, UT 84602

Kiyofumi Inanaga, Cochair
Sony Corp., Shinagawa Tec. 12-15-3, Tokyo 108-6201, Japan

Contributed Papers

3:15

4pEAb1. Enhanced voided piezoelectric polymer for underwater acoustic sensors. Juan Arvelo (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099), Ilene Busch-Vishniac, and James West (Johns Hopkins Univ., Baltimore, MD 21218)

A charged voided polymer has been shown to exhibit large piezoelectricity. This material consists of injected air bubbles into polypropylene. This sheet of voided material is then biaxially stretched to elongate the voids. After stretching this material, a strong electric field is applied to cause dielectric breakdown of the gas in the voids, creating electric charges that are trapped in the polymer frame. Since the sides of the voids have opposite charges, they form macroscopic dipoles. When an external force is applied to this material, the voids become narrower, causing stronger dipole strength. A simple model of this voided material was implemented to derive formulas to estimate its piezoelectric constant, electromechanical coupling factor, resonance frequency, and sensor sensitivity based on electrical and mechanical properties of the polymer and gas in the voids. These formulas and a survey of available polymers and gases yielded promising combinations that result in more sensitive voided materials that satisfy selected criteria. These criteria include high sensitivity and maximum service temperature, low dissipation factor, and high dynamic compressibility, but low hydrostatic compressibility. This talk will describe the model, derive the formulas, uncover measured properties of candidate polymers and gases, and show calculated sensitivity of selected polymer/gas combinations.

3:30

4pEAb2. Basic study on one-dimensional transducer array using hydrothermally synthesized lead zirconium titanete poly-crystalline film. Akito Endo, Tomohito Hasegawa, Norimichi Kawashima, Shinichi Takeuchi (1614, Kurogane-cho, Aoba-ku, Yokohama, Kanagawa, 225-8502, Japan), Mutsuo Ishikawa, and Minoru Kurosawa (Midori-ku, Yokohama, Kanagawa 226-8502, Japan)

Recently, high-frequency miniature medical ultrasound probes with high resolution were actively developed. However, it is difficult to fabricate such tiny ultrasound probes using piezoelectric ceramic vibrator with thickness less than 100 μm. We deposited a PZT poly-crystalline film on a titanium substrate using the hydrothermal method and developed transducers using the PZT poly-crystalline film for ultrasound probes. In this study, we applied it to a miniature medical one-dimensional (1-D)-array-type ultrasound probe with resonance frequency of 10 MHz. After sputtering of pure titanium on the surface of a hydroxyapatite substrate, the titanium film was etched using the photolithography method to form a 1-D titanium film electrode array with 75 μm element pitch, 40 μm element width, and 4 mm element length to scan an ultrasound beam electronically by sector scan mode using phased-array technique. Thereby we fabricated a miniature 1-D-array-type ultrasound probe. A transmitted ultrasound pulse from 10 MHz commercial ultrasound probe was received by this fabricated 1-D-array type ultrasound probe with hydrothermally synthesized PZT poly-crystalline film vibrators.

3:45

4pEAb3. Analysis of a barrel-stave flextensional transducer using MAVART (model to analyze the vibrations and acoustic radiation of transducers) and ATILA (analysis of transducers by integration of LAplace equations) finite-element codes. Richard A. G. Fleming, Mark Kwiecinski, and Dennis F. Jones (Defence R&D Canada—Atlantic, P.O. Box 1012, Dartmouth, NS, Canada B2Y 3Z7, dennis.jones@drdc-rddc.gc.da)

A small barrel-stave flextensional transducer, designed and tested at Defence Research and Development Canada—Atlantic, is a candidate sound source for underwater coastal surveillance and acoustic communications applications. This high-power transducer has an outside diameter, length, and mass of 5.7 cm, 12.7 cm, and 1.1 kg, respectively. The measured fundamental flexural resonance frequency was 1.8 kHz with a transmitting voltage response of 118 dB/1μPa-m/V and an omnidirectional radiation pattern. Two finite-element models were developed for this transducer using the finite-element codes MAVART (Model to Analyze the Vibrations and Acoustic Radiation of Transducers) and ATILA (Analysis of Transducers by Integration of LAplace equations). Comparisons are made between the calibration measurements and the model predictions. [Work supported in part by Sensor Technology Limited.]

4:00

4pEAb4. Thermal behavior of high-power active devices with the ATILA (analysis of transducers by integration of LAplace equations) finite-element code. Jean-Claude Debus (Institut Superieur de l'Electronique et du Numerique, 41 Bv Vauban, 59046 Lille, Cedex France), John BlottmanIII, and Stephen Butler (Naval Undersea Warfare Ctr. Div. Newport, RI 02841)

Many active devices using piezoelectric ceramics are driven with very high power densities and long pulse lengths. Due to mechanical and dielectric losses in the materials, this produces heat, causing a temperature rise in the devices, which may lead to their mechanical failure. The thermal issues have been shown to be the limiting device design criteria over electric field and mechanical stress limits, yet the effect of the temperature on performance is generally not considered in the numerical models used during the design stage. A coupled electro-mechanical thermal analysis is implemented in the ATILA code. For a steady-state or transient solution, a thermal behavior is weakly coupled to the electromechanical response. The method may take advantage of the order-of-magnitude-greater time constant for thermal effects compared to mechanical behavior. A two-step analysis is performed whereby the electromechanical behavior is first computed, and the resulting dissipated power is then applied as a heat generator to determine the resulting temperature of the device. A high-drive, 31-mode, free flooded ring transducer and a sonar projector serve as validation of the numerical model. The approach addresses both the transient thermal response and the steady temperature profile that results from the high-power, high-duty-cycle drive.

4:15

4pEAb5. Development of multichannel optical sensor and visualization of vibration distribution. Jun Hasegawa and Kenji Kobayashi (Faculty of Eng., Takushoku Univ., 815-1 Tatemachi, Hachioji-shi, Tokyo 193-0985 Japan, jhase@es.takushoku-u.ac.jp)

A multi-channel optical sensor system was developed to measure vibrations simultaneously with high spatial resolution. As sensor elements, optical displacement sensor units were developed not to disturb the natural vibration. Each sensor unit, which consists of the optical fiber bundle and focusing lens, can detect the displacement of the object as the variation of the reflected light power. The sensor unit has a displacement resolution of 10 nm, a dynamic range of more than 90 dB, and a frequency band width of up to 80 kHz. Up to 64 sensor units can be arrayed as one sensor head, which realizes the simultaneous measurement of vibration distribution with the high spatial resolution of 4 mm. A calibrating function under the measurement circumstances was developed. Under calibration mode, the sensor array head is moved by a linear actuator, while the vibration of the object is stopped. Thus the calibrated data of each sensor unit can be obtained for the displacement magnitude. Measured vibration distributions can be monitored as the three-dimensional animations. With the system developed, several actuators for vibratory micro-injection were measured, and the system could reveal their detailed vibration distributions and could detect the existence of a failure portion of some actuator.

4:30

4pEAb6. Prediction of howling for a sound system in an acoustical environment with both reverberant and direct sounds. Hideki Akiyama and Juro Ohga (Shibaura Inst. of Technol., 3-7-5 Toyosu, Koto-ku, Tokyo 135-8548, Japan, m106003@shibaura-it.ac.jp)

Prediction of howling is a key technology for a howling suppression design for a sound system with a loudspeaker and microphone. A howling occurrence prediction method for a sound system in a reverberant room has already been presented [J. Ohga, J. Sakaguchi, "Prediction of howling of a sound system in a reverberant room," W. C. Sabine Centennial Symposium (ASA, New York, 1994), 2aAAd4]. It is apparently useful for ordinary public address systems whose distances of loudspeakers from microphones are large. However, this result was not perfect because the direct sound component is not negligible in hands-free telephones or teleconference systems whose loudspeakers and microphones are set close to each other. This report gives a quantitative howling occurrence prediction method for a sound system in an acoustical environment with both reverberant and direct sounds. The following design parameters are obtained: (1) the increase of howling occurrence level from the power average value, (2) the level occurrence probability, and (3) the critical level chart given by an equation as a function of direct and reverberant sounds ratio. Prediction results for particular examples are compared with calculations of sound-field transfer functions. Results confirmed that it is practical.

4:45

4pEAb7. Effect of background noise on dialogue in telephony. Koichi Amamoto and Juro Ohga (Sibaura Inst. of Technol., 3-7-5 Toyosu, Koto-ku, Tokyo, 135-8548, Japan, m106006@shibaura-it.ac.jp)

Recent developments of mobile telephones include new sorts of impairments against speech. Conventional evaluation method for impairments by a talker and a few listeners cannot apply to these new ones, because they are brought by long signal delay. The effect of it cannot discriminate by "one-sided" test. This research relates to a speech quality evaluation by conversation between two persons. Variation of conversation stream is observed by addition of pink noise of various levels to a dialogue by microphones and earphones. Length of sentences and frequency of repeats are quantified and their meanings are discussed

5:00

4pEAb8. Best practices for auditory alarm design in space applications. Durand Begault and Martine Godfroy (Human Systems Integration Div., NASA Ames Res. Ctr., Moffett Field, CA 94035)

This presentation reviews current knowledge in the design of auditory caution and warning signals, and sets criteria for development of "best practices" for designing new signals for NASA's Crew Exploration Vehicle (CEV) and other future spacecraft, as well as for extra-vehicular operations. A design approach is presented that is based upon cross-disciplinary examination of psychoacoustic research, human factors experience, aerospace practices, and acoustical engineering requirements. Existing alarms currently in use with the NASA Space Shuttle flight deck are analyzed and then alternative designs are proposed that are compliant with ISO 7731 ("Danger signals for work places Auditory Danger Signals") and that correspond to suggested methods in the literature to insure discrimination and audibility. Parallel analyses are shown for a sampling of medical equipment used in surgical, periop, and ICU contexts. Future development of auditory sonification techniques into the design of alarms will allow auditory signals to be extremely subtle, yet extremely useful in subtly indicating trends or root causes of failures. [Work funded by NASA's Space Human Factors Engineering Project.]

5:15

4pEAb9. Acoustic signal analysis for forensic applications. Durand Begault and Christopher Peltier (Audio Forensic Ctr., Charles M. Salter Assoc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, durand.begault@cmsalter.com)

Acoustical analysis of audio signals is important in many legal contexts for determining the authenticity, originality, and continuity of recorded media; determining the circumstances of events in question that may have been recorded; for determining the audibility of signals; and for identification or elimination of talkers as a match to an unknown exemplar. Recorded media are analyzed in forensic applications using both familiar techniques (waveform and spectral analyses) and more novel methods (e.g., ferro fluid development of media; specialized tape heads with non-standard reproduction characteristics; crystal microscopy; detection and matching to power grid frequencies). Audibility analyses frequently require careful reconstructive field measurements and criteria in excess of normally accepted standards. Voice identification-elimination protocols must account for examiner bias and exemplar quality and can be described using a receiver operator curve (ROC) model. This presentation gives an overview of these techniques and their comparative advantages for specific forensic applications.

5:30

4pEAb10. Without low-pass filter for the 1-bit digital amplifier. Kiyoshi Masuda (Coroprate Res. and Development Group, SHARP Corp., 2613-1 Ichinomoto-cho, Tenri-shi, Nara, Japan) and Yoshio Yamasaki (Waseda Univerity, Okubo, Shinzyuku-ku, Tokyo, Japan)

SHARP collaborated with Waseda University from 1990 for 1-bit digital technology. SHARP had started to receive an order for the 1-bit digital amplifier "SM-SX100" on 20 August 1999. Until today, we have introduced the 1-bit digital amplifier for audio, flat panel TV (LCD TV), and PC. These 1-bit amplifiers provided low-pass filter for the final stage, which is provided after 1-bit digital switching. We have to achieve more good sound and reduce deterioration of this low-pass filter. We have introduced new 1-bit digital amplifier without this low-pass filter beginning this April. This means we controlled the 1-bit digital signal to directly operate the speaker. We have proved a better effect for sound to compare the new 1-bit digital amplifier with the PWM switching amplifier, the A-class amplifier and the 1-bit digital amplifier with low-pass filter. If we do not measure any improvement for this new 1-bit digital amplifier, it has large radiation noise. We had achieved a reduction to the limit level of FCC, Denanhou, etc.

5:45

4pEAb11. Force-frequency effect of thickness mode langasite resonators. Haifeng Zhang (W317.4 Nebraska Hall, Univ. of Nebraska, Lincoln, NE 68588-0526, hfzhang@bigred.unl.edu), Joseph A. Turner, Jiashi Yang (Univ. of Nebraska, Lincoln, NE 68588-0526), and John A. Kosinski (U.S. Army CECOM, Fort Monmouth, NJ 07703-5211)

Langasite resonators are of recent interest for a variety of applications because of their good temperature behavior, good piezoelectric coupling, low acoustic loss, and high Q factor. The force-frequency effect describes the shift in resonant frequency a resonator experiences due to the application of a mechanical load. A clear understanding of this effect is essential for many design applications such as pressure sensors. In this presentation, the frequency shift is analyzed theoretically and numerically for thin, circular langasite plates subjected to a diametrical force. The results are compared with experimental measurements of the same system for a variety of langasite resonators with various material orientations. In addition, the sensitivity of force-frequency effect is analyzed with respect to the nonlinear material constants. A comparison between the force-frequency effect of langasite and quartz resonators is also made. Finally, the application of such measurements for determining third-order elastic constants is discussed. [Work supported by ARO.]

FRIDAY AFTERNOON, 1 DECEMBER 2006 IAO NEEDLE/AKAKA FALLS ROOM, 1:20 TO 4:25 P.M.

Session 4pMU

Musical Acoustics and Psychological and Physiological Acoustics: Acoustic Correlates of Timbre in Musical Instruments

James W. Beauchamp, Cochair
Univ. of Illinois Urbana-Champaign, School of Music, Dept. of Electrical and Computer Engineering, 1002 Eliot Dr., Urbana, IL 61801

Mashashi Yamada, Cochair
Kanazawa Inst. of Technology, Dept. of Media Informatics, 3-1 Yatsukaho, Hakusan, Ishikawa 924-0838, Japan

Invited Papers

1:20

4pMU1. A meta-analysis of acoustic correlates of timbre dimensions. Stephen McAdams, Bruno Giordano (CIRMMT, Schulich School of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC, Canada H3A 1E3), Patrick Susini, Geoffroy Peeters (STMS-IRCAM-CNRS, F-75004 Paris, France), and Vincent Rioux (Maison des Arts Urbains, F-75020 Paris, France)

A meta-analysis of ten published timbre spaces was conducted using multidimensional scaling analyses (CLASCAL) of dissimilarity ratings on recorded, resynthesized, or synthesized musical instrument tones. A set of signal descriptors derived from the tones was drawn from a large set developed at IRCAM, including parameters derived from the long-term amplitude spectrum (slope, centroid, spread, deviation, skewness, kurtosis), from the waveform and amplitude envelope (attack time, fluctuation, roughness), and from variations in the short-term amplitude spectrum (flux). Relations among all descriptors across the 128 sounds were used to determine families of related descriptors and to reduce the number of descriptors tested as predictors. Subsequently multiple correlations between descriptors and the positions of timbres along perceptual dimensions determined by the CLASCAL analyses were computed. The aim was (1) to select the subset of acoustic descriptors (or their linear combinations) that provided the most generalizable prediction of timbral relations and (2) to provide a signal-based model of timbral description for musical instrument tones. Four primary classes of descriptors emerge: spectral centroid, spectral spread, spectral deviation, and temporal envelope (effective duration/attack time). [Work supported by CRC, CFI, NSERC, CUIDADO European Project.]

1:40

4pMU2. Perceptual acoustics of consonance and disssonance in multitimbral triads. Roger Kendall (Music Cognition and Acoust. Lab., Program in Systematic Musicology, UCLA, 405 Hilgard Ave., Los Angeles, CA 90095, kendall@ucla.edu) and Pantelis Vassilakis (DePaul Univ., Chicago, IL 60614)

Most studies of consonance and dissonance assume a singular spectrum for the constituent intervals of a dyad. Recently, the principal author conducted experiments evaluating triads consisting of digitally mixed combinations drawn from the MUMS single-note natural-instrument recordings. Results indicated that the main effect of ratings for consonance and dissonance correlated well with studies using artificial signals. However, interaction effects suggested perceptual differences related to the timbral differences across combinations. The present experiment evaluates perceptual and acoustical variables of the ten possible triadic combinations created with C4 as the lower and the ten with C5 as the upper notes. UCLA wind ensemble performers on oboe, flute, and clarinet, combinations designed to span timbral space, were digitally recorded. Analyses include perceptual ratings of consonance and dissonance, similarity, as well as acoustical analysis of roughness using a recently developed model. Since natural performances of any type vary in fundamental frequency, additional experiments will employ emulated oboe, flute, and clarinet (using the Kontakt Silver synthesizer in Sibelius 4) as well as purely synthetic stimuli, in order to ascertain the relationship of time-variant spectral properties to consonance, dissonance, and perceived similarity.

2:00

4pMU3. Multidimensional scaling analysis of centroid- and attack/decay-normalized musical instrument sounds. James W. Beauchamp (School of Music and Dept. of Elect. & Comput. Eng., Univ. of Illinois at Urbana–Champaign, Urbana, IL 61801, jwbeauch@uiuc.edu), Andrew B. Horner (Hong Kong Univ. of Sci. & Technol., Kowloon, Hong Kong), Hans-Friedrich Koehn, and Mert Bay (Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801)

Ten sustained musical instrument tones (bassoon, cello, clarinet, flute, horn, oboe, recorder, alto saxophone, trumpet, and violin) were spectrally analyzed and then equalized for duration, attack and decay time, fundamental frequency, number of harmonics, average spectral centroid, and presentation loudness. The tones were resynthesized both with time-varying harmonic amplitudes and frequencies (dynamic case) and fixed amplitudes and frequencies (static case). Tone triads were presented to ten musically experienced listeners whose tasks were to specify the most dissimilar and most similar pairs in each triad. Based on the resulting dissimilarity matrix, multidimensional scaling (MDS) was used to position the instruments in two- and three-dimensional metric spaces. Two measures of instrument amplitude spectra were found to correlate strongly with MDS dimensions. For both the static- and dynamic-case 2-D solutions, the ratio of even-to-odd rms amplitudes correlated strongly with one of the dimensions. For the dynamic case, spectral centroid variation correlated strongly with the second dimension. Also, 2-D solution instrument groupings agreed well with groupings based on coefficients of the first two components of a principle components analysis representing 90% of the instruments' spectral variance. [This work was supported by the Hong Kong Research Grants Council's CERG Project 613505.]

2:20

4pMU4. Sound synthesis based on a new micro timbre notation. Naotoshi Osaka (School of Eng., Tokyo Denki Univ., 2-2, Kanda-Nishikicho, Chiyoda-ku, Tokyo, 101-8457, Japan, osaka@im.dendai.ac.jp), Takayuki Baba, Nobuhiko Kitawaki, and Takeshi Yamada (Univ. of Tsukuba, Japan)

Timbre has become a major musical factor in contemporary and computer music. However, sufficient timbre theory has not yet been established. The author is challenging to create new timbre theory for music composition. The first step of its construction is to make the timbre descriptive. A micro timbre is defined, which is a perceptual impression of a sound with approximately 50 to 100-ms duration, and describe sound as a micro timbre sequence. This can be used as a new notation system in place of common music notation. In dictation, micro timbre sequence and correspondent duration sequence are perceptually recorded. When synthesizing from this notation, sounds corresponding to the notation systems are either physically synthesized or searched for in a large sound database to generate sound data for a given duration. Two sequential sound data instances are first represented in sinusoidal representations and then are concatenated using a morphing technique. Sounds generated by a stream of water and similar sounds are described using the method as examples. Then scripts describing electronic sounds are introduced and explained. The ability to record, transmit to others, and resynthesize timbre is one of the useful functions of the theory.

2:40

4pMU5. Timbre representation for automatic classification of musical instruments. Bozena Kostek (Gdansk Univ. of Technol., Narutowicza 11/12, PL-80-952 Gdansk, Poland)

Human communication includes the capability of recognition. This is particularly true of auditory communication. Music information retrieval (MIR) turns out to be particularly challenging, since many problems remain still unsolved. Topics that should be included within the scope of MIR are automatic classification of musical instruments/phrases/styles, music representation and indexing, estimating musical similarity using both perceptual and musicological criteria, recognizing music using audio and/or semantic description, language modeling for music, auditory scene analysis, and others. Many features of music content description are based on perceptual phenomena and cognition. However, it can easily be observed that most of the low-level descriptors used, for example, in musical instrument classification are more data- than human-oriented. This is because the idea behind these features is to have data

defined and linked in such a way as to be able to use it for more effective automatic discovery, integration, and reuse in various applications. The ambitious task is, however, to provide seamless meaning to low- and high-level descriptors such as timbre descriptors and linking them together. In such a way data can be processed and shared by both systems and people. This paper presents a study related to timbre representation of musical instrument sounds.

3:00–3:15 Break

3:15

4pMU6. An attempt to construct a quantitative scale of musical brightness for short melodies implementing timbral brightness. Masashi Yamada (Kanazawa Inst. of Technol., 3-1 Yatsukaho, Hakusan, Ishikawa 924-0838, Japan, m-yamada@neptune.kanazawa-it.ac.jp)

It is known that a major tune is brighter than a minor one, and that music played in a faster tempo and a higher register is brighter than a slower and lower one. However, it has not been clarified how these factors quantitatively determine the musical brightness. On the other hand, it has been clarified that the timbral brightness of a tone corresponds well to the spectral centroid. In the present study, major and minor scales and two short melodies were played with pure tones, and listeners evaluated their musical brightness. For each performance, the spectral centroid was calculated for the overall-term spectrum during the performance on the transformed frequency scale of the ERB rate. The results showed that the musical brightness of the ascending scale increases proportionally as the spectral centroid shown in the ERB rate increases. Using this, a quantitative scale of musical brightness, BM, was constructed. The results also showed that the difference in the musical brightness between major and minor scales corresponded to the transposition of approximately 5 ERB rate, and doubling the speed corresponded to the upper shift of the centroid in approximately 2.5 ERB rate.

3:35

4pMU7. Subjective congruency between a sound effect and a switching pattern of a visual image. Shinichiro Iwamiya, Motonori Arita, and Sun Su (Dept. of Acoust. Design, Kyushu Univ., 4-9-1, Shiobaru, Minami-ku, Fukuoka 185-8540, Japan)

The relationship between the transformation of a visual image and the pitch pattern of sound can create formal congruency between sounds and moving pictures. This effect by switching patterns of enlarging and reducing images that were combined with ascending and descending pitch scales was examined. Rating experiments showed two congruent patterns of the combination of switching and scale patterns; one was a combination of an ascending pitch scale and an enlarging image pattern, and the other a combination of a descending pitch scale and a reducing image pattern. These forms of matching might be based on a Doppler illusion. An additional pair of congruent patterns for combinations of switching and scale patterns was also found: one was a combination of an ascending pitch scale and a sliding movement from left to right, and the other a combination of a descending pitch scale and a sliding movement from right to left. These forms of matching might be based on the correspondence of a progressive sensation. Further, the formal congruency between a pitch pattern and the formal transportation can contribute to integrating auditory and visual information and to making audio-visual products more impressive.

Contributed Papers

3:55

4pMU8. SRA: An online tool for spectral and roughness analysis of sound signals. Pantelis Vassilakis (School of Music, ITD, Libraries, DePaul Univ., 2350 N. Kenmore Ave., JTR 207, Chicago, IL 60614)

SRA performs spectral and roughness analysis on user-submitted 250- to 1000-ms-long portions of sound files (.wav/.aif formats). Spectral analysis incorporates an improved STFT algorithm [K. Fitz and L. Haken, J. Aud. Eng. Soc. **50**(11), 879–893 (2002)] and automates spectral peak-picking using the Loris open source C++ class library [Fitz and Haken (CERL Sound Group)]. Users can manipulate three spectral analysis/peak-picking parameters: analysis bandwidth, spectral-amplitude normalization, and spectral-amplitude threshold. Instructions describe the parameters in detail and suggest settings appropriate to the submitted files and questions of interest. The spectral values obtained from the analysis enter a roughness estimation model [P. N. Vassilakis, Sele. Rep. in Ethnomusicol. **12**, 119–144 (2005)], outputting roughness values for each individual sine-pair in the file's spectrum and for the entire file. The roughness model quantifies the dependence of roughness on a sine-pair's (a) intensity (combined amplitude of the sines), (b) amplitude fluctuation degree (amplitude difference of the sines), (c) amplitude fluctuation rate (frequency difference of the sines), and (d) register (lower sine frequency). Presentation of the roughness estimation model and the online tool will be followed by a discussion of research studies employing it and an outline of future possible applications. [Work supported by DePaul University and Eastern Washington University. Programmed by K. Fitz.]

4:10

4pMU9. Further spectral correlations of timbral adjectives used by musicians. Alastair C. Disley, David M. Howard, and Andrew D. Hunt (Dept. of Electron., Univ. of York, Heslington, York, YO10 5DD, UK)

As part of a project to develop a synthesis interface which nontechnical musicians should find intuitive, the adjectives musicians use to describe timbre have been studied in a large-scale listening test covering the timbre space of Western orchestral instruments. These were refined in previous work by the authors (A. C. Disley *et al.* "Spectral correlations of timbral adjectives used by musicians," J. Acoust. Soc. Am. **119**, 3333, (2006)] to a set of ten words which had good common understanding and discrimination between the samples (bright, clear, dull, gentle, harsh, nasal, percussive, ringing, thin, and warm). To help explore potential relationships between these adjectives and spectral features, 20 listeners participated in a further listening experiment, comparing samples in pairs to produce dissimilarity data. Multidimensional scaling produced dimensions which were compared with a large number of spectral and time-domain analyses of the stimuli, suggesting a number of significantly correlated spectral cues with some of the adjectives. These results are compared with previous studies by the authors and others, showing both similarities and differences, suggesting that collective consideration of timbral adjectives is more likely to result in simultaneously applicable theories of acoustic correlates than individual consideration of words.

Session 4pNSa

Noise and Architectural Acoustics: Soundscapes and Cultural Perception II

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

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

Contributed Papers

1:00

4pNSa1. Mapping soundscapes in urban quiet areas. Gaetano Licitra (ARPAT, Tuscany Regional Agency for Environ. Protection, Via N. Porpora, 22-50144, Firenze, Italy), Gianluca Memoli (Memolix, Environ. Consultants, 56127 Pisa, Italy), Mauro Cerchiai, and Luca Nencini (ARPAT, 56127 Pisa, Italy)

Innovative action plans in noise-polluted environments require the description of the existing soundscape in terms of suitable indicators. The role of these indicators, giving the "fingerprint" of a fixed soundscape, would be not only to measure the improvement in the sound quality after the action taken, but also to guide the designer in the process, providing a reference benchmark. One of the open questions on new indicators is the way they relate to existing ones and to people's perception. The present work will describe a "Sonic Garden" in Florence, using both the "slope" indicator (constructed from the LA_{eq} time history and related in previous studies to people's perception) and classical psychoacoustical parameters (level, spectral structure, and perceived characteristics such as loudness, sharpness, fluctuation, and roughness). The latter parameters will be acquired using a binaural technique.

1:15

4pNSa2. A questionnaire survey of the attitude of Japanese and foreign residents in Japan to sound masking devices for toilets. Miwako Ueda and Shin-ichiro Iwamiya (Grad. School of Design, Kyushu Univ., Iwamiya Lab. 4-9-1, Shiobaru, Minami-ku, Fukuoka 815-8540 Japan, amaria@white.livedoor.com)

Unique sound masking devices for toilets can be used in women's restrooms in Japan. Such devices function to produce the sound of flushing water without actual flushing. To mask the sound of bodily functions, women tended to flush the toilet continuously while using it, thereby wasting a large amount of water. In the circumstances, sound masking devices have been introduced to public toilets. We have recently conducted a questionnaire survey to clarify the attitude of people toward such sound masking devices for toilets. The results of the survey showed that many Japanese women know such devices and often use them, that foreign women currently living in Japan also know that such devices exist, and that some Japanese men have heard of such devices but never used them. Many Japanese women are quite embarrassed at the thought that someone else can hear them while they are on the toilet. Many noted the necessity of such devices and required a wide range of setting for toilets in public spaces. However, they are not satisfied with the sound quality of the playback toilet flush sounds of currently available devices. The above results suggest that the sound quality of such devices should be improved.

1:30–2:00
Panel Discussion

FRIDAY AFTERNOON, 1 DECEMBER 2006 MAUI ROOM, 2:10 TO 4:35 P.M.

Session 4pNSb

Noise, Physical Acoustics, and Structural Acoustics and Vibration: Acoustics of Sports

Joseph Pope, Cochair
Pope Engineering Company, P.O. Box 590236, Newton, MA 02459-0002

Kenji Kurakata, Cochair
AIST, 1-1-1 Higashi, Tsukuba, Ibaraki 305-8566, Japan

Chair's Introduction—2:10

Invited Papers

2:15

4pNSb1. A review of the vibration and sounds of the crack of the bat and player auditory clues. Robert Collier (Thayer School of Eng., 8000 Cummings Hall, Hanover, NH 03755)

The purpose of this paper is to review the state-of-the-art in the acoustics of baseball. As is well known, the crack of the bat is an important phenomenon of solid wood bats and metal bats. Each has a very different sound signature. At the 148th meeting of the ASA in 2004, the author and coauthors Ken Kaliski and James Sherwood presented the results of laboratory and field tests, which showed

that the spectral characteristics of radiated sound are dependent on the ball-bat impact location and resultant bat vibration of both solid wood and tubular metal bats. These results will be reviewed together with those of other investigators in the context of player auditory clues and the player's response in game situations.

2:35

4pNSb2. Measurements of the impact sound of golf clubs and risk of hearing impairment. Kenji Kurakata (Natl. Inst. of Adv. Industrial Sci. and Technol. (AIST), 1-1-1 Higashi, Tsukuba, Ibaraki, 305-8566 Japan, kurakata-k@aist.go.jp)

The ball-impact sounds of golf clubs with metal heads and of a club with a wood head were measured to investigate their different acoustic properties. Hitting was executed using either a swing machine or a human player. Results of these analyses showed that the metal-head clubs generated sounds around 100 dB ($L_{p\mathrm{A,Fmax}}$). This level was 5–15 dB higher than that of the wood-head club. The sounds of the metal-head clubs had greater power in the high-frequency region of 4 kHz and above compared to the wood-head club, which particularly increased the overall sound levels. These results suggest that it would be desirable to develop a metal head with pleasant sound qualities, keeping the sound level lower to minimize hearing damage. Some of these measurement data were published in Japanese in a previous paper [K. Kurakata, J. INCE/J **26**, 60–63 (2002)].

2:55

4pNSb3. New underwater sound system for synchronized swimming: The 9th International Swimming Federation Championships. Takayuki Watanabe, Shinji Kishinaga (YAMAHA Ctr. for Adv. Sound Technologies, 203 Matsunokijima, Iwata, Shizuoka, 438-0192 Japan), Tokuzo Fukamachi (YAMAHA Motor Marine Operations, Arai, Hamana, 438-8501 Japan), and Osamu Maeda (YAMAHA Motor Adv. Technol. Res. Div., Iwata, 438-8501 Japan)

There have been concerns about the differences between underwater sound fields in a temporary fiberglass-reinforced plastic (FRP) pool and in a conventional reinforced concrete (RC) pool. A temporary FRP pool was to be used for competitions at the World Swimming Championships in Fukuoka. We considered three items as key factors for a swimming pool used for synchronized swimming: (1) the sound source itself (output level, fluctuations in frequency characteristics); (2) the effect of materials used in pool construction upon sound source installation conditions; and (3) the effect of the mth mode low-frequency cutoff in "shallow water." To improve basic problems related to the first factor, we developed a new actuator-driven underwater sound system (YALAS), which can eliminate the effect of installation conditions for underwater speakers in the FRP pool. This new underwater system has now seen practical use in competitions. The report summarizes this new underwater sound system and compares the system with conventional systems in terms of its acoustic characteristics. The system can offer music with sufficient audibility in water. We gained a good reputation with competitors because the system showed superior performance to conventional systems in sound volume and quality, and in uniformity of sound distribution.

3:15

4pNSb4. Acoustics of the Great Ball Court at Chichen Itza, Mexico. David Lubman (14301 Middletown Ln., Westminster, CA 92683)

The ball game has played a central role in Mayan religion and culture for 5000 years. Thousands of ball courts have been discovered. The Great Ball Court (GBC) at Chichen Itza is a late development and is architecturally unique. Two remarkable acoustical features were noticed during excavation in the 1920s, but never explained or interpreted. A whispering gallery permits voice communication between temples located about 460 feet (140 m) apart. A profound flutter echo is heard between the two massive parallel walls of the playing field, about 270 ft (82 m) long, 28 ft (8.5 m) high, and 119 ft (36 m) apart. Until recently, most archaeologists dismissed acoustical features at Mayan sites as unintended artifacts. That is now changing. Stimulated by archaeological acoustic studies and reports since 1999, eminent Mayanists Stephen Houston and Karl Taube have reinterpreted certain Mayan glyphs as vibrant sounds and ballcourt echoes, and have famously called for a new archaeology of the senses, especially hearing, sight, and smell [Cambridge Archaeol. J. **10** (2) 261–294 (2000)]. By interpreting architectural, psychoacoustic, and cognitive features of the GBC in the context of ancient Mayan culture, this paper speculates that acoustical effects at the GBC may be original design features.

Contributed Papers

3:35

4pNSb5. Sleep disturbance caused by shooting sounds. Joos Vos (TNO Human Factors, P.O. Box 23, 3769 ZG Soesterberg, The Netherlands, joos.vos@tno.nl)

In the present study relations between the sound level of shooting sounds and the probability of behaviorally confirmed noise-induced awakening reactions were determined. The sounds were presented by means of loudspeakers in the bedrooms of 30 volunteers. The shooting sounds had been produced by a small and a medium-large firearm, and the stimuli consisted of individual bangs or volleys of ten isolated or partly overlapping impulses. Aircraft sound was included as a reference source. The sounds were presented during a 6-h period that started 75 min after the beginning of the sleeping period. The time period between the various stimuli varied between 12 and 18 min, with a mean of 15 min. To cope with at least a relevant portion of habituation effects, each subject participated in 18 nights to be completed within 4 weeks. Preliminary results are presented both for the awakening reactions described above, and for various other dependent variables collected with the help of an actimeter or determined by means of subjective rating scales. [Work supported by the Dutch Ministry of Defense.]

3:50

4pNSb6. Sound inside a gymnasium. Sergio Beristain (ESIME, IPN, IMA., P.O. Box 12-1022, Narvarte, 03001, Mexico City, Mexico)

A new gymnasium for a sports club was designed taking acoustic confort into consideration, in order to accomodate sports practice, sports events with the public, or musical and drama presentations, taking advatage of its large capacity for the public and performers. The floor plan included room enough for a basketball court with public space on one side, where grades for about 200 people will be permanently installed. Walls were treated in a way that is useful for the sports practice (hard surfaces), with hidden absorption material to reduce the usual reverberant

4p FRI. PM

field inside the court, and to allow for sound events with only the addition of a mat (to protect the floor woodwork) and extra grades and the sound reinforcement system.

4:05

4pNSb7. Occupational and recreational noise exposures at stock car racing circuits. Chucri A. Kardous, Thais Morata, and Luann E. Van Campen (Natl. Inst. for Occupational Safety Health, 4676 Columbia Pkwy., Cincinnati, OH 45226, ckardous@cdc.gov)

Noise in stock car racing is accepted as a normal occurrence but the exposure levels associated with the sport have not been adequately characterized. Researchers from the National Institute for Occupational Safety and Health (NIOSH) conducted an exploratory assessment of noise exposures to drivers, pit crew, team staff, and spectators at three stock car racing events. Area measurements were made during race preparation, practice, qualification, and competition. Personal dosimetry measurements were conducted on drivers, crew members, infield staff, and spectators. Findings showed time-weighted averages (TWA) that ranged from 94 decibels A-weighted (dBA) for spectators to 114 dBA for car drivers. Peak sound-pressure levels exceeded the maximum allowable limit of 140 decibels (dB) during race competitions. Personal exposure measurements exceeded the NIOSH recommended exposure limit (REL) of 85 dBA as an 8-h TWA in less than a minute for one driver during practice, within 2 min for pit crew and infield staff, and 7 to 10 min for spectators during the race. Hearing protection use was variable and intermittent among crew, staff, and spectators. Among drivers and crew, there was greater concern for communication performance than for hearing protection.

4:20

4pNSb8. Sports acoustics: Using sound from resonant shells, vibrating cylinders, strumming shafts, and water impact to evaluate athletic performance. David G. Browning (Dept. of Phys., Univ. of Rhode Island, Kingston, RI 02881, decibeldb@aol.com) and Peter M. Scheifele (Univ. of Connecticut, Storrs, CT 06269)

The sound from equipment used and/or specific acts during athletic competition, such as hitting a baseball with an aluminum bat, carries beyond the playing field and can provide a nonobtrusive method to evaluate athletic performance—such as where on the bat the ball was hit. Standardized equipment guarantees repeatability, for example, every volleyball resonates at the same frequency. Each major sport can have unique noise interference which in some circumstances can be overwhelming, and the distance from the sound source can vary significantly during a game. Still, it will be shown that useful performance information can be obtained under realistic conditions for at least the following sports: volleyball, softball, baseball, golf, swimming and diving, soccer, and football.

FRIDAY AFTERNOON, 1 DECEMBER 2006 WAIANAE ROOM, 1:30 TO 6:20 P.M.

Session 4pPA

Physical Acoustics and Biomedical Ultrasound/Bioresponse to Vibration: Sound Propagation in Inhomogeneous Media II

Takahiko Otani, Cochair
Doshisha Univ., Lab. of Ultrasonic Electronics, Kyotonabe-shi, Kyoto 610-0321, Japan

James G. Miller, Cochair
Washington Univ., Dept. of Physics, 1 Brookings Dr., St. Louis, MO 63130

Invited Paper

1:30

4pPA1. Observables and prediction modeling in the presence of ultra-wideband heterogeneity. John J. McCoy (The Catholic Univ. of America, Washington, DC 20064)

Underlying virtually all propagation and scattering models is an intuitive understanding; the acoustic field is observable using a device of a sufficiently small size to obtain a sufficiently dense set of discrete measurements. This assumes the field variation cuts off at an inner length scale larger than the device size, assuring that no information is lost to the inherent spatial averaging in any measurement. This understanding is faulty in the presence of environment heterogeneity observed on an extreme range of length scales. The reason is that all physical devices have finite accuracy, which limits their ability to capture variation on scales significantly larger than their size, in the presence of variation on intermediate scales. A more refined understanding of the ability to observe a field requires multiple devices, an unbounded hierarchy in the limit, to obtain multiple dense sets of discrete "observables." This, then, suggests a different class of prediction models for environments with ultra-wideband heterogeneity, expressed in multiple sets of discrete variables, each set describing field variation in a limited subband. A framework for formulating these prediction models and their application to a scenario for which environment heterogeneity has no inner scale cutoff is presented.

1:50

4pPA2. Parameters arising from the Burridge-Keller formulation for poroelastic media, especially for granular media and marine sediments. Allan D. Pierce, William M. Carey, and Paul E. Barbone (Boston Univ., Boston, MA 02215)

It was previously shown [Pierce *et al.* (2006)] that the Burridge-Keller formulation [J. Acoust. Soc. Am. (1981)] rigorously justifies the low-frequency version [(1956a)] of Biot's equations. Implementation involves two microscale problems: (1) incompressible viscous flow driven in a highly irregular space with rigid walls by a uniformly and externally applied apparent pressure distribution and (2) elastostatic deformation of an intricate elastic web caused by the joint influence of a distributed constant body force and by uniform tractions (including an external pressure) on the web's exposed surface. Microscale averages produce the Biot "constants." Theoretical devices of applied mechanics and mathematics yield estimates of these and related parameters. In particular, it is shown that Wood's equation is a reasonable first approximation for the sound speed in sediments in the low-frequency limit. The formulation also yields an estimation for the sound speed in the high-frequency limit, when the viscous boundary layers become thin. The well-known result that the attenuation varies as $f^{1/2}$ in the high-frequency limit also results without the necessity of Biot's heuristic patching theory. Various heuristic approximations due to Gassmann, to Geertsma and Smit, and to Stoll and Bryant are analytically and numerically assessed.

2:05

4pPA3. Sound propagation in the mixtures of liquid and solid aggregate; similarities at micro- and nanoscales.. Hasson M. Tavossi (Dept. of Physical and Environ. Sci., Mesa State College, 1100 North Ave., Grand Junction, CO 81504)

Sound propagation phenomena in certain liquid and solid aggregate mixtures, at micrometer scales, in some cases resemble the wave propagation behaviors of materials observed at nanometer and atomic scales. For example, it can be shown that the sound wave dispersion, attenuation, and cutoff-frequency effects depend on the same structural parameters as those observed at nano or atomic levels and are similar at both scales. Therefore, to investigate theoretical models of wave and matter interactions it is more convenient to use, as experimental tools, the readily analyzable models of wave propagation, in mixtures of solid and liquid, constructed at micrometer scales. Theoretical findings on sound propagation in the mixtures of liquid and solid particles at micrometer scales will be discussed. These results show the resemblance to the behavior of acoustic phonons, the lattice thermal vibrations of crystalline structures, at radically different scales. Experimental data on wave dispersion, attenuation, bandpass, and cutoff frequency effects, measured for sound propagation, in inhomogeneous materials consisting of mixtures of solid and liquid will be presented, showing the similarities of wave propagation behaviors at micro- and nanoscales.

2:20

4pPA4. Nonlinear surface waves in soil. Evgenia A. Zabolotskaya, Yurii A. Ilinskii, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

Nonlinear effects in surface waves propagating in soil are investigated theoretically. Analytic solutions are derived for the second harmonics and difference frequency waves generated by a bifrequency primary wave propagating at moderate amplitude. The soil is modeled as an isotropic solid. As such, its elastic properties are described by five elastic constants, two at second order in the strain energy density (the shear and bulk moduli) and three at third order. Nonlinear propagation of the surface waves is based on a theory developed previously [Zabolotskaya, J. Acoust. Soc. Am. **91**, 2569–2575 (1992)]. Elements of the nonlinearity matrix associated with the interacting spectral components are expressed in terms of the five elastic constants. It was found convenient to express the nonlinearity matrix for soil as a function of a nonlinearity parameter corresponding to B/A for liquids, particularly for saturated soils exhibiting liquidlike properties. This nonlinearity parameter can vary by several orders of magnitude. For soils with shear wave speeds less than 20% of the compressional wave speeds, the nonlinearity of surfaces waves is found to be independent of the third-order elastic constants and dependent only on the shear modulus. [Work supported by ONR.]

2:35

4pPA5. The measurement of the hysteretic nonlinearity parameter of a field soil by the phase shift method: A long-term survey. Zhiqu Lu (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, Univ., MS 38677)

Soil properties significantly affect the performance of the acoustic landmine detection. The climate and seasonal changes cause the variations of soil properties and smear landmine signature over time. On the other hand, soil is a complicated granular material that exhibits strong nonlinear acoustic behaviors. To understand the weather and seasonal effects on nonlinear acoustic behaviors of soils, a phase shift method is used to measure the hysteretic nonlinearity parameter of a field soil. The technique is based on measuring the variation of phase difference between two transducers, i.e., the phase shift, induced by changing sound level. The hysteretic nonlinear parameter can be extracted from the measured phase shift as a function of sound level or dynamic strain. In a long-term survey, the nonlinearity parameter, sound speed, and environmental conditions such as temperature, moisture, soil water potential, and rainfall precipitation are measured. It is found that the nonlinearity parameter is much more sensitive than sound speed to the climate change. Soil water potential is the predominant factor that affects the nonlinearity parameter and sound speed of the shallow field soil.

2:50

4pPA6. Nonlinear acoustic landmine detection: Comparison of soil nonlinearity with soil-interface nonlinearity. Murray S. Korman, Kathleen E. Pauls, Sean A. Genis (Dept. of Phys., U. S. Naval Acad., Annapolis, MD 21402), and James M. Sabatier (Univ. of Mississippi, Univ., MS 38677)

To model the soil-top plate interface in nonlinear acoustic landmine detection, the soil-plate oscillator was developed [J. Acoust. Soc. Am. **116**, 3354–3369 (2004)]. A Lexan plate (2.39 mm thick, 18.5 cm diameter) is clamped at an inside diameter of 11.8 cm between two metal flanges. Dry sifted masonry sand (2-cm layer) is placed over the plate. Turning curves experiments are performed by driving a loudspeaker (located over the sand) by a swept sinusoid. The acceleration versus frequency is measured near resonance on a swept spectrum analyzer using an accelerometer centered on the surface. The corresponding backbone curve exhibits a linear decrease in resonant frequency f versus increasing acceleration, where $a = -a_o(f-f_o)/f_o$. Define a nonlinear parameter $\alpha = 1/a_o$. When the elastic plate is replaced by a "rigid" plate, α decreased from 0.128 to 0.070 (s^2/m), while f_o increased from 191 to 466 Hz. When a cylindrical drumlike mine simulant (rigid walls, thin acrylic top-plate) was buried 2 cm deep in a concrete sand box, "on the mine" results yielded $\alpha = 0.30$ (s^2/m) with $f_o = 147$ Hz, while "off the mine," $\alpha \sim 0.03$ (s^2/m) at $f_o = 147$ Hz. [Work supported by ONR.]

3:05

4pPA7. Causality conditions and signal propagation in bubbly water. Gregory J. Orris, Dalcio K. Dacol, and Michael Nicholas (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375)

Acoustic propagation through subsurface bubble clouds in the ocean can exhibit signal travel times with enormous variations depending on the acoustic signal frequency, bubble size distribution, and void fraction. Recent theories have predicted large variations in phase speeds and attenuation that have been largely validated for frequencies well below and well above bubble resonance. However, great care must be exercised when theoretically treating signal propagation at frequencies near resonance, termed the "Anomalous Absorption Regime" nearly 100 years ago in the pioneering work of Sommerfeld [A. Sommerfeld, Physik. Z. **8**, 841 (1975)] while investigating aspects of electromagnetic causality. We will discuss similarities between acoustic propagation in bubbly media and electromagnetic propagation in the presence of a conducting medium. We show that the signal travel time is dependent on the behavior of the dispersion formula in the complex frequency plane and place limits on the range of validity of these formulas, leading naturally to the necessary modifications to the current dispersion formulas to bring them into compliance with causality. Finally, we present theoretical results for the velocity of signals for a representative environment of experimental work carried out at the Naval Research Laboratory. [Work supported by the ONR.]

3:20

4pPA8. Measurements of the attenuation and sound speed in bubbly salt water. Gregory J. Orris, Dalcio K. Dacol, and Michael Nicholas (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375)

Bubble clouds were injected from below the surface of a 144-cubic-meter water tank, wherein hydrophones were placed at varying distances from an acoustic source. Measurements were made over a wide range of frequencies to verify and validate the theoretical predictions of the relevant dispersion formula. This work was undertaken under a variety of conditions by varying the relevant environmental parameters: void fraction, temperature, and salinity. Void fractions were varied from roughly 0.02% to 0.1%. Temperatures ranged from 9 °C to 18 °C, and the salinity was varied from zero to approximately 10% of typical oceanic values. Particular attention was paid to tracking the phase of the transmitted signal as the frequency progressed toward resonance starting from 100 kHz. This yielded phase-speed measurements in an essentially free-field environment using a modified version of phase spectral analysis. Time-of-flight measurements gave signal velocities, while the received energy yielded the attenuation. Results are compared to theoretical calculations, leading to the conclusion that current theoretical dispersion formula requires modification. [This work supported by the ONR.]

3:35

4pPA9. Efficient computation of 3-D acoustical scattering from multiple arbitrarily shaped objects using the boundary element method/fast multipole method (BEM/FMM). Nail A. Gumerov and Ramani Duraiswami (Perceptual Interfaces and Reality Lab., Inst. for Adv. Comput. Studies, Univ. of Maryland, College Park, MD 20742)

Many applications require computation of acoustic fields in systems consisting of a large number of scatterers, which may have complex shape. Despite the boundary element method being a well-known technique for solution of the boundary value problems for the Helmholtz equation, its capabilities are usually limited by the memory and speed of computers, and conventional methods can be applicable to relatively small (up to order of 10 000 boundary elements) problems. We developed and implemented an efficient computational technique, based on an iterative solver employing generalized minimal residual method in combination with matrix-vector multiplication speeded up with the fast multipole method. We demonstrate that this technique has $O(N)$ memory and computational complexity and enables solution of problems with thousands of scatterers (millions of boundary elements) on a desktop PC. The test problems solved are of moderate frequency (up to $kD \leqslant 150$, where k is the wavenumber and D is the size of the computational domain). Solution of large scale scattering problems was tested by comparison with the FMM-based T-matrix method applicable for simple shape objects reported earlier [Gumerov and Duraiswami, J. Acoust. Soc. Am., **117**(4), 1744–1761 (2005)], visualization, and physical interpretation of the results.

3:50

4pPA10. Fast acoustic integral-equation solver for complex inhomogeneous media. Elizabeth Bleszynski, Marek Bleszynski, and Thomas Jaroszewicz (Monopole Res., 739 Calle Sequoia, Thousand Oaks, CA 91360, elizabeth@monopoleresearch.com)

We describe elements and representative applications of an integral-equation solver for large-scale computations in acoustic wave propagation problems. In the solver construction we used elements of our previously developed fast integral-equation solver for Maxwell's equations. In comparison with the conventional integral equation approach (method of moments), our solver achieves significant reduction of execution time and memory through the FFT-based matrix compression. One particular aspect of the solver we discuss, pertinent to its high efficiency and accuracy, is an efficient treatment of problems associated with subwavelength discretization. We illustrate the approach and its application on the example of a numerical simulation of acoustic wave propagation through the human head. [Work was supported by a grant from AFOSR.]

4:05

4pPA11. Models for acoustic scattering in high contrast media. Max Denis, Charles Thompson, and Kavitha Chandra (Univ. Massachusetts Lowell, One University Ave., Lowell, MA 01854)

In this work a numerical method for evaluating backscatter from a three-dimensional medium having high acoustic contrast is presented. The solution is sought in terms of a perturbation expansion in the contrast amplitude. It is shown that limitations of the regular perturbation expansion can be overcome by recasting the perturbation sequence as a rational fraction using Padé approximants. The resulting solution allows for an accurate representation of the pressure and allows for the poles in the frequency response to be modeled. The determination of the pulse-echo response for a high-contrast medium is discussed and presented.

4:20

4pPA12. Multiple scattering and visco-thermal effects. Aroune Duclos, Denis Lafarge, Vincent Pagneux (Laboratoire d'Acoustique de l'Universite du Maine, Ave. Olivier Messiaen, 72085 Le Mans, France), and Andrea Cortis (Lawrence Berkeley Natl. Lab., Berkeley, CA 94720)

For modeling sound propagation in a rigid-framed fluid-saturated porous material it is customary to use frequency-dependent density and compressibility functions. These functions, which describe "temporal" dispersion effects due to inertial/viscous and thermal effects, can be computed by FEM in simple geometries and give complete information about the long-wavelength properties of the medium. When the wavelength is reduced, new effects due to scattering must be considered. To study this, we consider solving the sound propagation problem in a 2-D "phononic crystal" made of an infinite square lattice of solid cylinders embedded in a fluid. An exact multiple-scattering solution is first developed for an ideal saturating fluid and then generalized to the case of visco-thermal fluid, by using the concept of visco-thermal admittances. The condition to use this concept is that the viscous and thermal penetration depths are small compared to the cylinder radius. We validate our results in the long-wavelength regime by direct comparisons with FEM data [A. Cortis, "Dynamic parameters of porous media," Ph.D. dissertation (Delft U.P., Delft, (2002)]. When frequency increases, differences appear between the long-wavelength solution and the exact multiple-scattering solution, which could be interpreted in terms of "spatial" dispersion effects.

4:35

4pPA13. Effective parameters of periodic and random distributions of rigid cylinders in air. Daniel Torrent and José Sánchez-Dehesa (Wave Phenomena Group, Nanophotonics Technol. Ctr., Polytechnic Univ. of Valencia, C/Camino de vera s/n., E-46022 Valencia, Spain)

The scattering of sound by finite-size clusters consisting of two-dimensional distributions (periodic and random) of rigid cylinders in air is theoretically studied in the low-frequency limit (homogenization). Analyti-

cal expressions for the effective density and sound speed obtained in the framework of multiple scattering will be reported. For the case of circular-shaped cluster, we have theoretically analyzed the homogenization as a function of the filling fraction, the type of arrangement of the cylinders in the cluster (hexagonal and square lattice), and the number of cylinders in the cluster. When the number of cylinders in the cluster is small we found that for certain "magic numbers" their effective parameters (sound speed and density) are the same as those of the corresponding infinite array. [Work supported by MEC of Spain.]

4:50

4pPA14. The application of *k*-space acoustic propagation models to biomedical photoacoustics. Benjamin T. Cox (Dept. of Med. Phys. and Bioengineering, Univ. College London, Gower St., London, WC1E 6BT, UK, bencox@mpb.ucl.ac.uk), Simon. R. Arridge, and Paul C. Beard (Univ. College London, Gower St., London, WC1E 6BT, UK)

k−space models for broadband acoustic pulse propagation differ from pseudo−spectral time domain (PSTD) models in their treatment of the time step. By replacing a finite−difference scheme with a propagator, exact for homogeneous media, larger time steps can be taken without loss of accuracy or stability and without introducing dispersion. Three k−space models for modeling photoacoustically generated (PA) pulses are described here. A very simple, exact, model of PA propagation in a homogeneous fluid is used to introduce the k−space propagator, and two models of propagation in heterogeneous media, originally designed for modeling scattering in soft tissue, are adapted for use in photoacoustics [Mast *et al.*, IEEE Trans. UFFC **48**, 341−354 (2001); Tabei *et al.*, J. Acoust. Soc. Am. **111**, 53−63 (2002)]. Our motivation for describing these models comes from biomedical PA imaging, in which one of the current limitations is the assumption that soft tissue has a uniform sound speed. Efficient, accurate, and simple−to−encode forward models such as these are very useful for studying the effects of the heterogeneities encountered in practice. They may also be useful in designing PA imaging schemes that can account for acoustic heterogeneities. [This work was funded by the EPSRC, UK]

5:05

4pPA15. Emergence of the acoustic Green's function from thermal noise. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA, Earth System Res. Lab., 325 Broadway, Boulder, CO 80305, oleg.godin@noaa.gov)

Recently proposed applications of noise cross-correlation measurements to passive remote sensing range from ultrasonics and acoustic oceanography to helioseismology and geophysics, at wave frequencies that differ by more than ten orders of magnitude. At the heart of these applications is the possibility to retrieve an estimate of a deterministic Green's function from long-range correlations of diffuse noise fields. Apparently, S. M. Rytov [*A Theory of Electrical Fluctuations and Thermal Radiation* (USSR Academy of Sciences, Moscow, 1953)] was the first to establish theoretically a simple relation between the Green's function and the two-point correlation function of fluctuations of wave fields generated by random sources. He used reciprocity considerations to analyze fluctuations of electromagnetic fields. In this paper, an acoustic counterpart of Rytov's approach is applied to derive exact and asymptotic relations between respective acoustic Green's functions and cross-correlation of thermal noise in inhomogeneous fluid, solid, and fluid-solid media. Parameters of the media are assumed to be time independent, but can be arbitrary functions of spatial coordinates. Theoretical results obtained are compared to those previously reported in the literature.

5:20

4pPA16. Simulation of elastic wave scattering in living tissue at the cellular level. Timothy E. Doyle and Keith H. Warnick (Dept. of Phys., Utah State Univ., 4415 Old Main Hill, Logan, UT 84322-4415, timdoyle@cc.usu.edu)

Elastic wave scattering in biological tissue has been simulated at the cellular level by incorporating a first-order approximation of the cell structure and multiple scattering between cells. The cells were modeled with a concentric spherical shell-core structure embedded in a medium, with the core, shell, and medium representing the cell nucleus, the cell cytoplasm, and the extracellular matrix, respectively. Using vector multipole expansions and boundary conditions, scattering solutions were derived for a single cell with either solid or fluid properties for each of the cell components. Multiple scattering between cells was simulated using addition theorems to translate the multipole fields from cell to cell and using an iterative process to refine the scattering solutions. Backscattering simulations of single cells demonstrated that changes in the nuclear diameter had the greatest effect on the frequency spectra as compared to changes in cell size, density, and shear modulus. Wave field images and spectra from clusters of up to several hundred cells were also simulated, and they exhibited phenomena such as wave field enhancement at the cell membrane and nuclear envelope due to the scattering processes. Relevant applications for these models include ultrasonic tissue characterization and ultrasound-mediated gene transfection and drug delivery.

5:35

4pPA17. Acoustic analog of electronic Bloch oscillations and Zener tunneling. José Sánchez-Dehesa, Helios Sanchis-Alepuz, Yu. A. Kosevich, and Daniel Torrent (Wave Phenomena Group, Polytechnic Univ. of Valencia, C/Camino de Vera s.n., E-46022 Valencia, Spain)

The observation of Bloch oscillations in sound propagation through a multilayer of two different fluidlike components is predicted. In order to obtain the equivalent to the acoustic analog of a Wannier-Stark ladder [E. E. Mendez *et al.*, Phys. Rev. Lett. **60**, 2426–2429 (1988)], a set of cavities with increasing thickness is employed. Bloch oscillations were theoretically predicted as time-resolved oscillations in transmission in direct analogy to electronic Bloch oscillations in semiconductor superlattices [J. Feldmann *et al.*, Phys. Rev. B **46**, R7252–R7255 (1992)]. Finally, an experimental setup is proposed to observe the phenomenon by using arrays of cylindrical rods in air, which acoustically behaves as a fluidlike system with effective sound velocity and density [D. Torrent *et al.*, Phys. Rev. Lett. **96**, 204302 (2006)]. For the proposed system, Bloch oscillations and Zener tunneling are confirmed by using multiple scattering simulations. [Work supported by MEC of Spain.]

5:50

4pPA18. Comparison of time reversal acoustic and prefiltering methods of focusing of tone burst signals. Bok Kyoung Choi (Korea Ocean Res. and Development Inst., Sangrok-gu, 426-744, Korea), Alexander Sutin, and Armen Sarvazyan (Artann Labs., Inc., West Trenton, NJ 08618)

The concept of time reversal acoustics (TRA) provides an elegant possibility of both temporal and spatial concentration of acoustic energy in highly inhomogeneous media. TRA-based focusing is typically used for generation of short acoustic pulses, however, in some medical and industrial applications, longer pulses are required. TRA focusing of longer signals leads to an increase of side lobes in temporal and spatial domains. Another method for focusing, known as prefiltering, is based on measurements of the impulse response, which relates the signal at the TRA transmitter to that at the focusing point. After evaluating the impulse response, the excitation signal may be calculated to generate the desired waveform in the focus point. This method allows signal generation with any desired form including long tone-burst signals. Experiments on comparison TRA and prefiltering methods of ultrasound focusing were conducted in the frequency band of 200–1000 kHz. In the experiments, focused acoustic pulses with various forms and duration were generated: triangular, rectan-

gular, and amplitude-modulated tone burst signals. The prefiltering modes provide better temporal compression of the focused signal, and the signal energy outside the main pulse in the prefiltering mode was shown to be much lower than that in standard TRA focusing.

6:05

4pPA19. Modeling quasi-one-dimensional sound propagation in ducts having two propagation media using a cross-sectional averaging theory. Donald Bliss and Lisa Burton (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708)

Sound propagation of quasi-one-dimensional waves through a uniform duct partially filled with porous material has been studied theoretically and experimentally. The porous material makes the effective propagation wave number in the duct complex. A fairly simple theory based on cross-sectional averaging is derived and tested and found to work extremely well up to fairly high frequency. Interestingly, the basic theory depends only on the ratio of cross-sectional areas and the properties of the individual propagation media, but not on the specific configuration of material in a cross section. A higher order correction is developed to achieve excellent accuracy to very high frequency. This correction includes a coefficient that does depend on the specific cross-sectional configuration. Results are compared to exact solutions for layered and annular configurations, and also to experimental measurements with open cell foam as the porous material. An interesting application is to use measured wave numbers to predict the complex effective density and sound speed of porous media samples partially filling the duct. Other applications include fairly simple improved predictions of the behavior of sound in ducts lined with, or partially filled with, bulk reacting absorbing material.

FRIDAY AFTERNOON, 1 DECEMBER 2006 WAIALUA ROOM, 1:30 TO 4:20 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Auditory Physiology

G. Christopher Stecker, Cochair
Univ. of Washington, Dept. of Speech and Hearing Science, 1417 NE 42nd St., Seattle, WA 98105

Shigeto Furukawa, Cochair
NTT Communication Science Labs., Human and Information Science Lab., 3-1 Morinosato-wakamiya, Atsugi-shi, Kanagawa-ken 243-0198, Japan

Chair's Introduction—1:30

Contributed Papers

1:35

4pPP1. Transmission of bone-conducted sound measured acoustically and psycho-acoustically. Sabine Reinfeldt (Signals & Systems, Chalmers Univ. of Technol., SE-412 96 Goteborg, Sweden), Stefan Stenfelt (Linkoping Univ., SE-581 83 Linkoping, Sweden), and Bo Hakansson (Chalmers Univ. of Technol., SE-412 96 Goteborg, Sweden)

The transcranial transmission is important in the bone-conducted (BC) audiometry where the BC hearing thresholds depend on the stimulation position. It is also important for fitting of BC hearing aids; the transcranial transmission determines the amount of the sound that reaches the contralateral cochlea. Previous reported transcranial transmission results seem to depend on the method used. Here, a comparison between the transcranial transmission measured with BC hearing thresholds and ECSP is performed for both open and occluded ear canal. A BC transducer provided stimulation at both mastoids and the forehead. The ECSP was measured with a probe microphone and the BC hearing thresholds were obtained while masking the nontest ear. The transcranial transmission was determined as the BC hearing threshold or the ECSP for contralateral stimulation relative ipsilateral stimulation. The transmission from the forehead was calculated in a similar way. The transcranial transmission was similar for BC hearing thresholds and ECSP above 800 Hz; this indicates that the ECSP can be used as an estimator of the relative hearing perception by BC. The transcranial transmission results are also similar to vibration measurements of the cochleae made in earlier studies. Hence, vibration measurements of the cochleae can also estimate relative BC hearing.

1:50

4pPP2. Customization of head-related transfer functions using principal components analysis in the time domain. Ki H. Shin and Youngjin Park (Dept. of Mech. Eng., Korea Adv. Inst. of Sci. and Technol. (KAIST), Sci.e Town, Daejeon, 305-701, Republic of Korea)

Pinna responses were separated from HRIRs (head-related impulse responses) of 45 subjects in the CIPIC HRTF (head-related transfer function) database and modeled as linear combinations of five basic temporal shapes (basis functions) by PCA (principal components analysis) accounting for more than 90% of the variance in the original pinna responses per each selected elevation angle in the median plane. By adjusting the weight of each basis function computed for a specific height to replace the pinna response in the KEMAR HRIR at the same height with the resulting pinna response and listening to the filtered stimuli over headphones, four subjects were able to create a set of median HRIRs that outperformed the KEMAR HRIRs in producing elevation effects in the median plane. Since the monoaural spectral features due to the pinna are strongly dependent on elevation instead of azimuth, similar elevation effects could also be generated at different azimuthal positions simply by inserting the customized median pinna responses into the KEMAR HRIRs at other azimuths and varying the ITD (interaural time difference) according to the direction as well as the size of the subject's own head.

2:05

4pPP3. An electrophysiological measure of basilar membrane nonlinearity in humans. Christopher J. Plack (Dept. of Psych., Lancaster Univ., Lancaster, LA1 4YF, England) and Ananthanarayan Krishnan (Purdue Univ., West Lafayette, IN 47906)

A behavioral measure of the basilar membrane response can be obtained by comparing the growth in forward masking for maskers at, and well below, the signal frequency. Since the off-frequency masker is assumed to be processed linearly at the signal place, the difference in masking growth with level is thought to reflect the compressive response to the on-frequency masker. The present experiment used an electrophysiological analog of this technique, based on measurements of the latency of wave V of the auditory brainstem response elicited by a 4-kHz, 4-ms pure tone, presented at 65 dB SPL. Responses were obtained in quiet and in the presence of either an on-frequency or an off-frequency (1.8 kHz) pure-tone forward masker. Wave V latency increased with masker level, although the increase was greater for the off-frequency masker than for the on-frequency masker, consistent with a more compressive response to the latter. Response functions generated from the data showed the characteristic shape, with a nearly linear response at lower levels, and 5:1 compression at higher levels. However, the breakpoint between the linear region and the compressive region was at about 60 dB SPL, higher than expected on the basis of previous physiological and psychophysical measures.

2:20

4pPP4. Possible involvement of the spiral limbus in chinchilla cochlear mechanics. William S. Rhode (Dept. of Physiol., 1300 University Ave., Madison, WI 53706, rhode@physiology.wisc.edu)

Differences between cochlear mechanical tuning curves and those of auditory nerve fibers (ANFs) exist. In particular, mechanical transfer functions exhibit a high-frequency plateau; ANFs frequency threshold curves (FTCs) do not. ANF-FTCs may have a low-frequency slope due to a velocity forcing function operating on inner hair cells at low frequencies. Neither basilar membrane velocity nor displacement adequately explain the entire ANF tuning curve. A displacement sensitive interferometer was used to study basilar membrane and spiral limbus mechanics in the 6-kHz region of the chinchilla cochlea. The spiral limbus vibrates at the same phase as the basilar membrane nearly up to the location's characteristic frequency. In the plateau region, the limbus appears to vibrate 0 to 20 dB less than the basilar membrane. The basilar membrane/limbus amplitude transfer function has a low-frequency slope of ~3 dB/oct at low frequencies and is ~10 dB lower than the basilar membrane amplitude at 1 kHz. It appears that spiral limbus vibration may contribute to the excitation of the cilia of the inner hair cells. [Work supported by NIDCD grant R01 DC001910.]

2:35

4pPP5. The effects of antioxidants on cochlear mechanics. Barbara Acker-Mills, Melinda Hill, and Angeline Ebuen (U.S. Army Aeromedical Res. Lab., P.O. Box 620577, Fort Rucker, AL 36362-0577, barbara.acker@us.army.mil)

Several studies have evaluated the effect of N-acetylcysteine (NAC) on temporary threshold shifts (TTSs) in humans. Work at USAARL found that NAC did not reduce TTSs compared to a placebo, but suppressed otoacoustic emissions (OAEs) more than a placebo, indicating that NAC may reduce outer hair cell activity. Kramer *et al.* [JAAA, **17**(4), (2006)] found similar results, where NAC did not affect thresholds in people who had TTSs from exposure to loud music. However, OAEs also did not differ between NAC and placebo. Toppilla *et al.* [XXII Barany Society Meeting (2002)] measured thresholds and balance in people exposed to loud music and found that while NAC did not affect TTS, it reduced noise-induced balance impairment. The current study administered NAC and measured cochlear microphonics, compound action potentials, and OAEs to evaluate cochlear function. The vestibular myogenic potential was measured to assess the effect of NAC on the saccule. The results provide a comprehensive analysis of the effect of NAC on the auditory system and one component of the vestibular system. [Work supported by the U.S. Army ILIR program. The work is that of the authors and is not necessarily endorsed by the U.S. Army or the Department of Defense.]

2:50–3:05 Break

3:05

4pPP6. Time characteristics of distortion product otoacoustic emissions recovery function after moderate sound exposure. Miguel Angel Aranda de Toro, Rodrigo Ordoñez, and Dorte Hammershøi (Dept. of Acoust., Aalborg Univ., Fredrik Bajers Vej 7 B5, DK-9220, Aalborg, Denmark, maat@acoustics.aau.dk)

Exposure to sound of moderate level temporarily attenuates the amplitude of distortion product otoacoustic emissions (DPOAEs). These changes are similar to the changes observed in absolute hearing thresholds after similar sound exposures. To be able to assess changes over time across a broad frequency range, a detailed model of the recovery time characteristics is necessary. In the present study, the methodological aspects needed in order to monitor changes in DPOAEs from human subjects measured with high time resolution are presented. The issues treated are (1) time resolution of the measurements, (2) number of frequency points required, and (3) effects in fine structures, are they affected with the exposure? [Work supported by the Danish Research Council for Technology and Production.]

3:20

4pPP7. Probability characteristics of neural coincidence detectors in the brainstem. Ram Krips and Miriam Furst (Dept. of Elec. Eng. Systems, Faculty of Eng., Tel Aviv Univ., Tel Aviv 69978, Israel, mira@eng.tau.ac.il)

Auditory neural activity in the periphery is usually described as nonhomogeneous Poisson process (NHPP), characterized as either EE or EI, which operates as a coincidence detector. In order to apply a general probabilistic analysis of those brainstem nuclei activity, the stochastic properties of the axons that exit the EE and EI nuclei is essential. An analytical analysis of the probability characteristics of the output of an EE nucleus (EEout) will be presented. Assuming that an EE nucleus receives inputs from two neurons, each behaves as an NHPP with instantaneous rates λ_1 and λ_2, and an output spike is generated if both spike at a coincidence window (Δ) which is relatively small (this matches biological findings). Then EEout is also a NHPP with instantaneous rate $r(t) = \lambda_1(t)\int_{t-\Delta}^{t}\lambda_2(\zeta)\,d\zeta + \lambda_2(t)\int_{t-\Delta}^{t}\lambda_1(\zeta)\,d\zeta$. A similar derivation was applied for an EI nucleus. We found also that the output activity is NHPP for a relatively small coincidence window (Δ). The obtained IR is $r(t) = \lambda_e(t)[1-\int_{t-\Delta}^{t}\lambda_i(\zeta)\,d\zeta]$, where λ_E and λ_I are the excitatory and inhibitory input IRs. On the other hand, for larger Δ, the output activity is not a Poisson process. Those derivations enable theoretical analysis and performance evaluation of large neural networks, which perform binaural tasks as ITD and ILD.

3:35

4pPP8. Musical expertise and concurrent sound segregation. Benjamin Rich Zendel and Claude Alain (Rotman Res. Inst., Baycrest Ctr. & Dept. of Psych., Univ. of Toronto, Toronto, ON M6A 2E1, Canada)

There is growing evidence suggesting that musical training can improve performance in various auditory perceptual tasks. These improvements can be paralleled by changes in scalp recorded auditory evoked potentials (AEPs). The present study examined whether musical training modulates the ability to segregate concurrent auditory objects using behavioral measures and AEPs. Expert musicians and nonmusicians were presented with complex sounds comprised of six harmonics (220, 440, 660 Hz, etc.). The third harmonic was either tuned or mistuned by 1%–16% of its original value. Mistuning a component of a harmonic complex results in the precept of a second auditory object. Stimuli were presented passively (no response) and actively (participants responded by indicating if they heard one sound or two sounds). Behaviorally both musicians and nonmusicians perceived a second auditory object at similar levels of mistuning. In both groups, complex sounds generated N1 and P2 waves at fronto-central scalp regions. The perception of concurrent auditory objects was paralleled by an increased negativity around 150 ms post-stimulus onset. This increased negativity is referred to as object-related negativity (ORN). Small differences between musicians and nonmusicians were noted in the ORN. The implication of these results will be discussed in terms of current auditory scene analysis theory.

3:50

4pPP9. Study of auditory temporal processing of language and music in stroke patients. Li Hsieh and Tamara Baubie (Dept. of Commun. Sci. and Disord., Wayne State Univ., 207 Rackham Hall, 60 Farnsworth, Detroit, MI 48202)

This study focused on functional lateralization of temporal processing of language and music for nine stroke patients and nine normal controls. Subjects were asked to discriminate short versus long sounds in ABX tasks, and then to reproduce sequences of short and long sounds they heard. The reproduction tasks consisted of sequences of 3, 4, and 5 sounds. The linguistic stimuli were nonsense CVC syllables, and the musical stimuli were computer-generated musical notes with the timbres of French horn and trombone. Both linguistic and music stimuli were controlled for frequency and intensity, and varied only for duration (i.e., 500 and 750 ms). Our findings are consistent with previous studies; the left hemisphere specializes in linguistics, while the right hemisphere in music. Moreover, both hemispheres appeared to work closely together in processing temporal information. Both left- and right-hemisphere-damaged patients performed worse than controls. Most subjects performed better with music than language. Patients with left posterior lesions performed worse compared to patients with left or right anterior lesions, particularly with linguistic stimuli. Patients with right anterior lesions not only involved with temporal processing in music, but also in linguistic stimuli. Our study provided additional information regarding transient temporal processing in language and music.

4:05

4pPP10. Human bioacoustic biology: Acoustically anomalous vocal patterns used to detect biometric expressions relating to structural integrity and states of health. Sharry Edwards (Sound Health Res. Inst., P.O. Box 267, Sauber Res. Ctr., Albany, OH 45710)

Computerized analyses of acoustically anomalous vocal patterns are being used as biomarkers for predictive, prediagnostic, and efficient management of individual biological form and function. To date, biometrically distinct vocal data have resulted in outcomes that would be considered improbable by contemporary medical standards. For instance, independent EMG conclusions confirmed the regeneration of nerve tissue for a multiple sclerosis patient who used acoustic bioinformation to guide his primary therapy. Another study monitored the amounts of synthetic labor hormones present during induced labor. False labor costs amount to millions of dollars each year in insurance and hospital resources. The use of noninvasive, possibly remote, vocal profiling could ameliorate such costs. Anomalous vocal acoustics are being investigated by many health-related organizations including Pfizer Pharmaceuticals and the Institute of Automatic Control Engineering in Taiwan. Complementary research studying molecular frequencies of cellular chemistry is being done by James Gimjewski, Ph.D., UCLA, Department of Chemistry and Biochemistry. Known as BioAcoustic Biology, this research modality has the potential to advance current health care standards for biological function, disease processes, and metabolism. Organizations such as the Acoustical Society of America are considering standards for technically defining human bioacoustics. This paper would expand that language.

FRIDAY AFTERNOON, 1 DECEMBER 2006 HONOLULU ROOM, 1:00 TO 3:20 P.M.

Session 4pSAa

Structural Acoustics and Vibration: Vehicle Interior Noise and Vibration

Courtney B. Burroughs, Cochair
Noise Control Engineering, Inc. 1241 Smithfield St., State College, PA 16801

Hiroaki Morimura, Cochair
Japan Science and Technology Agency, Hosei Univ., Dept. of Mechanical Engineering, 3-7-2 Kajino-cho, Kogamei-city, Tokyo, Japan

Invited Papers

1:00

4pSAa1. Analytical model for flow-excited interior cavity resonance and its application to the Stratospheric Observatory for Infrared Astronomy (SOFIA). Jerry H. Ginsberg (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The Stratospheric Observatory for Infrared Astronomy (SOFIA) is a joint effort between NASA and the German Space Agency that has installed a 20 000-kg telescope in a modified 747-SP. The modifications entailed constructing bulkheads, one of which is used to provide the active mount for the telescope, and a door that rotates to open as much as one-quarter of the fuselage circumference to the atmosphere. This configuration represents a large compartment that can exhibit acoustic resonances at low frequencies. Concern arose that a Rossiter mode, which is an aerodynamic resonance in which vortices shed from the leading edge of a gap form a coherent standing pattern at certain speeds, would create a strong acoustic source for acoustic and structural modes, whose frequencies might coincide. A model consisting of a two-dimensional hard-walled waveguide having a Rossiter mode source and an elastic plate at one end was analyzed in order to understand these issues. Unlike conventional analyses of interior cabin noise, in which vibrating walls are the acoustic source, the elastic plate represents a compliant boundary that couples with the acoustic modes. The results lead to some general insights to the the possibility of a ‘‘triple resonance,’’ as well as the role of structural compliance for cavities that are excited by turbulent external flows.

1:20

4pSAa2. Energy finite energy analysis for shipboard noise. Raymond Fischer, Leo Boroditsky (Noise Control Eng. Inc., 799 Middlesex Tnpk, Billerica, MA 01821), Layton Gilfroy (DRDC-Atlantic), and David Brennan (Martec Ltd.)

Machinery-induced habitability noise is difficult to model efficiently and accurately. The potential of energy finite-element analysis (EFEA) is compared to other prediction tools such as statistical energy analysis (SEA). This paper will explore the benefits and costs of EFEA with respect to SEA for acoustic modeling. The focus will be on issues relating to structural modeling for EFEA purposes. EFEA techniques will be evaluated to see if they possess the capabilities of verified SEA approaches for predicting habitability and radiated noise, where it is necessary to account for the impact of diverse marine constructions and sources such as the lack of machinery source information with respect to force or moment inputs or the finite impedance of machinery foundations. The effort proposed herein will provide the necessary engineering to research and identify salient features of EFEA that are potentially applicable for the detailed analysis of the acoustic environment and response of surface ships to various excitation sources. The paper will also address the pros and cons of SEA versus energy-finite element analysis (EFEA) methods used to predict the habitability noise of surface ship platforms. [This work is supported by an Office of Navy Research contract.]

1:40

4pSAa3. Spectral-based multicoordinate substructuring model for vehicle noise, vibration, and harshness refinement. Teik C. Lim (Mech., Industrial and Nuclear Eng., 598 Rhodes Hall, Box 210072, Univ. of Cincinnati, Cincinnati, OH 45221, teik.lim@uc.edu)

The success of vehicle NVH (noise, vibration, and harshness) refinement often depends on the ability to identify and understand noise and vibration transmission paths within the mid-frequency range, i.e., 200–1000 Hz, throughout the assembled structure. Due to the complexity of the dynamics in this frequency range, most modal or finite element-based methods do not possess the fidelity needed. To address this gap, a multicoordinate substructuring theory applying measured structural-acoustic and vibration spectra is applied. Three forms of substructuring formulation, namely the nondiagonal, block-diagonal, and purely diagonal coupling cases, are developed. The performances of these approaches are studied numerically, and the net effects of these coupling formulations on the predicted joint and free substructure dynamic characteristics, and system response, are determined. Conditions for applying the simpler coupler that can simplify the testing process and overcome computational deficiencies are also derived. When the measured data is noise contaminated, the singular value decomposition (SVD) algorithm is found to be quite helpful. Using an actual vehicle, a comprehensive analysis of the measured and predicted vehicle system responses is performed. The results are employed to develop an understanding of the primary controlling factors and transfer paths and to cascade system requirements to the substructure level.

2:00

4pSAa4. Practical application of digital simulation with physical test in vehicle virtual. Toshiro Abe (ESTECH Corp., 89-1 Yamashita-cho, Naka-ku, Yokohama-shi, Kanagawa-ken, Japan 231-0023, toshiro.abe@estech.co.jp)

In the current vehicle design and development program, the Virtual Product Development process (hereafter, VPD process) is the innovation for automotive industry, which improves product quality and shortens time to market. In general, valid CAE applications are the key component of the VPD process as well as physical tests being indispensable to create valid simulation technologies. This presentation explains how physical-test-based CAE leverages the VPD process. In particular, the presentation is based on how physical testing supports the VPD process and why the ESTECH philosophy is that "The essence of CAE lies in its synergy with Testing;" a philosophy that differentiates the company from the competition. To demonstrate these activities, case studies based on automotive dynamic and real time simulation will be presented. In the case studies, vehicle body NVH and brake noise analysis will be used to show the interaction between physical test and computer simulation. Finally, practical application of the VPD process in one of the leading Japanese automotive companies will be shown, where the effectiveness of the front loading in the actual vehicle development program and the actual deployment of VPD process to the Functional Digital Vehicle project in the powertrain design are to be presented.

Contributed Papers

2:20

4pSAa5. Active noise control simulations using minimization of energy density in a mock helicopter cabin. Jared Thomas, Stephan P. Lovstedt, Jonathan Blotter, Scott D. Sommerfeldt (Brigham Young Univ., Provo, UT 84602), Norman S. Serrano, and Allan Egelston (Silver State Helicopters, Provo, UT 84601)

Helicopter cabin noise is dominated by low-frequency tonal noise, making it an ideal candidate for active noise control. Previous work in active control of cabin noise suggests an energy density approach to be a good solution [B. Faber and S.D. Sommerfeldt, Global Control in a Mock Tractor Cabin Using Energy Density, Proc. ACTIVE 04, Sept. 2004.] Simulations for active noise control using energy density minimization have been made using recorded data from a Robinson R44 helicopter. Initial computer models show substantial noise reductions in excess of 6 dB at the error sensor are possible. Performance results for computer models and simulations in a mock cab for different control arrangements will be compared.

2:35

4pSAa6. Modeling airborne interior noise in full vehicles using statistical energy analysis. Arnaud Charpentier and Phil Shorter (ESI, 12555 High Bluff Dr., Ste. 250, San Diego, CA 92130, arnaud.charpentier@esi-group-na.com)

SEA is particularly well suited for predicting airborne noise in vehicles. The acoustic sources found in such environment are typically spatially distributed around the vehicle and can be well represented with SEA diffuse acoustic loads. Multiple transmission paths contribute to interior noise levels including (1) mass law transmission through trimmed panels, (2) resonant radiation from vibrating structures, and (3) flanking paths through gaskets, seals, and holes. All these transmission mechanisms may be modeled using SEA techniques. Finally, interior trim (including carpet, headliner, seats) is a key contributor to the acoustic performance of modern vehicles. The vehicle sound package has a significant impact on both the strength of the transmissions paths into the vehicle as well as the acoustic absorption in the cabin. Both these effects can be accounted for with SEA through detailed models of the trim. SEA models of full vehicles are usually validated against experimental results at both component and system levels. The models can then be confidently used to (a) rank key design parameters governing interior levels and (b) quickly evaluate the impact of potential design changes. Example vehicle models and correlation results are presented here.

2:50

4pSAa7. Radiation from vibrating panels at high frequency including an inquiry into the role of edges and drive points. Donald Bliss (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708)

In the high frequency limit, a vibrating finite panel is shown to have broadband power and directivity characteristics that can expressed analytically by a limited set of parameters. Two-dimensional problems with subsonic structural wave speed are considered. Three basic directivity patterns are identified, associated with right and left traveling waves and the correlation between them. The role of boundary conditions at the panel edge is illustrated, as are the effects of types of forcing. Overall, relatively simple broadband behaviors are revealed. The analytical characterization of the radiation is shown to be particularly straightforward in the high frequency broadband limit. Interestingly, the radiated mean-square pressures are independent of the panel length, indicating the radiation is associated with the edges and the drive point. However, the radiation patterns cannot be explained in terms of simple volumetric sources placed just at the edges and the drive point, showing that the often-stated idea of uncanceled volumetric sources at these locations is not correct except under very restricted circumstances. A correct physical interpretation of the radiation is provided both in physical space and in terms of spatial Fourier transforms.

3:05

4pSAa8. Some effect of trim and body panels on the low-frequency interior noise in vehicles. Andrzej Pietrzyk (Volvo Car Corp., Noise and Vib. Ctr., 405 31 Gothenburg, Sweden)

Structure borne noise dominates the interior noise in vehicles at low frequencies. One of the basic vinroacoustic characteristics of the trimmed body is the noise transfer function, i.e., the acoustic response at a selected position in the passenger compartment, e.g., driver's ear due to a mechanical excitation at a selected body mount. Detailed CAE models based on the FE method are today available for calculating this characteristic at low frequencies, corresponding to the engine idling and road excitation. However, the accuracy of CAE predictions of interior noise is still considered insufficient for the so-called analytical sign off, i.e., zero-prototypes vision. The current paper describes some investigations into the contribution of individual body panels to the overall interior noise. Also, the effect of selected interior trim items on the area investigated. Relative errors of prediction at different trim levels and different frequency ranges are discussed. Both experimental and CAE results are provided. The aim is to better understand the way the interior noise in vehicles is created, and how it can be controlled.

FRIDAY AFTERNOON, 1 DECEMBER 2006 HONOLULU ROOM, 3:45 TO 5:45 P.M.

Session 4pSAb

Structural Acoustics and Vibration: General Vibration and Measurement Technology

Peter C. Herdic, Cochair
Naval Research Lab., Physical Acoustics Branch, Code 7136, 4555 Overlook Ave., SW, Washington, DC 20375

Naoya Kojima, Cochair
Yamaguchi Univ., Dept. of Mechanical Engineering, 2-16-1 Tokiwadai, Ube, Yamaguchi 755-8611, Japan

Contributed Papers

3:45

4pSAb1. Direct computation of degenerate elastodynamic solution of elastic wave propagation in a thick plate. Jamal Ghorieshi (Eng. Dept., Wilkes Univ., Wilkes-Barre, PA 18766)

The limiting form of elastodynamic solutions as frequency tends to zero leads to the elastostatic eigenvalue equations. However, this limiting procedure is not convenient. It is cumbersome when applied to the solutions obtained using Stokes potentials and, in the case of utilizing Lames potentials, it does not produce static solutions that are a function of position alone. In this paper it is shown that the exact solutions of elastostatic problems can, in general, be obtained in a straightforward manner by the use of harmonic potentials without recourse to any special limiting form of analysis. This method is applied to an infinite, elastic thick plate with traction-free parallel surfaces and the elastostatic eigenvalue equation. It is shown that the problem can be solved exactly in terms of harmonic functions, one of which is a scalar and the other one is a vector. It is noted that results are in agreement with the published solutions.

4:00

4pSAb2. Wave propagation characteristics of an infinite fluid-loaded periodic plate. Abhijit Sarkar and Venkata R. Sonti (Indian Inst. of Sci. Mech. Eng., IISc., Bangalore-560012, India)

A 1-D infinite periodic plate with simple supports placed along equidistant parallel lines is considered using the finite-element method. The plate is loaded with a finite-height fluid column covered on the top with a rigid plate. Results show a relation between the propagation constant of the fluid-loaded structure with its *in vacuo* counterpart. Since the acoustic medium is an additional wave carrier, the attenuation bands corresponding to the *in vacuo* structure turn out to be propagating. However, the presence of the fluid can also bring about attenuation regions within the *in vacuo* propagation bands. Primary propagation constants bring additional waves called space harmonics with them. Hence, a localized coincidence effect is seen where a particular harmonic falls below or above the acoustic wave number, leading to propagation or a mass loading effect. Occasionally, a complete attenuation band is created. This is verified by decomposing the single span displacement profile into the space harmonics and also by computing the frequency response function (FRF) for a finite fluid-loaded periodic plate and observing the huge antiresonance dip in frequency in the exact same frequency band where an attenuation band was predicted for the infinite structure.

4:15

4pSAb3. The dynamic response of a plate subjected to various edge excitations. Baruch Karp (Faculty of Aerosp. Eng., Technion Israel Inst. of Technol., Haifa 32000, Israel)

Plane strain response of a semi-infinite, elastic plate to harmonic edge excitation is investigated analytically. The exact solution to this problem is obtained as a series expansion of the Rayleigh-Lamb modes of the plate. The variation of energy partition among the propagating modes with frequency of the edge excitation was found for load and displacement (symmetrical) perturbations of uniform and cosine form. The biorthogonality relation was employed in deriving the relative amplitudes of each mode to the given perturbation. The emphasis here is on the sensitivity of the far-field response, represented by propagating waves, to the details of the excitation. Within the frequency range investigated it was found that the plate's response is remarkably insensitive to whether the excitation is load or displacement type. The two types of edge excitation distributions, on the other hand, result in different patterns of energy partition above the first cutoff frequency with similar energy partition only within limited range of frequencies. The effect of the nature of the excitation on the dynamic response of the plate and a possible implication to dynamic equivalence is discussed.

4:30

4pSAb4. Measurement of structural intensity using patch near-field acoustical holography. Kenji Saijyou (Fifth Res. Ctr., TRDI, Japan Defense Agency, 3-13-1 Nagase, Yokosuka City, Kanagawa Prefecture, Japan, 239-0826)

Measurement of power flow in a structure, called the structural intensity (SI), is essential for vibration control and noise reduction. The near-field acoustical holography (NAH)-based measurement method is suitable to analyze the interrelationship between SI and acoustic intensity (AI) because NAH-based methods provide measurements of SI and AI simultaneously. Use of NAH requires the measurement of a pressure field over a complete surface located exterior to the structure. Therefore, if the measurement aperture is smaller than the structure, reconstructed results from the pressure on the finite aperture are seriously contaminated. This finite aperture effect prevents implementation of this NAH-based method on an actual large-scale structure such as a ship. Patch NAH and regularization method for SI measurement are applied to overcome this difficulty. This method enables implementation of the NAH-based SI measurement method in a large-scale structure. The effectiveness of this method is demonstrated by experiment.

4:45

4pSAb5. Flexural component and extensional component of vibration energy in shell structure. Taito Ogushi, Manabu Yahara, Masato Mikami, and Naoya Kojima (Dept. of Mech. Eng., Yamaguchi Univ., 2-16-1 Tokiwadai, Ube, Yamaguchi 755-8611, Japan, j008ve@yamaguchi-u.ac.jp)

In this research, the behavior of the flexural component and the extensional component of vibration intensity and their transmission in curved shells are presented. L-shaped shell model was employed as an analysis model of FEM. As FEM analysis methods, both the frequency response analysis and the transitional response analysis were employed. The flexural component and the extensional component of vibration intensity (VI) were calculated by the results of FEM analysis. In the flexural component of the VI, the vibration energy supplied in the flat part decreased at the boundary from the flat part to the curved part and VI vectors flew in circumferential direction in the curved part. In the extensional component of the VI, the vibration energy appeared at the boundary from the flat part to the curved part and most VI vectors flew parallel to the shell axis in the curved part. The total vibration energy of the flexural component and the extensional component was conserved. So, the vibration energy transformed to each other between the flexural component and the extensional component in L-shaped shell.

5:00

4pSAb6. Seismic/acoustic detection of ground and air traffic for unattended ground sensor technology. Peter C. Herdic (Naval Res. Lab., Physical Acoust. Branch, Washington, DC 20375 and SFA Inc., Crofton, MD), Brian H. Houston, Phillip A. Frank, and Robert M. Baden (Naval Res. Lab., Washington, DC 20375)

Human footfall and vehicle traffic create surface waves in soil media that can easily be detected by seismic sensors. Field measurement data have been acquired with a triaxial geophone at several experimental sites. The in-plane-surface wave components dominate the response and decay at a rate of approximately $1/R$, where R is distance. This decay rate is due to the combined effect of spreading $(1/\mathrm{sqrt}(R))$ and damping losses in the soil. Further, the detection range is dependent upon the level of environmental noise, soil compliance, moisture content, and topography. Human detection was achieved in rural environments at distances up to $\sim 30-40$ m, and vehicle detection was possible at much greater distances. Seismic signals due to aircraft are small when compared to the acoustic signature. Ground-based microphone measurements clearly show the blade passage frequency tones of propeller airplanes and the broader band signature of turbojet aircraft. Time- and frequency-domain signal-processing methods for the detection and identification will also be introduced. These experimental results will be discussed with particular emphasis placed on wave phenomenon, detection and identification algorithms, and the related physics.

5:15

4pSAb7. Modeling of acoustic and elastic wave phenomena using plane wave basis functions. Tomi Huttunen, Jari P. Kaipio (Dept. of Phys., Univ. of Kuopio, P.O. Box 1627, FI-70211 Kuopio, Finland), and Peter Monk (Univ. of Delaware, Newark, DE 19716)

When simulating acoustic, elastodynamic, or coupled fluid-solid vibration problems using standard finite element (FE) techniques, several elements per wavelength are needed to obtain a tolerable accuracy for engineering purposes. At high wave numbers, the requirement of dense meshes may lead to an overwhelmingly large computational burden, which significantly limits the feasibility of FE methods for the modeling of wave phenomena. A promising technique for reducing the computational complexity is to use plane wave basis functions opposed to the low-order polynomials that are used in conventional FE methods. A possible method for utilizing the plane wave basis is the ultra-weak variational formulation (UWVF). The UWVF method can be used for acoustic Helmholtz problems, elastodynamic Navier problems, or fluid-solid systems characterized by a coupled Helmholtz-Navier model. A comparison of the UWVF technique with a low-order FE method shows reduced computational complexity and improved accuracy.

Contributed Poster Paper

Poster paper 4pSAb8 will be on display from 1:00 p.m. to 5:45 p.m. The author will be at the poster from 5:30 p.m. to 5:45 p.m.

4pSAb8. Application of vibration energy flow to evaluation of thickness. Akihiko Higashi (Dept. of Maritime Sci. and Technol., Japan Coast Guard Acad., 5-1 Wakaba-cho, Kure, Hiroshima, 737-8512, Japan)

In this study, the possibility of the useful application of the vibration energy flow is investigated. The vibration energy flow means the propagation of the vibration energy of the flexural waves. The vibration energy flow is expressed by the structural intensity. Here, it is easy to input the flexural waves in the thin plates and beam elements. Then, large structures such as ships use many of these thin plates and beam elements. But the usual methods of the evaluation and the inspection of these large structures are inefficient. Then, we investigated the possibility of the evaluation of the changed thickness of the structure by using the vibration energy flow analysis. As the result of analysis, the structural intensity suddenly changes at the position of the changed thickness. The changed quantity of the structural intensity corresponds to the change quantity of the thickness. Then, the evaluation method of the thickness of the structure is proposed. As a result, it was found that the change of the structural intensity indicates the change of the thickness. And the evaluation of the change of thickness of beams could be possible by using the proposed method.

FRIDAY AFTERNOON, 1 DECEMBER 2006 MOLOKAI ROOM, 1:00 TO 4:00 P.M.

Session 4pSC

Speech Communication: Variation in Production and Perception of Speech (Poster Session)

Heriberto Avelino, Cochair
Univ. of California—Berkeley, Dept. of Linguistics, 1203 Dwinelle Hall, Berkeley, CA 94720-2650

Haruo Kubozono, Cochair
Kobe Univ., Dept. of Linguistics, Faculty of Letters, Nada-ku, Kobe 657-8501, Japan

Contributed Papers

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

4pSC1. Cross-language perception of voice and affect. Christer Gobl, Irena Yanushevskaya, and Ailbhe N. Chasaide (Phonet. and Speech Lab., School of Linguistic, Speech and Commun. Sci., Trinity College Dublin, Dublin 2, Ireland, yanushei@tcd.ie)

The paper reports on a cross-language study of how voice quality and $f0$ combine in the signaling of affect. Speakers of Irish-English and Japanese participated in perception tests. The stimuli consisted of a short utterance where $f0$ and voice source parameters were varied using the LF-model implementation of the KLSyn88a formant synthesizer, and were of three types: (1) VQ only involving voice quality variations and a neutral $f0$ contour; (2) $f0$ only, with different affect-related $f0$ contours and modal voice; (3) VQ+$f0$ stimuli, where the voice qualities of (1) combine with specific $f0$ contours from (2). Overall, stimuli involving voice quality variation were consistently associated with affect. In (2) only stimuli with high $f0$ yielded high affective ratings. Striking differences emerge between the ratings obtained from the two language groups. The results show that not only were some affects consistently perceived by one language group and not the other, but also that specific voice qualities and pitch contours were associated with very different affects across the two groups. The results have important implications for expressive speech synthesis, indicating that language/culture-specific differences need to be considered. [This work is supported by the EU-funded Network of Excellence on Emotion, HUMAINE.]

4pSC2. An articulatory study of coronal consonants in Arrernte. Marija Tabain (LaTrobe Univ., Melbourne, Australia), Richard Beare (Monash Univ., Melbourne, Australia), Catherine Best (Univ. of Western Sydney, Sydney, Australia), and Louis Goldstein (Haskins Labs., CT)

This paper presents electro-magnetic articulography (EMA) data on the four coronal stops of Arrernte, an Australian language. The stops are: the lamino-dental ‘‘th,’’ the apico-alveolar ‘‘t,’’ the apico-postalveolar (or ‘‘retroflex’’) ‘‘rt,’’ and the lamino-palatal ‘‘ty.’’ Jaw, tongue tip (TT), and tongue body (TB) data were collected for two female speakers of the language. Results for the first speaker show a fronted tongue position for the laminal consonants, with the TT reflecting a similar location for both the dental and the palatal. However, for the palatal, the TB position is much higher, whereas for the dental, the TB is very low. For the apical consonants, the TT is not as far forward, and the TB is not quite as high as for the lamino-palatal. For both TT and TB, apico-postalveolar is further back than apico-alveolar. For the second speaker, the TT sensor failed, but in line with the first speaker, the TB sensor showed a higher position for the palatal. The other stops were lower and more forward, with the postalveolar TB position higher than the laminal or alveolar stop position. For both speakers, the jaw position is lowest for the postalveolar. [Work supported by Australian Research Council and NIH: NIDCD.]

4pSC3. Symbolic phonetic features for pronunciation modeling. Rebecca A. Bates,[a)], Mari Ostendorf (Dept. of Elec. Eng., Univ. of Washington, Box 352500, Seattle, WA 98195), and Richard A. Wright (Univ. of Washington, Seattle, WA 98195)

A significant source of variation in spontaneous speech is due to intraspeaker pronunciation changes, often realized as small feature changes, e.g., nasalized vowels or affricated stops, rather than full phone transformations. Previous computational modeling of pronunciation variation has typically involved transformations from one phone to another, partly because most speech processing systems use phone-based units. Here, a phonetic-feature-based prediction model is presented where phones are represented by a vector of symbolic features that can be on, off, unspecified, or unused. Feature interaction is examined using different groupings of possibly dependent features, and a hierarchical grouping with conditional dependencies led to the best results. Feature-based models are shown to be more efficient than phone-based models, in the sense of requiring fewer parameters to predict variation while giving smaller distance and perplexity values when comparing predictions to the hand-labeled reference. A parsimonious model is better suited to incorporating new conditioning factors, and this work investigates high-level information sources, including both text (syntax, discourse) and prosody cues. Detailed results are under review with *Speech Communication*. [This research was supported in part by the NSF, Award No. IIS-9618926, an Intel Ph.D. Fellowship, and by a faculty improvement grant from Minnesota State University Mankato.] [a)]Currently at Minnesota State University, Mankato.

4pSC4. Acoustic phonetic variability and auditory word recognition by dyslexic and nondyslexic children. Patricia Keating, Kuniko Nielsen (Phonet. Lab., Linguist., UCLA, Los Angeles, CA 90095-1543, keating@humnet.ucla.edu), Frank Manis, and Jennifer Bruno (USC, Los Angeles, CA 90089)

The hypothesis that dyslexia involves degraded phonological representations predicts impairments in behaviors that rely on these representations, such as auditory word recognition. Normal adult listeners recognize different pronunciations of a word as instances of the same lexical item, but more slowly and less accurately; dylexics should be even more impaired by acoustic phonetic variability. Children with and without dyslexia performed a word recognition task: on each trial, a child hears a target word, then eight probes (matching the target or not), responding yes/no to each probe. On some trials, probes are spoken by multiple talkers who differ in age, sex, speech style, etc.; on some trials the match probes also differ from the target in final stop consonant allophone. Responses are scored for accuracy and speed. Research questions include: Do all children demonstrate less accurate/slower recognition of words spoken by multiple talkers versus by one talker? Do all children demonstrate less accurate/slower recognition of words spoken with different allophones? Do dyslexic children demonstrate less accurate/slower recognition than nondyslexic children and, if so, for all trials, only for multiple talker trials, and/or only for different allophone trials; for all dyslexic children, or only those with particular phonological impairments? [Work supported by NIH.]

4pSC5. Intertalker differences in intelligibility of cochlear-implant simulated speech. Tessa Bent, Adam B. Buchwald, and David B. Pisoni (Indiana Univ., Dept. of Psychol. and Brain Sci., 1101 E. 10th St., Bloomington, IN 47405, tbent@indiana.edu)

Are the acoustic-phonetic factors that promote highly intelligible speech invariant across different listener populations? Two approaches have been taken to investigate intelligibility variation for a variety of listener populations including hearing-impaired listeners, second language learners, and listeners with cochlear implants: studies on how speaking style affects intelligibility and other research on how inherent differences among talkers influence intelligibility. Taking the second approach, we compared intertalker differences in intelligibility for normal-hearing listeners under cochlear implant (CI) simulation ($n=120$) and in quiet ($n=200$). Stimuli consisted of 20 native English talkers' productions of 100 sentences. These recordings were processed to simulate listening with an eight-channel CI. Both clear and CI-processed tokens were presented to listeners in a sentence transcription task. Results showed that the most intelligible talkers in quiet were not the most intelligible talkers under CI simulation. Furthermore, listeners demonstrated perceptual learning with the CI-simulated speech but showed little learning in the quiet. Some of the acoustic-phonetic properties that were correlated with intelligibility differed between the CI-simulated speech and the speech in the quiet. These results suggest that the intertalker variations that result in highly intelligible speech observed in earlier studies are dependent on listener characteristics. [Work supported by NIDCD.]

4pSC6. The effect of phonological neighborhood density and word frequency on vowel production and perception in clear speech. Rajka Smiljanic, Josh Viau, and Ann Bradlow (Dept. of Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208)

Previous research showed that phonological neighborhood density and word frequency influence word recognition (Luce and Pisoni, 1998) and vowel production (Wright, 2002; Munson and Solomon, 2004; Munson, to appear), suggesting an interaction of lexical and phonetic factors in speech production and perception Here, we explore whether hyperarticulated, intelligibility-enhancing clear speech shows similar sensitivity to lexical-level structure. Nine American English talkers (five females, four males) produced 40 monosyllabic easy (frequent words with few lexical neighbors) and hard (infrequent words with many lexical neighbors) words in conversational and clear speech. Twenty-four subjects participated in a word-in-noise listening test. Results revealed a large effect of style on intelligibility and vowel production: words were more intelligible and vowels were longer and more dispersed in clear compared to conversational speech. Moreover, the female talkers produced larger vowel spaces than male talkers in both speaking styles. Vowels in hard words were marginally more dispersed than vowels in easy words in both speaking styles. However, within both speaking styles, easy and hard words were equally intelligible and of approximately equal duration. These results showed that phonetic properties of vowels were enhanced equally in clear speech regardless of their lexical properties.

4pSC7. Phoneme dependency of accuracy rates in familiar and unknown speaker identification. Kanae Amino, Takayuki Arai (Dept. of Elec. and Electron. Eng., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo, 102-8554 Japan, amino-k@sophia.ac.jp), and Tsutomu Sugawara (Sophia Univ., Chiyoda-ku, Tokyo, 102-8554 Japan)

For perceptual speaker identification, the identification accuracy depends on the speech contents presented to subjects. Our previous studies have shown that stimuli containing nasals are effective for identifying familiar speakers [Amino *et al.*, Acoust. Sci. Tech. **27**(4) (2006)]. We have also presented the possibility that the interspeaker spectral distances reflect perceptual speaker similarities. In the present study, we conducted an experiment in which four unknown speakers were identified by 15 subjects. The stimuli were identical to those used in the previous study, in which ten speakers were identified by familiar listeners, although the speakers were fewer this time. Nine consonants in the CV structure were used as stimuli. The consonants were /d/, /t/, /z/, /s/, /r/, /j/, /m/, /n/, and /nj/; the vowel was restricted to /a/ for all CV syllables to simplify the experiment. The results showed that the nasals /n/ and /nj/ obtained higher scores. Tendencies in the differences among consonants were on the same order as those of the

previous experiment, but the average scores were lower than those for familiar listeners. [Work supported by Grant-in-Aid for JSPS Fellows 17-6901.]

4pSC8. Speech style and stereotypical character in Japanese. Akiko Nakagawa (Grad. School of Cultural Studies and Human Sci., Kobe Univ., 1-2-1 Tsurukabuto, Nada-ku, Kobe 657-8501, Japan, akiko.nakagawa@atr.jp) and Hiroko Sawada (Kyoto Univ., Kyoto 606-8501, Japan)

This study shows that ''stereotypical character'' is necessary to understand Japanese speech communication in addition to existing conceptions such as emotion, communicative strategy, register, and so on. Stereotypical character is here defined as a complex entity, consisting of information about gender, age, social status, physical features, characteristics, and speech style. The necessity of stereotypical character was shown through an auditory experiment involving a total of 70 speech sounds comprised of 15–19 short phrases (mean duration 1.4) selected from recordings of spontaneous speech of four adult female speakers of Japanese. Ten participants were asked to listen to these speech sounds randomly, and to classify them into four speakers. Each of the resulting auditory-perceptual categories was found to contain speech sounds from more than one speaker. Further analyses of these results suggested that the participants classified the speech sounds not according to invariant speaker characteristics but according to virtual stereotypical characters that are common in Japanese society. Therefore, such changeable speaker characteristics as ''busybody'' ''thoughtful,'' ''high-handed,'' and so on, can be elicited through speech sounds by Japanese speakers. [This work was partially supported by the Ministry of Education, Science, Sport, and Culture, Grant-in-Aid for Scientific Research (A), 16202006.]

4pSC9. Perceived vocal age and its acoustic correlates. Hiroshi Kido (Dept. of Commun. Eng., Tohoku Inst. of Technol., Taihaku-ku, Sendai, Japan 989-8577, kidoh@tohtech.ac.jp) and Hideki Kasuya (Intl. Univ. of Health and Welfare, Otawara, Japan 324-8501)

This study investigates relationships between perceived and chronological age of talkers and acoustic correlates of the perceived age. Most of the past studies were primarily concerned with the instability of the vocal-fold vibration extracted from sustained vowels. This study focuses on the dynamic nature of sentence utterances. Talkers included 115 healthy men, aged 20–60 years, who read a short sentence in Japanese. Listeners consisted of 70 men and women, aged 20–40 years, who made direct estimations of age. The results showed a strong correlation ($r=0.66$) between the perceived and chronological age as well as the tendency toward overestimating the ages of younger talkers and underestimating those of older talkers, supporting past investigations [e.g., R. Huntley *et al.*, J. Voice **1**, 49–52 (1987)]. Acoustic parameters considered were median of the fundamental frequency ($F0$) contour, $F0$ range, declination of $F0$ contour, spectral tilt, median of the boundary frequencies above which irregularities dominate, and speaking rate. From both statistical graphical modeling and regression tree analysis, the speaking rate, $F0$ declination, and spectral tilt were found to be dominant acoustic correlates to the perceived age. [Work supported partly by a Grant-in-Aid for Scientific Research, JSPS (16300061).]

4pSC10. A cross-linguistic study of informational masking: English versus Chinese. Bruce A. Schneider, Liang Li, Meredyth Daneman (Dept. of Psych., Univ. of Toronto at Mississauga, Mississauga, ON, L5L 1C6 Canada, bschneid@utm.utoronto.ca), Xihong Wu, Zhigang Yang, Jing Chen, and Ying Huang (Peking Univ., Beijing, China 10087)

The amount of release from informational masking in monolingual English (Toronto, Canada), and Chinese (Beijing, China) listeners was measured using the paradigm developed by Freyman *et al.* [J. Acoust. Soc. Am. **106**, 3578–3588]. Specifically, psychometric functions relating percent-correct word recognition to signal-to-noise ratio were determined under two conditions: (1) masker and target perceived as originating from the same position in space; (2) masker and target perceived as originating from different locations. The amount of release from masking due to spatial separation was the same for English and Chinese listeners when the masker was speech-spectrum noise or cross linguistic (Chinese speech masking English target sentences for English listeners or English speech masking Chinese target sentences for Chinese listeners). However, there was a greater release from masking for same-language masking of English (English speech masking English target sentences) than for same-language masking of Chinese (Chinese speech masking Chinese target sentences). It will be argued that the differences in same-language masking between English and Chinese listeners reflect structural differences between English and Mandarin Chinese. [Work supported by China NSF and CIHR.]

4pSC11. Cross-linguistic differences in speech perception. Keith Johnson and Molly Babel (UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720-2650)

This research explores language-specific perception of speech sounds. This paper discusses two experiments: experiment 1 is a speeded forced-choice AX discrimination task and experiment 2 is a similarity rating task. Experiment 1 was intended to investigate the basic auditory perception of the listeners. It was predicted that listeners' native languages would not influence responses in experiment 1. Experiment 2 asked subjects to rate the similarity between two tokens on a five-point equal interval scale; the purpose of this experiment was to explore listeners' subjective impression of speech sounds. In experiment 2 it was predicted that listeners' language would affect their responses. The same stimuli were used in both experiments. The stimuli consisted of vowel-fricative-vowel sequences produced by a trained phonetician. Six fricatives were used: /f, th, s, sh, x, h/. These fricatives were embedded in three vowel environments: /a_a/, /i_i/, and /u_u/. Tokens were presented to listeners over headphones with a 100-ms interval. Independent groups of 15 native Dutch and English listeners participated in each of the two experiments. Results suggest that listeners' language influenced responses in both experiments, albeit the result was larger in experiment 2. [Work supported by NIH.]

4pSC12. Neural coding of perceptual interference at the preattentive level. Yang Zhang (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, Minneapolis, MN 55455), Patricia Kuhl, Toshiaki Imada (Univ. of Washington, Seattle, WA 98195), Toshiaki Imada, and Masaki Kawakatsu (Tokyo Denki Univ., Inzai-shi, Chiba 270-1382, Japan)

Language acquisition involves neural commitment to language-specific auditory patterns, which may interfere with second language learning. This magnetoencephalography study tested whether perceptual interference could occur at the preattentive level. Auditory mismatch field (MMF) responses were recorded from ten American and ten Japanese adult subjects in the passive oddball paradigm. The subjects read self-chosen books and ignored the sounds. Three pairs of synthetic /ra-la/ syllables were used: one cross-category pair varied only in the third formant (F3), and the other two within-category pairs varied only in the second formant (F2). ANOVA results showed a main effect of acoustic dimension with significant interaction with subject groups ($p<0.01$). As reported earlier, American listeners showed larger but later MMF responses for the F3 change. By contrast, Japanese listeners showed larger and earlier MMFs than Americans for changes in F2. Moreover, Japanese listeners had larger and earlier MMF responses for the changes in F2 as against changes in F3, which was more prominent in the right hemisphere than in the left. These results provided further support for the hypothesis that language experience produces neural networks dedicated to the statistical properties of incoming speech experienced in infancy, which later interfere with second language acquisition.

4pSC13. Russian and Spanish listener's perception of the English tense/lax vowel contrast: Contributions of native language allophony and individual experience. Maria V. Kondaurova (Program in Linguist., Purdue Univ., West Lafayette, IN 47907) and Alexander L. Francis (Purdue Univ., West Lafayette, IN 47906)

We examined the influence of listeners native phonology on the perception of American English tense and lax front unrounded vowels ([i] and [ɪ]). These vowels are distinguishable according to both spectral quality and duration. Nineteen Russian, 18 Spanish, and 16 American English listeners identified stimuli from a *beat-bit* continuum varying in nine spectral and nine duration steps. English listeners relied predominantly on spectral quality when identifying these vowels, but also showed some reliance on duration. Russian and Spanish speakers relied entirely on duration. Three additional tests examined listeners allophonic use of vowel duration in their native languages. Duration was found to be equally important for the perception of lexical stress for all three language groups. However, the use of duration as a cue to postvocalic consonant voicing differed due to phonotactic differences across the three languages. Group results suggest that non-native perception of the English tense/lax vowel contrast is governed by language-independent psychoacoustic factors and/or individual experience. Individual results show large variability within all three language groups, supporting the hypothesis that individual differences in perceptual sensitivity as well as the more frequently cited factors of second language education and experience play an important role in cross-language perception.

4pSC14. An analysis of acoustic deviation manner in spontaneous speech. Norimichi Hosogai, Kanae Okita, Takuya Aida, and Shigeki Okawa (Chiba Inst. of Technol., 2-17-1 Tsudanuma, Narashino, Chiba 275-0016, Japan)

Natural speech typically contains various phenomena deviated from the formal mode such as read speech. It is well known that those paralinguistic phenomena have an important role to give the human emotions and the state of the speakers in speech communication. This study attempts to extract the deviation as an acoustic "vagueness," defined by temporal and dynamical acoustic features of speech. Especially the change of the vagueness during a certain period of speech, such as a 10-minute presentation, is focused. As the acoustic features, it used (*i*) modulation spectrum and (*ii*) syllable speed, which may have relations to the speech clarity and the tempo. For the experiments, 70 academic presentation speech data in the Corpus of Spontaneous Japanese (CSJ) are used. As the experimental results, significant properties in the patterns of the modulation spectrum and the syllable speed are obtained as a difference of the beginning and the ending periods of the presentation. This result will contribute to a human-like speech dialog system.

4pSC15. Nondurational cues for durational contrast in Japanese. Kaori Idemaru (Dept. of East Asian Lang. and Lit., Univ. of Oregon, 1248 Univ. of Oregon, Eugene, OR 97403) and Susan G Guion (Univ. of Oregon, 1290 Eugene, OR 97403)

This study explores potential secondary cues to a durational contrast by examining short and long stop consonants in Japanese. Durational contrasts are susceptible to considerable variability in temporal dimensions caused by changes in speaking rate. In this study, the proposal is examined that multiple acoustic features covary with the stop length distinction and that these features may aid in accessing the percept intended by the speaker, even when the primary cue, closure duration, is unreliable. The results support the proposal, revealing the presence of multiple acoustic features covarying with the short versus long contrast. Not only are there durational correlates to this contrast—the preceding vowel is longer and the following vowel is shorter for geminates than singletons—but there also are nondurational features covarying with this contrast. Greater fundamental frequency and intensity drops are found from the preceding to the following vowel for the geminate than the singleton stops. These results suggest the possibility that systematic variation of these acoustic features is used in the perceptual categorization of the contrast in addition to the primary cue of closure duration. Moreover, the nondurational correlates are promising candidates for speech-rate resistant features.

4pSC16. Different motor strategies for increasing speaking rate: Data and modeling. Majid Zandipour, Joseph Perkell, Mark Tiede, Frank Guenther (M.I.T., Res. Lab Electron., Speech Commun. Group, 50 Vassar St, Cambridge, MA 02139, majidz@speech.mit.edu), Kiyoshi Honda (ATR Human Information Processing Res. Lab., Kyoto 619-0288, Japan), and Emi Murano (Univ. Maryland Dental School, Baltimore, MD, 21209)

Different motor strategies for increasing speaking rate: data and modeling EMG, kinematic and acoustic signals were recorded from two male subjects as they pronounced multiple repetitions of simple nonsense utterances. The resulting data indicate that the two subjects employed different motor strategies to increase speaking rate. When speaking faster, S1 significantly increased the size of the articulatory target region for his tongue movements, increased the speed of the tongue movements and the rate of EMG rise somewhat, while decreasing the movement duration significantly and movement distance slightly. In contrast, at the fast rate, S2 had the same size articulatory target region and rate of EMG rise as at the normal rate, but decreased the speed, distance, and duration of tongue movement slightly. Each subject had similar dispersions of acoustic targets in $F1-F2$ space at fast versus normal rates, but both shifted target centroids toward the center of the vowel space at the fast rate. Simulations with a biomechanical model of the vocal tract show how modulations of motor commands may account for such effects of speaking rate on EMG, kinematics, and acoustic outputs. [Work supported by NIDCD, NIH.]

4pSC17. Effect of speaking rate on individual talker differences in voice-onset-time. Rachel M. Theodore, Joanne L. Miller, and David DeSteno (Dept. of Psych., 125 NI, Northeastern Univ., 360 Huntington Ave., Boston, MA, 02115-5000, r.theodore@neu.edu)

Recent findings indicate that individual talkers systematically differ in phonetically relevant properties of speech. One such property is voice-onset-time (VOT) in word-initial voiceless stop consonants: at a given rate of speech, some talkers have longer VOTs than others. It is also known that for any given talker, VOT increases as speaking rate slows. We examined whether the pattern of individual differences in VOT holds across variation in rate. For example, if a given talker has relatively short VOTs at one rate, does that talker also have relatively short VOTs at a different rate? Numerous tokens of /ti/ were elicited from ten talkers across a range of rates using a magnitude-production procedure. VOT and syllable duration (a metric of speaking rate) were measured for each token. As expected, VOT increased as syllable duration increased (i.e., rate slowed) for each talker. However, the slopes as well as the intercepts of the functions relating VOT to syllable duration differed significantly across talkers. As a consequence, a talker with relatively short VOTs at one rate could have relatively long VOTs at another rate. Thus the pattern of individual talker differences in VOT is rate dependent. [Work supported by NIH/NIDCD.]

4pSC18. Variation in vowel production. Joseph Perkell, Majid Zandipour, Satrajit Ghosh, Lucie Menard (Speech Commun. Group, Res. Lab. of Electron., Rm. 36-511, M.I.T., Cambridge, MA 02139), Harlan Lane, Mark Tiede, and Frank Guenther (M.I.T., Cambridge, MA 02139)

Acoustic and articulatory recordings were made of vowel productions by young adult speakers of American English—ten females and ten males—to investigate effects of speaker and speaking condition on measures of contrast and dispersion. The vowels in the words teat, tit, tet, tat, tot, and toot were embedded in two-syllable "compound words" consisting of two CVC syllables, in which each of the two syllables comprised a real word, the consonants were /p/, /t/ or /k/, the two adjoining consonants were always the same, the first syllable was unstressed and the second stressed. Variations of phonetic context and stress were used to induce dispersion around each vowel centroid. The compound words were embedded in a carrier phrase and were spoken in normal, clear, and fast conditions. Initial analyses of $F1$ and $F2$ on 15 speakers have shown significant effects of speaker, speaking condition (and also vowel, stress, and context) on vowel contrast, and dispersion around means. Generally, dispersions increased and contrasts diminished going from clear to normal to fast conditions. Results of additional analyses will be reported. [Work supported by NIDCD, NIH.]

4pSC19. Region, gender, and vowel quality: A word to the wise hearing scientist. Richard Wright (Dept. of Linguist., Univ. of Washington, Box 354340, Seattle, WA 98195-4340, rawright@u.washington.edu), Stephanie Bor, and Pamela Souza (Univ. of Washington, Seattle, WA 98105)

Sociophonetic research has established effects of regional accent and gender on spoken vowels. Many gender differences are due to sociolinguistic factors and thus vary by region. The implications for researchers and clinicians are important: gender variation must be controlled for according to the region of the listener and talker population. Moreover, speech perception stimuli used in research and in clinical applications have limited regional application. This poster illustrates these factors using the Pacific Northwest regional accent. The data, collected for a previous study on hearing aid processing, consist of three repetitions of eight vowels produced in real-word /h_d/ (or /_d/) contexts by six males and six females ranging in age from 19 to 60. Formants were measured using an LPC with an accompanying FFT and spectrogram for verification. The results revealed vowel-specific differences in the male and female speech over and above those typically associated with physiologic predictions, and different again from those observed in past studies from different regions. Taken as a whole, these data suggest that speech and hearing researchers should take care in selecting stimuli for general-use speech perception tests. [Work supported by NIDCD training grant (#DC00033) and NIH RO1 (1 RO1 DC006014).]

4pSC20. Acoustic characteristics of vowels in three regional dialects of American English. Ewa Jacewicz, Robert Allen Fox, Yolanda Holt (Speech Acoust. and Percept. Labs., Dept. of Speech and Hearing Sci., The Ohio State Univ., Columbus, OH 43210), and Joseph Salmons (Univ. of Wisconsin—Madison, Madison, WI)

Most of the comparative sociophonetic studies of regional dialect variation have focused on individual vowel differences across dialects as well as speaker variables. The present work seeks to define basic acoustic characteristics of entire vowel systems for three different regional variants of American English spoken in southeastern Wisconsin (affected by the Northern Cities Shift), western North Carolina (affected by the Southern Vowel Shift), and central Ohio (not considered to be affected currently by any vowel shift). Three groups of speakers (men and women) aged 20–29 years were recorded from each geographic area defined by two to three counties (creating a highly homogeneous set of speakers). Acoustic measures for the set of 14 monophthongs and diphthongs in /h_d/ context included vowel space area for each speaker, global spectral rate of change for diphthongized vowels (defined over the first three formant slopes), the amount of frequency change for F1 and F2 at two temporal points located close to vowel onset and offset (vector length), and vowel duration. These measures will establish both systemic and vowel inherent characteristics across the three dialects, serving as a basis for future examination of conditioning factors on vowels in chain shifts. Dialectal differences will be discussed. [Work supported by NIH NIDCD R01 DC006871.]

4pSC21. The rhythmic characterization of two varieties of Portuguese. Verna Stockmal, Emilia Alonso Marks, Audra Woods, and Z. S. Bond (Ohio Univ., Athens, OH 45701)

As spoken in Europe, Portuguese is said to be stress-timed, while Brazilian Portuguese appears to display characteristics of both stress and syllable timing [P. A. Barbosa, D.E.L.T.A. **16**, 369–402 (2000)]. We employed the Ramus *et al.* metric, based on acoustic-phonetic measurements [Ramus *et al.*, Cognition **73**, 265–292 (1999)], to investigate the possibility of distinguishing between the two varieties of the language. Five native speakers of European Portuguese and five native speakers of Brazilian Portuguese recorded the same short prose passage taken from a magazine. The talkers were asked to read at a normal, comfortable rate. The reading time of the passage averaged 60 s, with considerable differences among the talkers. From the vocalic and consonantal intervals, the Ramus metrics, percent vocalic interval and standard deviation of consonantal and vocalic interval, were calculated. The five talkers of the two language varieties differed on the values of these parameters. The values of %V and SD-V showed overlap between the two talker groups, while the BP talkers tended to show lower values for SD-C. Apparently, the rhythmic characterization of the two varieties of the language is not clearly categorical, but rather ranges along a continuum.

4pSC22. Indexical cues to talker sex and individual talker identity extracted from vowels produced in sentence-length utterances. Michael J. Berkowitz (Dept. of Psych., 301 Wilson Hall, Vanderbilt Univ., 111 21st Ave. South, Nashville, TN 37203, michael.j.berkowitz@vanderbilt.edu), Jo-Anne Bachorowski (Vanderbilt Univ., Nashville, TN 37203), and Michael J. Owren (Georgia State Univ., Atlanta, GA 30302)

The purpose of this study was to replicate and extend a previous study of indexical cuing [J.-A. Bachorowski and M. J. Owren, J. Acoust. Soc. Am. **106**, 1054–1063 (1999)] by including more vowel sounds spoken in more diverse contexts. Specific combinations of acoustic parameters that should represent talker sex and individual talker identity were identified using predictions based on known sources of variation in vocal production-related anatomy. This study utilized 100 recordings of sentence-length utterances, produced by each of 43 male and 44 female undergraduates, as well as 22 stock-phrase recordings produced by these same participants. One of five vowel sounds (/æ, ɛ, i, ə, u/) was isolated from each sentence and analyzed for F_0, $F1$, $F2$, $F3$, $F4$, vowel duration, jitter, shimmer, and harmonicity. Classification by talker sex was nearly perfect using a combination of cues related to both vocal-fold and vocal-tract anatomy. The accuracy of classification by individual identity depended strongly on cues relating to vocal tract-variation within sex.

4pSC23. Utterance-final position and projection of femininity in Japanese. Mie Hiramoto and Victoria Anderson (Dept. of Linguist., Univ. of Hawaii, 1890 East-West Rd., Honolulu, HI 96822)

Japanese female speakers frequent use of suprasegmental features such as higher pitch, longer duration, wider pitch range, and more instances of rising intonation *vis-a-vis* male speakers, is recognized as Japanese womens language (JWL) prosody. However, these features normally co-occur with gender-specific sentence-final particles (SFPs) like the strongly feminine ‘‘kashira.’’ In this study, we examined the use of pitch and duration in utterances without SFPs, to determine whether JWL prosody is a function of SFPs or of utterance-final position. Eight male and eight female native Japanese speakers were instructed to read prepared sentences as though auditioning for a masculine theater role and then as though auditioning for a feminine role. Results indicate that utterance-final position is the projection point of JWL prosody even in the absence of SFPs. The data used for this study show high pitch, wide pitch range, long duration, and rising intonation at utterance-final positions when produced (by both men and women) in the feminine gender role. Conversely, in the masculine gender

role, both men and women avoid the use of such prosodic features, and may even avoid using rising intonation in interrogative sentences, where the tonal grammar calls for it.

4pSC24. Attitudinal correlate of final rise-fall intonation in Japanese. Toshiyuki Sadanobu (Kobe Univ., Tsurukabuto 1-2-1, Nada, Kobe, 657-8501, Japan)

Abrupt rise and subsequent fall intonation is common at the end of intonation units in Japanese, but its attitudinal correlate has not been fully elucidated yet. This intonation appears in the literature of the 1960's as politicians' way of speech, and nowadays not only politicians but many speakers including older generations often use it. However, this intonation is stigmatized as childish, and many people devalue it as an unintelligent way of speaking by young people. Where does this great gap between reality and image of this intonation come from? This presentation addresses this problem by focusing on natural conversation of Japanese daily life. The conclusions are as follows: (*i*) Rise-fall intonation often appears when the speaker talks about high-level knowledge, whereas it disappears when the speaker talks about their personal experience. (*ii*) Rise-fall intonation at the end of an intonation conveys the speaker's being so occupied with speaking that intonation unit. The childish image comes from the speaker's unawareness of their overall speech because of being occupied with local process. [Work supported by the Ministry of Education, Science, Sport, and Culture, Grant-in-Aid for Scientific Research (A), 16202006, and by the Ministry of Internal Affairs and Communications, SCOPE 041307003.]

4pSC25. Vowel devoicing in Japanese infant- and adult-directed speech. Laurel Fais, Janet Werker (Dept. of Psych., Univ. of BC, 2136 West Mall, Vancouver, BC V6T 1Z4 Canada, jwlab@psych.ubc.ca), Sachiyo Kajikawa, and Shigeaki Amano (NTT Commun. Sci. Labs., Seika-cho, Soraku-gun, Kyoto 619-0237 Japan)

It is well known that parents make systematic changes in the way they speak to infants; they use higher pitch overall, more pronounced pitch contours, more extreme point vowels, and simplified morphology and syntax (Andruski and Kuhl, 1996; Fernald *et al.*, 1989). Yet, they also preserve information crucial to the infants ability to acquire the phonology of the native language (e.g., phonemic length information, Werker *et al.*, 2006). The question examined in this paper is whether information other than phonemic segmental information is also preserved, namely, information concerning the phonological process of vowel devoicing. Devoicing of high vowels between voiceless consonants and word-finally after a voiceless consonant is a regular and well-attested phonological process in Japanese (Shibatani, 1990). A corpus of speech by Japanese mothers addressed to their infants and addressed to another adult was examined, and the degree and frequency with which they apply vowel devoicing in each type of speech was analyzed. Rates of vowel devoicing in speech to adults and infants are compared, accommodations made to infants and to hearing-impaired children are discussed (Imaizumi *et al.*, 1995), and the implications of these accommodations for the acquisition of vowel devoicing by Japanese infants are explored.

4pSC26. Language and gender differences in speech overlaps in conversation. Jiahong Yuan, Mark Liberman, and Christopher Cieri (Univ. of Pennsylvania, Philadelphia, PA 19104)

Language and gender differences in speech overlaps in conversation were investigated, using the LDC CallHome telephone speech corpora of six languages: Arabic, English, German, Japanese, Mandarin, and Spanish. To automatically obtain the speech overlaps between two sides in a conversation, each side was segmented into pause and speaking segments, and the overlap segments during which both sides were speaking were time stamped. Two types of speech overlaps are distinguished: (1) One side takes over the turn before the other side finishes (turn-taking type). (2) One side speaks in the middle of the other side's turn (backchannel type). It was found that Japanese conversations have more short (less than 500 ms) turn-taking type of overlap segments than the other languages. The average number of such segments per 10 min of conversation for Japanese was 62.6 whereas the average numbers for the other languages ranged from 37.9 to 43.3. There were, however, no significant differences among the languages on the backchannel type of overlaps. Cross-linguistically, conversations between two females contained more speech overlaps (both types) than those between a male and a female or between two males, and there was no significant difference between the latter two.

4pSC27. An acoustic study of laringeal contrast in three American Indian Languages. Heriberto Avelino (Dept. of Linguist., UC Berkeley, Berkeley, CA 94720-2650)

A contrast between modal and nonmodal phonation is commonly found in American Indian languages. The use of laryngealized voice has been reported in a number of languages from different linguistic families. This paper investigates the acoustics of laryngealized phonation in three indigenous languages spoken in Mexico, Yalalag Zapotec, Yucatec Maya, and Mixe. These languages differ in terms of the use of other features controlled by action of the larynx, i.e., tone. In Zapotec there is a contrast between high, low, and falling tones; Maya has phonemic high and low tones, whereas Mixe does not present phonemic pitch. The results show that the production of phonemic laryngeal vowels differs from language to language. The data suggest that the specific implementation of laryngealization depends in part on the relationship with contrastive tone. The patterns of the languages investigated provide new evidence of the possible synchronization of phonation throughout the vowel. With this evidence, a typology of modal/nonmodal phonation in phonation-synchronizing languages is proposed.

4pSC28. The comparison between Thai and Japanese temporal control characteristics using segmental duration models. Chatchawarn Hansakunbuntheung and Yoshinori Sagisaka (GITI, Waseda Univ., 29-7 Bldg. 1-3-10, Nishi-Waseda, Shinjuku-ku, Tokyo 169-0051, Japan, chatchawarnh@fuji.waseda.jp)

This paper compares the temporal control characteristics between Thai and Japanese read speech data using segmental duration models. The same and the different control characteristics have been observed from phone level to sentence level. The language-dependent and language-independent control factors have also been observed. In phone and neighboring phone level, different characteristics are found. Japanese vowel durations are mainly compensated by only adjacent preceding and following phones, which results from mora timing. Unlike Japanese, Thai vowel durations are affected by two succeding phones. It can be guessed that the differences come from syllabic structures. In word level, most content words tend to have longer phone durations while function words have shorter ones. In phrase level, both languages express duration lengtening of syllable/mora at the phrase initial and final. For language-specific factors, Thai tones express small alteration on phone duration. The comparisions explore the duration characteristics of the languages and give more understanding to be used in speech synthesis and second-language learning research. [Work supported in part by Waseda Univ. RISE research project of ''Analysis and modeling of human mechanism in speech and language processing'' and Grant-in-Aid for Scientific Research A-2, No. 16200016 of JSPS.]

4pSC29. Articulatory settings of French and English monolinguals and bilinguals. Ian L. Wilson (Univ. of Aizu, Tsuruga, Ikki-machi, Aizu-Wakamatsu City, Fukushima, 965-8580, Japan, wilson@u-aizu.ac.jp) and Bryan Gick (Univ. of BC, Vancouver, BC V6T1Z1 Canada)

This study investigated articulatory setting (AS), a language's underlying posture of the articulators. Interspeech posture (ISP) of the articulators (their position when motionless during interutterance pauses) was used as a measure of AS in Canadian English and Quebecois French.

Optotrak and ultrasound imaging were used to test whether ISP is language specific in monolingual and bilingual speakers. Results show significant differences in ISP across the monolingual groups, with English exhibiting a higher tongue tip, more protruded upper and lower lips, and narrower horizontal lip aperture. Results also show that upper and lower lip protrusion are greater for the English ISP than for the French ISP, in all bilinguals who were perceived as native speakers of both languages, but in no other bilinguals. Tongue tip height results mirror those of the monolingual groups, for half of the bilinguals perceived as native speakers of both languages. Finally, results show that there is no unique bilingual-mode ISP, but instead one that is equivalent to the monolingual-mode ISP of a bilingual's currently most-used language. This research empirically confirms centuries of noninstrumental evidence for AS, and for bilinguals it suggests that differences between monolingual and bilingual modes do not hold at the phonetic level.

4pSC30. Temporal and spectral variability of vowels within and across languages with small vowel inventories: Russian, Japanese, and Spanish. Franzo F. LawII (Speech Acoust. and Percept. Lab., City Univ. of New York—Grad. Ctr., 365 Fifth Ave., New York, NY 10016-4309, flaw@gc.cuny.edu), Yana D. Gilichinskaya, Kikuyo Ito, Miwako Hisagi, Shari Berkowitz, Mieko N. Sperbeck, Marisa Monteleone, and Winifred Strange (City Univ. of New York—Grad. Ctr., New York, NY 10016-4309)

Variability of vowels in three languages with small vowel inventories (Russian, Japanese, and Spanish) was explored. Three male speakers of each language produced vowels in two-syllable nonsense words (VCa) in isolation and three-syllable nonsense words (gaC1VC2a) embedded within carrier sentences in three contexts: bilabial stops in normal rate sentences and alveolar stops in both normal and rapid rate sentences. Dependent variables were syllable duration and formant frequency at syllable midpoint. Results showed very little variation across consonant and rate conditions in formants for /i/ in Russian and Japanese. Japanese short /u, o, a/ showed fronting (F2 increases) in alveolar context, which was more pronounced in rapid sentences. Fronting of Japanese long vowels was less pronounced. Japanese long/short vowel ratios varied with speaking style (isolation versus sentences) and speaking rate. All Russian vowels except /i/ were fronted in alveolar context, but showed little change in either spectrum or duration with speaking rate. Spanish showed a strong effect of consonantal context: front vowels were backed in bilabial context and back vowels were fronted in alveolar context, also more pronounced in rapid sentences. Results will be compared to female productions of the same languages, as well as American English production patterns.

4pSC31. Does infant-directed speech in Tagalog resemble infant-directed speech in American English? Emmylou Garza-Prisby, Shiri Katz-Gershon, and Jean Andruski (Aud. & Speech-Lang. Pathol. Dept., Wayne State Univ., 207 Rackham, 60 Farnsworth St., Detroit, MI 48202)

This study compared the speech of a Filipino mother in infant- and adult-directed speech in order to investigate whether the mother used the acoustic features typically found in the infant-directed speech of American English-speaking mothers. Little acoustic documentation is available on the acoustic features of Tagalog, and no acoustic comparison of speech registers has so far been conducted. Impressionistically, Tagalog-speaking mothers' do not appear to use the features typically found in American mothers speech to their young infants. The mother was recorded talking to her infant and to another adult native speaker of Tagalog. Recordings were made in the mother's home and visits occurred during the first 6 months of the infant's life. Specific acoustic features analyzed include (a) vowel duration, (b) vowel format frequencies, (c) vowel triangle size, (d) rate of speech, (e) fundamental frequency, and (f) F0 range. Morphological and syntactic features were also analyzed, including (g) mean length of utterance and (h) sentence length. Results support a need for further study of speech registers in Filipino mothers.

4pSC32. Restricting relativized faithfulness and local conjunction in optimality theory. Haruka Fukazawa (Keio Univ., 4-1-1 Hiyoshi, Kohoku-ku, Yokohama, Japan)

Within the framework of optimality theory (OT), the two mechanisms, relativized faithfulness and local conjunction, play inevitable roles, especially when a simple constraint ranking fails to account for the data. However, their domain of application are too unrestricted and even overlapping each other. For instance, there are some cases which could be explained both by the ranking with relativized faithfulness and by the one with local conjunction. Syllable-final neutralization in German and geminate devoicing in Japanese loanwords are of interest in this context. The present paper proposes formal restrictions mostly on the local conjunction mechanism: the soundness of constraint combination, the number of constraints involved in a conjunction, and the local domain of conjunction. They not only can simplify the analysis but also give a more principled account for the overlapping cases. In fact, relativized faithfulness approach wins over local conjunction approach both in German neutralization and in Japanese loanwords. It is desirable for the universal grammar to be more restricted. Removing an overlap of theoretical devices is an important step toward the goal.

Session 4pUWa

Underwater Acoustics: Session in Honor of Leonid Brekhovskikh II

William A. Kuperman, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, Marine Physical Lab., La Jolla, CA 92093-0238

Oleg A. Godin, Cochair
NOAA, Earth System Research Lab., 325 Broadway, Boulder, CO 80305-3328

Invited Papers

1:00

4pUWa1. Underwater noise as source of information on conditions and dynamics of ocean environments. Alexander V. Furduev (N. N. Andreyev Acoust. Inst., 4 Shvernika St., Moscow 117036, Russia)

Leonid Brekhovskikh wrote in his book *The Ocean and the Human* (1987): ‘‘Ocean noise is as important oceanographic parameter as temperature, current, and wind.’’ Brekhovskikh created and headed the Laboratory of Acoustic Methods of Ocean Research in 1956. One of the scientific directions of the Laboratory was investigation of underwater noise both as interference for sound reception and a source of environmental information. Long-term studies on the unique acoustic research vessels created under the initiative of Brekhovskikh resulted in numerous important findings, including ambient noise spectra and envelopes of acoustic fluctuations, depth dependence of noise directivity, and mechanisms of ambient noise generation. Brekhovskikh was always eager to find practical applications of scientific results. Different methods of ensuring noise immunity of hydroacoustic arrays were developed under his supervision. Passive methods of acoustic navigation based on use of natural noise were suggested. Techniques for underwater acoustic monitoring of the ocean based either on ambient noise analysis or reemission of noise from a point away from the receiving system have been developed. The success of the team of scientists headed by Brekhovskikh was determined by the creative atmosphere around him: there was neither competition nor commercial interests. The common goal was knowledge of the ocean.

1:20

4pUWa2. Distributed acoustic sensing in shallow water. Henrik Schmidt (Ctr. for Ocean Eng., MIT, Cambridge, MA 02139)

The significance of Leonid Brekhovskikh to the ocean acoustics is undisputed. He was pioneering not only in terms of fundamental understanding of the ocean acoustic waveguide, but also the development of efficient and numerically stable approaches to propagation of sound in a stratified ocean. As such he has been an inspiration to a whole generation of model developers, leading to today’s suite of extremely powerful wave theory models, capable of accurately representing the complexity of the shallow-water ocean waveguide physics. The availability of these computational tools have in turn led to major advances in adaptive, model-based signal-processing techniques. However, such computationally intensive approaches are not necessarily optimal for the next generation of acoustic sensing systems. Thus, ocean observation in general is currently experiencing a paradigm shift away from platform-centric sensing concepts toward distributed sensing systems, made possible by recent advances in underwater robotics. In addition to a fully autonomous capability, the latency and limited bandwidth of underwater communication make on-board processing essential for such systems to be operationally feasible. In addition, the reduced sensing capability of the smaller physical apertures may be compensated by using mobility and artificial intelligence to dynamically adapt the sonar configuration to the environment and the tactical situation, and by exploiting multiplatform collaborative sensing. The development of such integrated sensing and control concepts for detection, classification, and localization requires extensive use of artificial intelligence incorporating a fundamental understanding of the ocean acoustic waveguide. No other sources in literature provide this with the clarity and depth that is the trademark of Academician Brekovskikh’s articles and classical textbooks. [Work supported by ONR.]

Contributed Papers

1:40

4pUWa3. When the shear modulus approaches zero: Fluids don’t bend and Scholte leaves the room. Robert I. Odom (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698), Donna L. G. Sylvester, and Caitlin P. McHugh (Seattle Univ., Seattle, WA 98122-1090)

The 4×4 linear system of differential equations describing the propagation of the displacements and tractions in an elastic layered medium becomes singular as the shear modulus of the elastic medium approaches zero. There are a number of approximate ways to handle this singularity in order to impart numerical stability to the computation of the elastic waves in a layered medium. For example, one can impose an irrotational constraint on the displacements or introduce a massive elastic interface (MEI). Both of these ways of handling the weak shear strength are approximate, but avoid the need for singular perturbation theory [Gilbert, 1998]. Scholte waves are interface waves that propagate along the interface between an elastic solid and a fluid. They have nodes near or on the interface and decay exponentially into the bounding media. Scholte waves do not occur at the boundary between fluids. As the shear speed in the bounding elastic medium approaches zero, the Scholte waves disappear from the spectrum. We investigate this disappearance by applying singular perturbation theory to the coupled fluid-elastic system. Among other things, we will address the rate in wave-number space at which the Scholte waves disappear from the spectrum.

1:55

4pUWa4. Measurement of the plane-wave reflection coefficient of the ocean bottom and the legacy of Leonid Brekhovskikh. George V. Frisk (Florida Atlantic Univ., 101 N. Beach Rd., Dania Beach, FL 33004) and Luiz L. Souza (Bose Corp., Framingham, MA 01701)

Leonid Brekhovskikh's classic text [*Waves in Layered Media* (Academic, New York, 1980)] inspired the development of several techniques for measuring the plane-wave reflection coefficient of the ocean bottom. Specifically, his application of the geometrical acoustics approximation to the problem of reflection of a spherical wave from a horizontally stratified medium provided the theoretical foundation for evaluating the strengths and weaknesses of various measurement methods. The most popular method assumes that the reflected field also consists of a spherical wave multiplied by the reflection coefficient evaluated at the specular angle. However, Brekhovskikh's work showed that this interpretation is confined to a limited range of angles and bottom structures and, if applied improperly, can lead to unphysical results such as negative bottom loss. This paper describes a technique which circumvents these deficiencies. It consists of measuring the pressure field magnitude and phase versus range due to a cw point source and Hankel transforming these data to obtain the depth-dependent Green's function versus horizontal wavenumber. The reflection coefficient is then obtained from the Green's function using the analytical relationship between the two quantities. The method is demonstrated using 220-Hz data obtained in a near-bottom geometry in the Icelandic Basin. [Work supported by ONR.]

2:10

4pUWa5. Field from a point source above a layered half-space; theory and observations on reflection from the seabed. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA)

L. M. Brekovskikh's book *Waves in Layered Media* has provided decades of graduate students and researchers alike with a comprehensive and enormously useful reference. One topic from that work, reflection from a point source above a plane-layered medium (the seabed), is discussed here. Both theoretical underpinnings and observations of reflection from a homogeneous halfspace, a transition layer with smoothly varying density and velocity profiles, and discrete layered media are considered for various shallow water sediment fabrics. [Work supported by the Office of Naval Research and NATO Undersea Research Centre.]

2:25

4pUWa6. Plane-wave model and experimental measurements of the directivity of a Fabry-Perot, polymer film, ultrasound sensor. Benjamin T. Cox and Paul C. Beard (Dept. of Med. Phys. and Bioengineering, Univ. College London, Gower St., London, WC1E 6BT, UK, bencox@mpb.ucl.ac.uk)

Optical detection of ultrasound is popular due to the small element sizes that can be achieved. One method exploits the thickness change of a Fabry-Perot (FP) interferometer caused by the passage of an acoustic wave to modulate a laser beam. This detection method can have greater sensitivity than piezoelectric detectors for sub-millimeter element sizes. The directivity of FP sensors and the smallest achievable effective element size are examined here. A plane-wave model of the frequency-dependent directional response of the sensor, based on Brekhovskikh's work on elastic waves in layered media, is described and validated against experimental directivity measurements made over a frequency range of 15 MHz and from normal incidence to 80 deg. In terms of applications, the model may be used to provide a noise-free response function that can be deconvolved from sound field measurements in order to improve accuracy in high-frequency metrology and imaging applications, or, for example, as a predictive tool to improve sensor design. Here, the smallest achievable effective element radius was investigated by comparing the directivity with that of a rigid circular pressure transducer, and found to be $\sim 0.9d$, where d is the thickness of the FP interferometer. [Funding was provided by the EPSRC, UK]

2:40

4pUWa7. The interference head wave and its parametric dependence. Jee Woong Choi and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington)

The interference head wave that can exist in the presence of a sound-speed gradient in the sediment, is a precursor arrival representing a transition between the first-order head wave and the zeroth-order refracted wave. Using a parabolic equation (PE) simulation, Choi and Dahl [J. Acoust. Soc. Am. **119**, 2660–2668 (2006)] showed how the small shift in the dominant frequency of the interference head wave behaves as a function of the nondimensional parameter zeta, which itself is a function of center frequency, gradient, and range. For example, it was shown that the maximum frequency shift occurring in the vicinity of zeta equals 2. In this work, we investigate the amplitude and additional spectral properties of the interference head wave and analyze the cause of the frequency shift phenomenon using the ray theory. The limitation on the application of ray method also will be discussed. Finally, the conclusion will be verified by the time-dependent simulation using the RAM PE algorithm. [Work supported by the ONR.]

Session 4pUWb

Underwater Acoustics: Session in Honor of Fredrick Fisher

William A. Kuperman, Chair
Scripps Inst. of Oceanography, Univ. of California, San Diego, Marine Physical Lab., La Jolla, CA 92093-0238

Chair's Introduction—3:10

Invited Papers

3:15

4pUWb1. FLIP (Floating Instrument Platform): A major Fred Fisher contribution to ocean science. Fred N. Spiess, Robert Pinkel, William S. Hodgkiss, John A. Hildebrand (Marine Physical Lab., Scripps Inst. of Oceanogr., UCSD 0205, 9500 Gilman Drive; La Jolla, CA 92093-0205, fspiess@ucsd.edu), and Gerald L. D'Spain (UCSD, La Jolla, CA 92093-0205)

Frederick H. Fisher, a loyal and zealous member of the Acoustical Society of America, was an imaginative and effective developer of new techniques for research in both laboratory and seagoing acoustics. Most notable among his contributions was his work in bringing into being and enhancing the usefulness of the spar buoy laboratory, FLIP, from its inception in 1960. Not only did Fred use FLIP in his own research, its existence and many of its ancillary capabilities constituted a base for the seagoing research of others. The authors of this paper have benefited from FLIP's unique capabilities, starting with long-range sound propagation studies in the 1960's and 1970's. FLIP's stability and deep draft structure provided the platform for development of acoustic Doppler techniques for the measurement of ocean currents. Most recently, FLIP has been involved in studies of marine mammal vocalizations and use of multielement arrays to investigate details of shallow-water propagation. Fred's initial studies of sonar bearing accuracy, for which FLIP's construction was funded, and his dedication to advancing FLIP's ability to contribute to ocean science, constitute a legacy that is being utilized today, more than 40 years after FLIP's launching.

3:35

4pUWb2. Absorption of sound in seawater and ocean ambient noise, the scientific passions of Fred Fisher. John A. Hildebrand (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0205)

Fred Fisher made seminal contributions to ocean acoustics in the understanding of the absorption of sound in seawater and ocean ambient noise. Laboratory data and long-range sound propagation data revealed excess acoustic absorption in seawater. Fred Fisher spent much of his scientific career, beginning with his Ph.D dissertation, teasing out the contributions of various seawater components to sound absorption, and his work on this topic set the standard for understanding and modeling these phenomena. Ambient noise is an important aspect of underwater signal detection and is the focus of recent concerns about disturbance of marine organisms. Fred Fisher made important contributions to ambient noise studies by conducting measurements of vertical directionality, thereby testing models for ambient noise production. The value of archival ambient noise data and recent increases in ambient noise will be discussed.

Contributed Papers

3:55

4pUWb3. Fred Fisher's high-pressure work with eyewash and epsom salts. Christian de Moustier (Ctr. for Coastal & Ocean Mapping, Chase Ocean Eng. Lab, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824)

Starting in 1957 Fred Fisher led research programs devoted to high-pressure measurements related to the physical chemistry of sound absorption in seawater due to magnesium sulfate and other salts. As he put it, he spent his professional lifetime squeezing epsom salt. His interest in the low-frequency anomalous sound absorption in the ocean below 1 kHz led to the discovery of boric acid as the cause of the low-frequency relaxation. This paper is a short review of Fred Fisher's contributions to our knowledge of sound absorption in seawater, based in part on his carefully handwritten lecture notes and numerous low-pressure discussions.

4:10

4pUWb4. Fred Fisher and research with acoustic vector sensors; Marine Physical Laboratory's vertical array of directional acoustic receivers and ocean noise. Gerald L. D'Spain and William S. Hodgkiss (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jollla, CA 93940-0701)

Fred Fisher had boundless enthusiasm for all topics acoustic. A chance encounter with him in the hallway usually led to a half-hour discussion of the latest research efforts at the lab and recent results he found exciting. In the 1980s, Fred became interested in the problem of identifying the physical phenomena forming the pedestal about the horizontal in vertical directionality measurements of the deep ocean's low-frequency noise field. Two competing mechanisms had been proposed: downslope conversion of coastal shipping and noise from high latitude winds coupling into the deep

4p FRI. PM

sound channel due to the shoaling of the sound channel axis. The relative contributions of these two mechanisms possibly could be separated if the azimuthal ambiguity of a vertical line array of hydrophones somehow could be broken. Therefore, Fred proposed to build a vertical array of "DIFAR" sensors, which led to the design and construction of the Marine Physical Lab's Vertical "DIFAR" Array. This talk will reminisce a bit about Fred as well as present some results from an ambient noise experiment conducted in 1992 on the continental shelf using the Vertical DIFAR Array co-deployed with MPL's freely drifting vector sensors, the Swallow floats. [Work supported by ONR and ONT.]

FRIDAY EVENING, 1 DECEMBER 2006 HAWAII BALLROOM, 7:00 TO 10:00 P.M.

Awards Ceremony

Anthony A. Atchley, President
Acoustical Society of America

Yôiti Suzuki, President
Acoustical Society of Japan

Acknowledgment of Honolulu Local Meeting Organizing Committees

Presentation of Fellowship Certificates

Anders Askenfeldt
Sergio Beristain
Philippe Blanc-Benon
David A. Conant
Andes C. Gade
Anthony W. Gummer
Charles W. Holland
Jody E. Kreiman
Kevin D. LePage
James A. McAteer
David R. Palmer
Marehalli G. Prasad
Hiroshi Riquimaroux
Peter A. Rona
Mark V. Trevorrow
Michael Vörlander
Joos Vos
Ben T. Zinn

Science Writing Award in Acoustics for Journalists to Radek Boschetty

Science Writing Award for Professionals in Acoustics to Edwin Thomas

Announcement of 2005 A. B. Wood Medal and Prize to Aaron Thode

Distinguished Service Citation to Thomas D. Rossing
Silver Medal in Noise to Alan H. Marsh
Silver Medal in Physical Acoustics to Henry E. Bass
Silver Medal in Psychological and Physiological Acoustics to William A. Yost
Wallace Clement Sabine Award to William J. Cavanaugh

Recognition of Acoustical Society of Japan meeting organizers
Recognition of Acoustical Society of America meeting organizers