Keynote Lecture

Invited Paper

8:20

Reservoir acoustics and its applications. Xiuming Wang (State Key Lab of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China)

Reservoir is an underground formation with various pores that is able to store oil, water and/or gas, while reservoir acoustics is related to studies and applications of acoustic wave interactions with multiphase porous media. The reservoir acoustics stems from seismic petroleum exploration (including single well imaging, vertical seismic profile, and cross-well tomography), acoustic logging, rock acoustic measurements, sedimentology, digital acoustic signal processing and imaging, and so on, through which it has developed into a comprehensive acoustics branch for the petroleum exploration and exploitation. In this presentation, I will give a comprehensive review for reservoir acoustics and discuss the technical problems in this recently developed acoustics branch with its major applications. This will include the concepts of reservoir acoustics, the research areas of reservoir acoustics, important applications, and future developments. I will focus on the progress of acoustic wave propagation in multiphase porous mediums, time reversal acoustic imaging for seismic data, and hydrostatic pressured fracture detection in enhanced oil recovery using micro-seismics, three-dimensional borehole acoustic well logging. Especially, I will discuss the borehole acoustic mode characteristics for borehole acoustic wavefield representation in complex borehole configurations. Also, I will discuss the applications of borehole acoustic well logging in formation anisotropic identifications, stress evaluations, and so on.

FRIDAY MORNING, 18 MAY 2012 HALL A, 9:15 A.M. TO 12:40 P.M.

Session 5aAA

Architectural Acoustics and Psychological and Physiological Acoustics: Objective and Subjective Parameters of Spatial Impression in Performing Arts Spaces I

Michelle Vigeant, Cochair
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Jin Yong Jeon, Cochair
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Chair's Introduction—9:15

Invited Papers

9:20

5aAA1. 45 years of spatial impression. Mike Barron (Fleming & Barron, Combe Royal Cottage, Bathwick Hill, Bath BA2 6EQ, UK, m.barron@btinternet.com)

The story of spatial hearing has a certain circularity about it. To date, two separate spatial effects have been identified. The first effect is that created by a reverberant field in an enclosure, which is particularly obvious in a cathedral-type space. A second spatial effect was proposed 45 years ago linked to early lateral reflections. These two are now known as Listener Envelopment (LEV) and Source Broadening (or Apparent Source Width, ASW). Interesting research had been conducted into "Raumlichkeit" in Germany during the '60s. But following the revelations by Marshall, interest in the effects of early lateral reflections overshadowed interest in what is now called LEV. Sound level also contributes to the magnitude of these spatial effects. Measures have been proposed for both ASW and LEV, which include both reflection directional distribution and sound level. The paper will summarise the history of spatial impression and what implications it has for concert hall design. Historical reference: Applied Acoustics (2001) 62, 91-108 and 185-202.

9:40

5aAA2. Spatial impressions from late overhead reflections in concert halls. Toshiki Hanyu and Kazuma Hoshi (Nihon University, 7-24-1 Narashinodai, Funabashi, Chiba 274-8501, hanyu@arch.jcn.nihon-u.ac.jp)

It is known that not only late lateral reflections but also late overhead reflections contribute to listener envelopment (LEV) which is one of the perceptive factors in the spatial impression of sounds in a concert hall. This paper describes results of psychological experiments which are intended to clarify differences between spatial impressions of the late overhead reflections and that of the late lateral reflections. The results are the following: (1) Late overhead reflections have a different acoustic effect from late lateral reflections. (2) Late overhead reflections increase not only LEV but also overhead sound image (OSI). (3) Late lateral reflections contribute mainly to horizontal listener envelopment (HLEV). (4) If OSI is increased in addition to HLEV, a listener feels vertical listener envelopment (VLEV).

10:00

5aAA3. Spatial room acoustics descriptors from a concentric source-receiver array: potential applications for auditoria. Luis Miranda, Densil Cabrera, Ken Stewart, and William L. Martens (Faculty of Architecture, Design and Planning, The University of Sydney, NSW 2006, Australia, lmir9852@uni.sydney.edu.au)

Electroacoustic transducer arrays, both microphones and loudspeakers, are emerging as promising tools for measuring and evaluating room acoustics – accounting for source directivity and the angular distribution of the received room reflections over time. This study examines the potential of a concentric source-receiver array for characterizing the spatial response of rooms. Using this concept, the acoustic response at a point in a room is represented by a matrix of impulse responses comprising the combined spherical harmonic series for source and receiver. The spatial analysis of this matrix yields the spatial room response for a source of arbitrary directivity and orientation (limited by the spherical harmonic order implemented). The reduction of such data to parameters can be approached in many ways, and this paper considers the mean and standard deviation of diffusivity index for an nth order cardioid source. Measurements from a prototype transducer are presented. In auditoria, this approach could be well suited to the evaluation of acoustic conditions on stage, and should be especially relevant to describing the effect of acoustics on solo performance.

10:20

5aAA4. The preference is driven by intimacy—which is furthermore provided by lateral reflections. Tapio Lokki, Antti Kuusinen, Jukka Pätynen, and Sakari Tervo (Aalto University School of Science, P.O. Box, 15400 FI-00076 Aalto, Finland, Tapio.Lokki@aalto.fi)

Nine concert halls were measured with a calibrated loudspeaker orchestra which eliminates all variables except the acoustics of the halls. The loudspeaker orchestra was located on the stage and the microphone array at 12 m distance from it in the audience area. Subjective data including preference ratings of 17 assessors and subjective sensory profiles were collected with the individual vocabulary profiling process. The objective room acoustic parameters were also calculated. All data were analyzed in a common factorial space with the hierarchical multiple factor analysis and preference mapping. The results show that the preference of assessors is divided into two groups and the factors influencing the preference can be explained with sensory profiles. However, the most interesting result is that the overall preference is driven by the intimacy, i.e., how close to the listener the sound is perceived by the listener. None of the current objective parameters could explain the perceived intimacy. The presentation will explain that the perceived distance, thus intimacy, is mainly affected by the seat-dip effect and the number of early lateral reflections, in addition to overall loudness particularly at low frequencies.

10:40–11:00 Break

11:00

5aAA5. Coupled volumes in concert halls—impact on spatial perception. Tateo Nakajima and Todd L. Brooks (Artec Consultants Inc, New York, tn@artecconsultants.com)

It has long been understood that the use of variably coupled volumes in concert hall design has an impact on room acoustics beyond reverberation and the perception of reverberance. In this presentation, Artec will discuss recent examinations of and experiences in coupled volume halls on spatial perception, both in terms of subjective experience and measured data. Various spatial phenomena experienced by both musicians and audience members will be compared and discussed by the presenter. Artec will further discuss future design considerations related to these new observations in light of previous designs with variably coupled upper volumes and with variably coupled side chambers.

11:20

5aAA6. Acoustical considerations for improving spatial impression in designing concert halls. Jin Yong Jeon, Yong Hee Kim, and Hyung Suk Jang (Hanyang University, Seongdong-gu, Seoul 133-791, Korea, jyjeon@hanyang.ac.kr)

One of the major acoustical considerations in designing concert hallsisimproving spatial impressions through providing more early lateral reflections.The lateral energy fraction (LF) and the interaural cross correlation (IACC) are useful measures for assessing design alternatives in a viewpoint of better spatial impression. In this paper, design developing process for improving spatial impression was demonstrated with a concert hall case.In the early design stage, a bell-shaped cross-section with vineyard seating was proposed to enhance the lateral reflections for the audience. The main reflecting surfaces, which were stage enclosure, lateral walls, balcony fronts, and ceiling, were evaluated to optimize the shape and angle for early reflections. The horizontal diffusers were additionally installed for the improving the spatial impression. The validation was performed with computer simulations and 1:10 scale model measurements for the evaluation of the alternatives.

11:40

5aAA7. The effects of focussing elements on spatial sound indices. John O'Keefe (Aercoustics Engineering Limited, 50 Ronson Drive, Suite 165, Toronto, M9W 1B3, Canada, johno@aercoustics.com)

Building geometries that focus sound are generally thought to be acoustically deleterious. This despite evidence to the contrary, such as barrel vaulted naves in churches and cathedrals. There are successful concert venues with barrel-vaulted ceilings, notably London's Wigmore Hall and sections of rooms that benefit domed ceilings, e.g. the balcony of Vancouver's Orpheum. The present study experiments with focusing elements in computer models, concentrating on barrel vaulted ceilings above a shoe-box plan. Building on the author's previous experiments with Height/Width ratios, similar studies have been carried out on single and double radius curved ceilings. In terms of spatial impression, a tall narrow shoe-box shaped room benefits from a barrel-vaulted ceiling whose focus (or foci) are appropriately high enough about the listening plane.

12:00

5aAA8. Evaluation of differences in perceived overall acoustical quality and listener envelopment from binaural recordings in a 900-seat hall. Michelle C. Vigeant (Acoustics Prog. and Lab., Dept. of Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., W. Hartford, CT 06117, vigeant@hartford.edu), Jenna M. Daly, and Michael J. Dick

Impulse response measurements and binaural recordings were taken in a 900-seat multi-purpose hall in three clusters of nine adjacent seats. Each cluster contained a central seat, the seats immediately to the left and to the right, and the three seats in the rows directly in front of and behind the central seat. Differences between the seats were analyzed in terms of the standard room measures and spatial measures lateral energy fraction (LF) and late lateral energy level (GLL). The typical JNDs were used to evaluate the differences between seat pairs within a given cluster. Differences on the order of 1 or more JNDs were found between most seat pairs within a cluster in most octave bands between 125 Hz to 4000 Hz for early decay time, clarity index, strength, and LF. In terms of GLL, the average difference between pairs of seats was 0.6 dB. However, a total of 14 pairs of seats had differences in GLL between 1.0 to 2.0 dB. A listening test was carried out to determine if subjects could hear differences in the binaural recordings in terms of the listener envelopment and overall acoustical quality. The measurement and subjective test data will be presented.

12:20

5aAA9. Why the conventional RT is not applicable for testing the acoustical quality of unroofed theatres. Fangshuo Mo and Jiqing Wang (Institute of Acoustics, Tongji University, Shanghai 200092, China, mfs@tongji.edu.cn)

The rate of sound decay in an enclosed space has been used as the measure for assessing the reverbrance initiated by Sabine a century ago. Such evaluation is based on the energy consideration and can be tested by a monophonic receiving system. But, it is questionable for reverberance evaluation of unroofed spaces, such as, traditional Chinese courtyard theatres or Greek/Roman amphitheatres. It can be seen that even the decay rates of the enclosed and unroofed spaces are similar, but their reflectograms show significant difference due to the absence of reflections from top in the unroofed space. The spatial distributions of the reflections toward the listeners for both spaces are also very different. Series of subjective comparison tests of synthetic room impulse responses through stereo-system (pickup and playback) in our laboratory also showed that the reverberance in an unroofed space was quite different from the result through mono-system as in the conventional way following ISO 3382. However, subjective tests through mono-system for enclosed and unroofed spaces with similar decay rate do not show significant difference of reverbrance. Therefore, in an unroofed theatre, the factor of spatial information of the reflections should not be neglected for reverbrance criteria.

Session 5aAB

Animal Bioacoustics: Acoustic Monitoring of Animals

Douglas Cato, Cochair
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Kotaro Ichikawa, Cochair
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Contributed Papers

9:20

5aAB1. Gliders, floats, and robot sailboats: autonomous platforms for marine mammal research. David K. Mellinger, Holger Klinck (Coop. Inst. for Marine Resources Studies, Oregon State University and NOAA Pacific Marine Environmental Laboratory, 2030 SE Marine Science Drive, Newport, OR 97365, David.Mellinger@oregonstate.edu), Neil M. Bogue, Jim Luby (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105), Haru Matsumoto (Coop. Inst. for Marine Resources Studies, Oregon State University and NOAA Pacific Marine Environmental Laboratory, 2030 SE Marine Science Drive, Newport, OR 97365), and Roland Stelzer (Austrian Society for Innovative Computer Sciences, Haussteinstrasse 4/2, 1020 Vienna, Austria)

Passive acoustic monitoring (PAM), now widely used for marine mammal research, is typically conducted using hydrophone arrays towed behind ships, providing real-time data from large areas over short time spans (days to weeks), or using fixed autonomous hydrophones, providing non-real-time data from small areas over long time spans (months to years). In contrast, mobile platforms can supply near-real-time data over spatiotemporal scales large in both space and time. These systems are deployed from a vessel, communicate via satellite with shore stations for navigation and control updates, and report in near-real time upon detecting marine mammal or other sounds of interest. Acoustically-equipped gliders are buoyancy-driven devices that are capable of traversing long distances (hundreds to thousands of kilometers) over weeks to months of autonomous operation. Autonomous floats such as QUEphones drift with currents or park on the seafloor, rising to the surface upon detecting sounds of interest. Robot sailboats such as the Roboat use wind to propel themselves quickly over long distances. All platforms can store large datasets and carry additional sensors (e.g., temperature, salinity, chlorophyll, pH, O2), and are therefore well-suited for investigating oceanographic and ecological questions. Advantages and disadvantages of these platforms for various applications will be discussed.

9:40

5aAB2. Simultaneously monitoring noise and cetaceans in the Ligurian Sea. Giacomo Giorli (University of Hawaii—Department of Oceanography, 1000 Pope Road, Honolulu, HI 96822, giacomog@hawaii.edu), David Hughes (NATO Undersea Research Center, Viale San Bartolomeo 400, 19126 La Spezia, Italy), and Withlow Au (University of Hawaii—Hawaii Institute of Marine Biology, P.O. Box 1106, Kaneohe, HI 96734)

The interest in the effect of anthropogenic noise on marine life increasing rapidly, this is driven primarily by legislations in many countries that noise can be considered to be a pollutant, as for the European Marine Strategy Framework Directive. One particular area of concern has been in the interaction of military tactical sonar and the acoustically more sensitive species of cetacean, in particular the effect on beaked whales. Five ecological acoustics recorders (EAR) buoy having a sampling frequency of 80 kHz and a hydrophone with a flat sensitivity of -193.5 $\pm$ 1dB re 1 μPa up to 40 kHz have been deployed in the Northern Mediterranean at a depth of about 850 m in a canyon region known to host a population of Cuvier beaked whales (Ziphius cavirostris). The area is also located in the nearby of different shipping harbors and ferries routes. The utility of the EAR systems is that they simultaneously allow monitoring of noise levels across a wide frequency range while simultaneously allowing assessment of the presence of a range of different species – including Cuvier beaked whale sperm whale and dolphins. Here we discuss the initial results for noise assessment and for cetacean monitoring.

10:00

5aAB3. Classification of odontocete using spectral properties of echolocation clicks. Yang Lu, Holger Klinck, and Dave Mellinger (CIMRS, 2030 SE Marine Science Drive, Newport, OR 97365, lu.yang@noaa.gov)

The spectral properties of echolocation clicks are investigated for six species' sounds collected from ships and platforms in the North Pacific: bottlenose dolphins (Tursiops truncatus), melon-headed whales (Peponocephala electra), short- and long-beaked common dolphins (Delphinus delphis and D. capensis), Hawaiian spinner dolphins (Stenella longirostris longirostris) and beaked whales (family Ziphiidae). Spectral properties of different frequency bands are compared among the six species and derivatives of spectral slopes and differences between unique spectral peaks and notches are calculated as features to characterize the clicks. Both classification and regression trees (CART) and random forests are employed for species classification. The performance of these methods is compared in terms of complexity and accuracy.

10:20

5aAB4. Individual variability of echolocation and diving behavior of Yangtze finless porpoises. Satoko Kimura (Graduate School of Informatics, Kyoto University, Kyoto 606-8501, Japan, sk0130@bre.soc.i.kyoto-u.ac.jp), Tomonari Akamatsu (National Research Institute of Fisheries Engineering, Fisheries Research Agency, Hasaki, Kamisu, Ibaraki 314-0408, Japan), Kexiong Wang, Ding Wang (Key Laboratory of Aquatic Biodiversity and Conservation of the Chinese Academy of Sciences, Institute of Hydrobiology, The Chinese Academy of Sciences, Wuhan 430072, P.R. China), Songhai Li (Marine Mammal Research Laboratory, Tropical Marine Science Institute, National University of Singapore, 14 Kent Ridge Road 119223, Singapore), and Nobuaki Arai (Graduate School of Informatics, Kyoto University, 606-8501 Kyoto, Japan)

Passive acoustic monitoring has been widely applied to observe the presence, movement and behavior of the target species. For quantitative analysis, the acoustic cue production rate of the target animals must be observed in advance. We examined the detailed pattern of biosonar and swimming behaviors of 10 Yangtze finless porpoises (1 female and 9 males) obtained by electronic tags attached to the animals. The click trains produced by the tagged animal were identified out of other animals' vocalizations using bearing angles calculated by the time-of- arrival differences of the sounds between the two hydrophones in the acoustic tag. The number of click trains

produced in 10 min did not relate to the body size or sex and varied from 0 to 290. Although deeper the animals dive, faster they swam, the speed or maximum depth of the animals had no correlation with the number of click trains produced or body size. All parameters we examined had no diurnal rhythm but had independent aperiodic variation. The sound production rate without day or night bias is suitable for the quantitative passive acoustic monitoring of this species.

10:40–11:00 Break

11:00

5aAB5. Study of cetaceans in Istanbul Strait using passive acoustic method. Saho Kameyama (Biosphere Informatics 36-1 Yoshida-Honmachi, Sakyo-ku, Kyoto 606-8501, Japan, kamesaho@bre.soc.i.kyoto-u.ac.jp), Ayhan Dede, Ayaka A. Ozturk (Istanbul University), Tomonari Akamatsu (National Research Institute of Fisheries Engineering, 7620-7 Hasaki, Kamisu, Ibaraki 314-0408, Japan), Arda M. Tonay, Bayram Ozturk (Istanbul University), and Nobuaki Arai (Biosphere Informatics 36-1 Yoshida-Honmachi, Sakyo-ku, Kyoto 606-8501, Japan)

The Istanbul Strait connects to the Aegean Sea and the Black Sea. Three cetaceans appear in this Strait, harbor porpoise (Phocoena phocoena relicta), common dolphin (Delphinus delphis ponticus), bottlenose dolphin (Tursiops truncatus ponticus). We used stereo passive acoustic monitoring system (A-tag) to monitor the moving pattern of these cetaceans from July 2009 to September 2010. This system enables to record the sound source direction calculated by the sound arrival time difference between two hydrophones. They have different frequency sensitivity, which enable to distinguish Phocoenidae from Delphinidae. Phocoenidae and Delphinidae were detected most frequently in April. They stay near the A-tag in March and April than in August. Acoustic sensing distance, which is proportional to the inter-click-interval of the sonar signals, was short in the same months. This was more clear in Phocoenidae. Dominant behavior pattern of both species was staying rather than moving in all seasons. However, Phocoenidae did not stay long time in August. Moving behavior was associated with longer sensing distance comparing with staying. Especially in August, short distance sensing was less frequent in Phocoenidae. All these findings suggest that cetaceans may feed in this area mainly in March and April and travel in August.

11:20

5aAB6. Underwater acoustic detection and classification for cetaceans' vocalizations of Marine Observatory in the Northeastern Taiwan (MONET). Yin-Ying Fang, Kao-Chao Wu, and Chifang Chen (Department of Engineering Science and Ocean Engineering, National Taiwan University, Taipei, Taiwan 10617, R.O.C., ininsupersmart@gmail.com)

Passive acoustics is as an important tool for observing marine animals and long-term underwater environment monitoring. Since the amount of data is enormous, an effective auto-detector to select critical features and classify their patterns from the recorded acoustic signal. In this study, we had developed an automatic detector with both the feature extraction and classification modules. In the feature extraction module, we select features from the energy and end-point of the time signal in needed. Then, we normalized the extracted features as inputs for the classification module based on the theory of back propagation neural network (BPNN). The BPNN will be trained and tested using both the cetaceans' acoustic signals which recorded from the hydrophone of Marine Cable Hosted Observatory (MACHO system) until the network becomes stable and convergent. Identification objects are chosen commonly seeing cetaceans from the northeastern offshore of Taiwan, Guishan Island. The detector is a robust tool which has good recognition rate for classifying cetaceans. It supersedes the experienced human operators due to less time consuming and low labor cost. (sponsored by National Science Council of Republic of China under project "Marine Observatory in the Northeastern Taiwan (MONET)" No. NSC 100-2221-E-002-027)

11:40

5aAB7. Acoustic estimation of effective gathering range of shipboard fishing light for squid jigging. Yoshimi Takao, Hideo Takahara (National Research Institute of Fisheries Engineering, ytakao@affrc.go.jp), Takafumi Shikata (Ishikawa Prefecture Fisheries Research Center), Susumu Namari, Toyoki Sasakura (Fusion Incorporation), and Toshihiro Watanabe (National Research Institute of Fisheries Engineering)

The effective range of fishing light for the Japanese common squid jigging was investigated using acoustic methods in the Japan Sea on August 21- 24, 2011. The research vessel collected acoustic signals of a sonar and a quantitative echosounder from squid around the jigging boat equipped with metal halide lamps. Digital images of sonar echogram were composed to make maps which can be used to measure locations, sizes, and relative echo intensity of squid school. Behavior of individual squid was observed by an acoustic tag. The receiver was prepared on each vessel. The tag was attached on a fin of squid onboard the research vessel. Total 22 tagged squids were released from the research vessel in three nights and the distance between the vessels at release time ranged from 0.25 to 1.1 nmi. Total 7 squids were received at the jigging boat after 1 - 3 hours from their release. The distance between vessels at release time ranged 0.25 to 0.7 nmi. The squids which were released within 0.5 or 0.6 nmi were received at the jigging boat in every night. From these results, the effective gathering range of the fishing light was estimated to be at least 0.5 nmi.

12:00

5aAB8. Behavioral in-air audiogram of two Arctic fox (*Alopex lagopus*) at the Columbus Zoo and Aquarium. Amanda Stansbury (Western Illinois University, Department of Biology, 3561 60th Street Moline, IL 61265, stansbury.amanda@gmail.com), Jeanette Thomas (Western Illinois University, Department of Biology, 3561 60th Street Moline, IL 61265), Colleen Stalf, Lisa Murphy (Niabi Zoo, 12908 Niabi Zoo Road, Coal Valley, IL 61240), Dusty Lombardi, Jeremy Carpenter, and Troy Mueller (Columbus Zoo and Aquarium, 4850 West Powell Rd., Powell, OH 43065)

The aerial audiograms of two captive adult, male Arctic fox (Alopex lagopus) were measured using a two-alternative, forced-choice paradigm and descending staircase method of signal presentation at the Columbus Zoo and Aquarium. Both fox displayed a typical mammalian U-shaped audiometric curve, with a functional hearing range of 125 Hz to 16 kHz (sensitivity < 60 dB re: 20 μPa) and average peak sensitivity of 22 dB re: 20 μPa at 4 kHz. This range is similar to airborne hearing thresholds previously measured for two closely related species, the kit fox (Vulpes marotis) and the domestic dog (Canis familiaris). There was little variability around threshold and no significant difference between the hearing curves of the two fox. The peak sensitivities of the Arctic fox overlaps their vocal range, the vocal range of their prey; field voles (Microtus agrestis), collared lemmings (Dicrostonyx groenlandicus), Canada geese (Branta canadensis), and the hearing range of a predator; the polar bear (Ursus maritimus). Ambient noise levels were monitored and test frequencies below 5 kHz were possibly masked. Potential response bias was examined using Receiver Operator Characteristic analysis and had a conservative bias. Masking effects and conservative response bias may have resulted in slightly underestimated hearing curves.

Session 5aBA

Biomedical Acoustics and Physical Acoustics: Acoustic Microscopy Imaging Methods for Biomedical Applications I

Jonathan Mamou, Cochair
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Tadashi Yamaguchi, Cochair
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Invited Papers

9:20

5aBA1. Measurement of acoustic properties of tissues using ultrasonic tissue imaging system. Hiroyuki Hachiya (Graduate School of Science and Engineering, Tokyo Institute of Technology, 2-12-1 S5-17 Ookayama, Meguro-ku, Tokyo 152-8550, Japan, hachiya@ctrl.titech.ac.jp), and Tadashi Yamaguchi (Chiba University, Chiba, Japan)

Acoustic properties of living tissues are an important parameter for quantitative estimation of the tissue structure. It is very important to determine the relationship between the physical and the chemical change of tissue structure and the change of acoustic properties. We have developed ultrasonic tissue imaging system (UTIS) to measure the special distribution of acoustic properties at frequencies form 3 MHz to 50 MHz. The speed and attenuation of sound were measured by the noncontact technique based on frequency-time analysis of a reflected pulse response. From rat tissues measurement, The speed of sound in normal tissues varied minimally between individuals and was not related to body weight or age. The relationship between speed of sound and density in normal, fatty and cirrhotic livers can be fitted well on the line which is estimated using the immiscible liquid model assuming a mixture of normal liver and fat tissues. We have also measured the distribution of acoustic properties in human tissues, and obtained relationship between the sound speed and the attenuation of tissue. These findings of the relationship between tissue structure and acoustic structure are used for development of a quantitative diagnosis technique.

9:40

5aBA2. High resolution biomedical imaging—multimodal ultrasound microscope and combination with optics. Yoshifumi Saijo (Tohoku University, 980-8575, saijo@idac.tohoku.ac.jp)

High frequency ultrasound imaging has evolved from classical acoustic microscopy to the multimodal ultrasound microscope which is available for quantitative C-mode, surface acoustic impedance mode and 3D-mode imaging. The evolution has realized both quantitative parametric imaging and easier observation. Quantitative C-mode representing two-dimensional distribution of attenuation or sound speed is realized by frequency-domain analysis of a single pulse by a high speed digitizer. Because the square of sound speed is proportional to the tissue elasticity, sound speed imaging provides biomechanical information of the tissues which is especially important in cardiology and orthopedic surgery. The data also help understanding clinical echo features, especially important in grading of liver fibrosis. Surface acoustic impedance mode without thin-slicing has been applied for imaging of fresh brain tissues and real time observation of high-intensity focused ultrasound procedures. High frequency 3D-mode imaging has visualized 3D structure of sebaceous gland in dermis. Compared with optical coherence tomography which provides higher resolution imaging, ultrasound is superior in the penetration depth and assessment of tissue elasticity. Photoacoustic imaging provides not only morphology but also small blood flow distribution. Both ultrasound and optical methods should develop together to realize high resolution and "gentle" biomedical imaging.

10:00

5aBA3. Charactarization techniques of particular proteins in cerebellar cortex using acoustic impedance microscope. Sachiko Yoshida, Ryoichi Minowa, Shiho Masaki (Toyohashi University of Technology, Toyohashi 441-8580, Japan, syoshida@ens.tut.ac.jp), Seiji Yamamoto (Hamamatsu University School of Medicine, Hamamatsu 431-3192, Japan), Kazuto Kobayashi (Honda Electronics Co. Ltd., Toyohashi 441-3193, Japan), and Naohiro Hozumi (Toyohashi University of Technology, Toyohashi 441-8580, Japan)

Two-dimensional acoustic impedance imaging is useful for observation of living organs without invasion. Because acoustic impedance is proportional to sonic speed and density, organelle having larger density, e.g. nucleus, showed higher impedance. We have proved that cerebellar cortical layers and Purjinje cell bodies were identified using the acoustic impedance microscopy. In order to visualize the distribution of specific functional proteins in acoustic imaging, we proposed direct or complex-including heavy metal treatment to elevate the density and acoustic impedance of a particular protein. Heavy metal binding was useful for acoustic impedance imaging; however, metal binding to a protein molecule was not always specific. To observe the distribution of wanted molecules, we investigated two types of heavy metal binding materials; one was *p*-cymene ruthenium (Ru)-binding calcium channel binding peptides, and another was cadmiun nanocrystal binding antibodies, Qdot^{TM}. We could observe the characterized acoustic area by both *p*-cymene Ru-binding peptide treatment and Qdot treatment, whereas specimen plate was required hydrophobic condition to stabilize acoustic impedance. We suggest that a metal-binding reagent would be useful for specialization of an acoustic imaging.

10:20

5aBA4. Quantitative ultrasound microscopy imaging of cells. William OBrien, Thomas Auger, Aiguo Han, and Lauren Wirtzfeld (University of Illinois, wdo@uiuc.edu)

Using high-frequency ultrasound, two approaches were used to quantify eukaryotic cell properties. One approach (10-100 MHz) is model-based wherein live cells of known number and volume density are placed in a mixture of bovine plasma and thrombin to form a clot. Backscatter coefficient estimates are modeled against a concentric sphere scattering model to yield cell and nucleus diameters as well as density and speed of cytoplasm and nucleus of Chinese hamster ovary (CHO) and 3T3 fibroblast cells. Estimated cell and nucleus diameters were consistent with direct light microscope measures [CHO cell: 13 microns, nucleus: 6.6 microns and 3T3 cell: 23 microns, nucleus: 13 microns]. For CHO cells, both density and speed of the cytoplasm were less than those of the nucleus. For the 3T3 cells, both density and speed of the cytoplasm were not significantly different from those of the nucleus. The other approach (65-170 MHz) is a direct through-transmission round-trip measure (RF echo reflected from strong reflector), yielding estimates of attenuation and speed of cytoplasm and nucleus of MAT B III mammary adenocarcinoma cell. Speed of cytoplasm was not significantly different from that of the nucleus. The attenuation of the cytoplasm and nucleus were 0.61-0.77 and 0.94 dB/cm-MHz, respectively. [Supported by NIH R01CA111289; l'Ecole Centrale de Lille for TA; Canadian National Science and Engineering Research Council for LAW]

10:40–11:00 Break

11:00

5aBA5. Ultrahigh frequency ultrasonic transducers. K. Kirk Shung (Univ of Southern California, Department of Biomedical Engineering, Los Angeles, CA 90089, kkshung@usc.edu)

Acoustic microscopy and ultrahigh frequency (UHF) ultrasound microbeam applications require high performance single element transducers from a few hundred MHz to a few GHz. Design and fabrication of such transducers differ drastically from lower frequency transducers. A top- down approach i.e. lapping, grinding and polishing a bulk piezoelectric material to the desirable thickness no longer works. Instead a bottom-up approach i.e., depositing thin layers of a piezoelectric material is preferred. Although ZnO has been the piezoelectric material of choice for many years because it can be easily sputtered in thin films onto a substrate, it however suffers from a low electromechanical coupling coefficient. Better design and piezoelectric materials are sought to enhance the performance of UHF transducers. Thin film PZT, PMN-PT and other novel materials have been shown to offer superior performance. Lensless design may be an alternative to that used in conventional acoustic microscopic transducers.

11:20

5aBA6. Optical scanning photoacoustic microscopy with acoustic near-field detection. Hsin-Yu Chen (Dept. of EE, National Taiwan University, No. 1, Sec. 4, Roosevelt Road, Taipei, Taiwan 106, b95901167@ntu.edu.tw)

Photoacoustic microscopy (PAM) can be used to provides high-resolution microscopic images with contrast determined by optical absorption differences. Nevertheless, contemporary PAM suffers from low image acquisition speed, and the resolution depends crucially on accuracy of confocal alignment. To overcome these problems, we propose near-field acoustic detection, with which image resolution is solely determined by laser spot size and a MEMS based optical mirror for rapid 2D scanning. Phantom studies have been conducted to characterize performance of this prototype device. With a hair phantom, the lateral resolution is assessed at 14.0um. Further improvement is possible if the fiber-lens coupling can be made more precise. The axial resolution approximates 50um, which corresponds to the wavelength of a 30MHz ultrasound frequency in water. Relationship between the photoacoustic signal amplitude and absorption coefficient of the image object is approximately linear, with a 0.95 correlation coefficient after first-order regression fit. Noise equivalent pressure(NEP) reaches 49.11(Pa), which is comparable to other PAM systems. For high speed image data acquisition, we used a MEMS optical scanning mirror. The commercial scanning mirror raster-scans the laser beam across 130+ pixels per second. This design not only increases frame rate, but reduces the overall device dimensions. In addition to typical applications such as microscopic imaging of biological tissues, we demonstrate the potential of this device in nondestructive testing by inspecting the surface quality of carbon based glucose test strip, a disposable component in glucose meters.

11:40

5aBA7. Ultrasonic C-mode imaging with phase conjugation. Masahiro Ohno (Chiba Institute of Technology, 2-17-1 Tsudanuma, Narashino, Chiba 275-0016, Japan, ohno.masahiro@it-chiba.ac.jp)

Phase conjugation, or Time-Reversal equivalently, is a technique that produces retro-propagating waves for arbitrary incident waves with their wavefront structures preserved. We have been working to construct a C-mode ultrasonic imaging system that incorporates phase conjugation procedures. This system is composed of a conventional C-mode imager and a phase conjugator working via parametric w-2w interaction in a nonlinear piezoelectric medium. In this paper, some observation results by this system at an ultrasonic frequency of 10 MHz, mostly for thick soft biological samples, are presented. In the transmission imaging mode, images by phase conjugation showed the ultrasonic attenuation distributions that are approximately coincident with the real tissue structures, whereas conventional C-mode images for the same samples showed more complex patterns that reflected both the shape and the attenuation factors of the sample. Some simpler-structured artificial samples made of agarose gels were also tested to study the essential features of this phase conjugate image-forming method.

12:00

5aBA8. The life and death of single cells: an acoustic microscopy investigation. Eric Strohm, Maurice Pasternak, and Michael Kolios (Ryerson University, 350 Victoria Street, Toronto, M5B2K3, Canada, estrohm@ryerson.ca)

Changes in cell structure as tumours respond to cancer therapies can be detected using high frequency ultrasound (20-60 MHz). However, it is unknown how single cell variations influence the ultrasound signal as these ultrasound frequencies cannot resolve individual cells. Acoustic microscopy uses ultra-high frequency ultrasound (over 100 MHz) to quantify the structural and mechanical properties of single cells. The backscattered echoes from the cell membrane and substrate were used to determine the thickness, sound speed, density, bulk modulus and attenuation of breast cancer cells undergoing various biological processes, such as mitosis (cell division) and apoptosis (programmed death). Moreover, measurements were made to examine differences between malignant and benign breast tumour cells. Statistically significant changes in the thickness, sound speed and bulk modulus (2%), and attenuation (60%) between malignant and benign cells, and thickness (70%) and attenuation (21%) between cells undergoing mitosis and apoptosis were observed. Differences in bulk modulus (4%) and attenuation (30%) were observed between early and late stage apoptosis, which can be attributed to the irreversible changes in cell structure during cell death. In summary, acoustic microscopy can be used to probe single cells and help understand their contribution to ultrasonic changes in tumours responding to chemotherapy treatments.

Contributed Paper

12:20

5aBA9. Experimental and theoretical characterization of high frequency polymer ultrasound contrast agents. Pavlos Anastasiadis (Dept. of Mechanical Eng, 2540 Dole Street. Holmes 302, University of Hawaii-Manoa, Honolulu, HI 96822, pavlos@hawaii.edu), Parag Chitnis (Lizzi Center for Biomedical Engineering, Riverside Research, 156 William St., New York, NY 10038), Sebastain Brand, Georg Lorenz (Fraunhofer Institute of Material Mechanics), Jeffrey Ketterling (Lizzi Center for Biomedical Engineering, Riverside Research, 156 William St., New York, NY 10038), and John Allen (Dept. of Mechanical Eng., 2540 Dole Street, Holmes Hall 302, University of Hawaii-Manoa, Honolulu, HI 96822)

Lipid-shelled contrast agents have been the focus of combined experimental and theoretical efforts in order to characterize their dynamics with respect to shell material parameters. The use of independent mechanical measurements such as atomic force microscopy (AFM) for the shell materials has begun to receive attention. Polymer-shelled agents are less understood in terms of behavior and their destruction scenarios are exemplified by complex buckling modes. We present ultra-high frequency (GHz) scanning acoustic microscopy (SAM) measurements of polymers (Philips Healthcare, Best, The Netherlands) with various shell elastic properties and their corresponding thicknesses. Acoustic microscopy offers unique advantages in terms of non-invasive microscale measurements and visualization of the agents. The obtained material parameters are used in numerical simulations to predict pressure thresholds for the onset of the sub-harmonic response.

5a FRI. AM

Session 5aEAa

Engineering Acoustics, Underwater Acoustics, and Biomedical Acoustics: Acoustic Sensors and Actuators II and Civil Non-Destructive Testing with Ultrasound or Other Non-Contact Methods II (Poster Session)

Michael Scanlon, Cochair
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Contributed Papers

All posters will be on display from 9:20 a.m. to 10:40 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:20 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 10:40 a.m.

5aEAa1. A prototype of total-internal-reflection type of microphone using optical fibers. Yasushi Suzuki (Gunma National College of Technology, 580 Toriba-cho, Maebashi-shi, Gunma, Japan, suzuki@elc.gunma-ct.ac.jp), and Ken'iti Kido (Professor Emeritus of Tohoku University, 543-1-504 Niiharu-cho, Midori-ku, Yokomaha, Kanagawa, Japan)

This paper describes a new type of microphone which uses the total internal reflection at the curved interface between glass and air. The critical angle for total reflection changes by the refractive index of air, which depends on the air density. As the density changes by the sound pressure, the sound pressure is measurable by detecting the intensity of the reflected light from the total reflection area. The acoustical characteristics of a prototype of the proposed microphone using optical fibers as an optical transmission path are investigated experimentally. The experimental results show that although the sensitivity of the microphone is low in the present stage, it can be used for the measurement of great volume of sound. The microphone has no limitation in the frequency range in principle because the mechanical vibration is not used.

5aEAa2. An airborne acoustic fiber sensor using fiber Bragg grating with wavelength division coupler demodulating method. Jin Cheng, Jie Feng, and Longjiang Zhao (The Third Research Institute of China Electronics Technology Group Corporation, Beijing 100015, China, nanocheng@163.com)

In this paper, we report an airborne acoustic fiber sensor. It consists of a broadband ASE light source, a circulator, a fiber Bragg grating (FBG) with a central reflecting wavelength of 1550nm, a round nickel membrane with a diameter of 9mm and a thickness of 3μm, and a wavelength division multiplexed (WDM) coupler demodulating module. The fiber Bragg grating is bonded to the nickel membrane along the central line with epoxy. The principles of the fiber sensor are described as follows: the sound wave causes the membrane to vibrate and the FBG bonded to the membrane to be bent. The shifts of the central reflecting wavelength caused by FBG bending are modulated by sound waves. The shifts are demodulated by a WDM system with an acoustic signal processing unit. The fabricated fiber sensor shows some advantages, such as water-proof character (about 2m deep under water), easy extending to arrays and easy fabrication procedures. It has widely potential applications in acoustic detection fields.

5aEAa3. Efficiency analysis on a piezoelectric micro-machined ultrasonic transducer as a high intensity wave generator in air. Haksue Lee (Agency for Defense Development, P.O. Box 18, Jinhae, Changwon, Republic of Korea, swallowtail@add.re.kr), Yub Je, Wonkyu Moon (POSTECH, Pohang, Republic of Korea), Young-Nam Na, Hee-Seon Seo, and Jooyoung Hahan (Agency for Defense Development, P.O. Box, 18 Jinhae, Changwon, Republic of Korea)

It is difficult to efficiently produce high-intensity acoustic/ultrasonic waves in air with a conventional piezoelectric transducer because of the huge acoustic impedance mismatch between solid-state transducers and air. In this work, the mechanoacoustic efficiency of a thin-plate flexural mode transducer is analytically compared with that of a conventional 1/4λ thickness mode vibrator. Radiation and internal mechanical quality factors are applied in the analysis. In the case of the thickness mode piezoelectric vibrator, the radiation quality factor does not depend on design factors, but only on material properties. Consequently, the mechanoacoustic efficiency of the thickness mode vibrator depends only on the material properties, and is less than 3% for most piezoelectric ceramics. For a thin-plate flexural mode transducer, the radiation quality factor can be controlled by the aspect ratio of the thin-plate, which is one of design parameters. Theoretically, the mechanoacoustic efficiency of the flexural mode transducer can be designed to be nearly 100% at the resonance frequency. By experimental analysis, the mechanoacoustic efficiency of the micro-machined transducer was about 65.9%, and the overall electroacoustic efficiency was 58.4% in the resonance. The transducer arrays designed based on this analysis have been successfully applied in parametric array applications in air.

5aEAa4. Theoretical analysis on shear horizontal polarized surface acoustic waves propagation in periodic metal grating using the variational method. FangQian Xu (Zhejiang University of Media and Communications, xufangqian2005@163.com)

A new theoretical method is presented to analyze dispersion characteristics of SH-type SAWs propagating on periodic metallic grating structures using the variational theory and coupling-of-modes (COM) equations. The characteristics of SH-type SAWs propagating in short-circuited Al gratings on various crystals are studied and the calculated results agree well with those of typical Hashimoto's method. Also, some involved problems like dispersion parameters of SH-SAW along the ST-90oX quartz with Al

electrode were first described. The advantages of this method are that the complicate Green's function is avoided, resulting in simple, clear and fast theoretical analysis.

5aEAa5. Ray tracing of acoustic wave in the lossy atmosphere. Yang Song, Chen Zhou, Zhengyu Zhao (School of Electronic Information, Wuhan University, phsongyang@yahoo.cn), and Yuannong Zhang

An acoustic ray tracing model considering the real atmospheric acoustic attenuation is developed base on the equation of the acoustic local dispersion relation in the stratified atmosphere. The acoustic attenuation coefficient and growth factor in the moving atmosphere are deduced from the imaginary part of the dispersion relation, and the acoustic attenuation coefficient is corrected by using the theory of attenuation in the real atmosphere. The ray equations in the lossy atmosphere are obtained through Hamilton equations. The numerical study of this ray tracing model shows that the absorption of sound could have some influence to the acoustic trajectory. The influence could be neglected in the near field propagation but not in the far field.

5aEAa6. Design and fabrication of a high frequency annular array for medical ultrasound systems. Jue Peng, Zhenhua Hu, Hu Tang, Tianfu Wang, and Siping Chen (School of Medicine, Shenzhen University, erica@szu.edu.cn)

Most high frequency (>15MHz) medical ultrasound systems are based on single element transducers mechanically scanned. These systems can provide images with excellent resolution. However, single element transducers are often limited by the fixed focal point and small depth of field. High frequency medical ultrasound annular arrays consisting of concentric rings of elements are focused electronically. These arrays are desirable to avoid the fixed focal point of the single element transducers and improve the depth of field. This paper demonstrate an over 10MHz annular array transducer with novel piezoelectric single crystal. This transducer exhibits good energy conversion performance with a very low insertion loss at the center frequency. The transducer is promising for intravascular ultrasonic imaging and other applications. This work was supported by the National Natural Science Foundation of China (Grant Nos. 10904093 and 61031003), the Science and Technology Grant Scheme funds from Shenzhen Government (No. 08CXY-23).

5aEAa7. Measurement of vibration distribution on focusing source using a thin rod reflector. Hirokazu Yanagisawa (Tokai University, Shimizu-ku, Shizuoka 424-8610 Japan, 1boum005@mail.tokai-u.jp), Jung-Ho Kim (GW Corporation, Ohkubo, Shinjuku-ku, Tokyo 169-0072), and Shigemi Saito (Tokai University, Shimizu-ku, Shizuoka 424-8610, Japan)

Utilizing the concept of time reversal and the transmitter-receiver reciprocity, the distribution of the vibration amplitude and phase is estimated for a focusing source without using a hydrophone. When a tone burst wave is radiated in water, the sound wave reflected from the top of a thin rod is detected with the concave source itself. Scanning the reflector surface in the plane normal to the acoustic axis, two-dimensional data of the amplitude and phase of the output voltage are stored and then the calculations of square root and time reversal are executed for the data. Comparing with the case detecting the sound with a needle type hydrophone set at the same position as the reflector, the advantage and disadvantage of the present method are discussed taking a piezoelectric concave source with a star-shape electrode for instance. The experiment is also carried out for the high-frequency concave-source with an aperture radius of 4 mm which is too small to measure with an ordinary hydrophone.

5aEAa8. Sound source distance estimation using a small-size microphone array. Satoshi Esaki (Nagoya University, 464-8603, satoshi.esaki@g.sp.m.is.nagoya-u.ac.jp), Takanori Nishino (Mie University, 514-8507), and Kazuya Takeda (Nagoya University, 464-8603)

A method for estimating the sound source depth, i.e., the distance between a source and receiver, using a small-size array is proposed. The proposed method uses the spatial distribution pattern of quasi-independent signal components obtained by the frequency-domain independent component analysis (FDICA) as the cue for depth estimation. The quasi-independent components are calculated by applying FDICA to array signals with very high redundancy, for example, 60 microphone signals for a pair of sources; therefore, signal components associated with reflection signals are obtained even though they are correlated with the direct signal. Experimental evaluation using a small-size microphone array with a large number of elements confirms that the average (RMS) estimation error of the proposed method is 0.33 m, which is sufficiently accurate for our applications.

5aEAa9. Precise simulation of surface acoustic wave devices using frequency-dependent coupling-of-modes parameters. Hao Wang (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, whsdjn45857427@163.com), Weibiao Wang (The 55th Research Institute of China Electronic Technology Group Corporation, Nanjing 210016, China), Haodong Wu, Bo Su, and Yongan Shui (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China)

Surface acoustic wave (SAW) devices have been extensively applied as the core components of modern mobile communication systems. In the fierce competition, developing precise and fast simulation models is very important. The widely-used coupling-of-modes (COM) model has been improved for many times, but it is still not satisfied yet. The fatal problem of extracting frequency-dependent COM parameters is that the reflection coefficient and the propagation velocity could not be obtained independently. In this work, a new method to evaluate all the frequency-dependent COM parameters is proposed. Using the FEM/BEM tool, the field distributions (including mechanical displacements and potential distribution) of forward and backward surface acoustic waves within periodic gratings of finite length could be calculated at every frequency. From the characteristics of the field distribution curves, all the COM parameters are evaluated as functions of frequency, and in particular, the reflection coefficient and the propagation velocity are extracted independently. Using these frequency-dependent COM parameters, the one-port resonators on the substrates of 128° YX-LiNbO3, ST-Quartz and 42° YX-LiTaO3 are simulated, which show good agreement with the results by FEM/BEM. The results verify the validity of the method. This work is supported by Natural Science Foundation of China under Grant No. 10774073 & 11174143.

5aEAa10. The temperature range limitation of passive, wireless monitoring by surface acoustic wave device. Hao Wang, Haodong Wu, and Yongan Shui (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, whsdjn45857427@163.com)

There were many studies devoted to SAW passive, wireless temperature monitoring. There are two ways to implement SAW passive, wireless temperature monitoring: through time domain and through frequency domain. Most researches utilized the time domain approach, on behalf its accompanied identification function. For time domain approach, it is necessary to use the phase information for judging the temperature so as to reach enough precision. However, in order to have an exclusive measuring value, there exists a temperature range limitation, which is related to the error for used system, required reliability, the permitted frequency band, and the needed temperature accuracy. It is important to note it when people want to monitor an object in a wide temperature range. In this paper, the relation among the temperature range and the error, required reliability, the permitted frequency band, and the needed accuracy is discussed. Furthermore, a novel method is raised to increase the temperature range under the same conditions. This work is supported by Natural Science Foundation of China under Grant No. 11174205.

5aEAa11. Detection of second harmonic components of Lamb waves generated from fatigued magnesium plate. Makoto Fukuda and Kazuhiko Imano (Deparment of Elctrical and Electronic Engineering, Akita Univ., 1-1 Tegata Gakuen-machi, Akita, Japan, mfukuda@gipc.akita-u.ac.jp)

Detections of second harmonic components generated from micro cracks created by fatigue tests using finite amplitude Lamb wave were carried out using double-layered piezoelectric transducer (DLPT) with pulse-echo method. Three pure magnesium (Mg) plates applied to fatigue test were used in this experiment. In received waveforms from the plate had fatigue test of 100,000 and 200,000 cycles, second harmonic components were increased by 6 dB and 10 dB, respectively, compared with in the plate had 0 cycles. Their waveforms and spectra were captured in real time by our DLPT system. Detection of second harmonic component is resulted in the useful tool in evaluating defect in the material.

5aEAa12. The use of parametric array for measuring absorption coefficient of sound absorbing material. Zheng Kuang and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, kuangzheng06@hotmail.com)

The measurements of the absorption coefficient of sound absorbing materials are mainly using the standing wave tube or the reverberant room. Both of them are not suitable for measurement in situ. Based on the high directional property of the self-demodulated sound from parametric array, the absorption coefficients of the material can be measured in situ. However, the spurious noise signals generated by nonlinear interaction between the high intensity primary sound and the microphone reduce the accuracy of the measurement results especially at the low frequency range. To solve the problem, phase-cancellation method is introduced into the measurement. The transmitters of parametric array are divided into two parts radiating the primary sound with opposite phases. Experimental results demonstrate that the method decreases the primary sound significantly along the axial direction in the near-field with little influence on the audible sound field, which can satisfy the requirement of the measurement.

5aEAa13. Finite element simulation of Lamb wave scattering at delamination damage in composite laminates. Haiyan Zhang (149 Yanchang Road, School of Communication and Information Engineering, Shanghai University, Shanghai 200072, hyzh@shu.edu.cn)

Due to the anisotropy and multilayer characteristics of composite laminates, the analytical solution for Lamb wave scattering at delamination damage do not exist. This study explores the effectiveness of the 2D finite element method (FEM) to model Lamb wave scattering at a delamination which is across the full width of the composite laminates. The capability of 2D FEM to simulate Lamb wave scattering characteristics at delaminations with different sizes by means of contour snapshots at different time instances and the scattering directivity pattern in different laminated location are illustrated. The results suggest that 2D FEM can be used to investigate Lamb wave propagation and their scattering characteristics at delamination damage, which help to validate and improve the performance of Lamb wave damage detection configuration.

FRIDAY MORNING, 18 MAY 2012 — S221, 11:00 A.M. TO 12:40 P.M.

Session 5aEAb

Engineering Acoustics, Underwater Acoustics, and Biomedical Acoustics: Acoustic Sensors and Actuators III

Michael Scanlon, Cochair
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Zhushi Rao, Cochair
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Yichun Yang, Cochair
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Invited Papers

11:00

5aEAb1. Development of PZT piezoelements for simultaneous generation and reception of longitudinal and shear ultrasonic waves. Rymantas Kazys (Ultrasound Institute, Kaunas University of Technology, Studentu str. 50, Lithuania, rkazys@ktu.lt), Algirdas Voleisis, and Egidijus Zukauskas

Ultrasonic methods are often used for measurement of various non-electric quantities, for example, pressure or displacements. In this case in order to increase accuracy of measurements longitudinal L and shear SH ultrasonic waves simultaneously propagating in solids along the same path must be used. For simultaneous generation/reception of L and SH waves dual mode PZT transducers may be used, but up to now only theoretical studies are known with a little experimental confirmation. A dual mode piezoelement can be manufactured as a rotated Z-cut element from a large thick PZT block. Objective of our study was investigation of transient processes in dual mode PZT piezoelements, optimization of their geometry and development of manufacturing technology. Vibrations of rotated Z-cut disc and rectangular shape piezoelectric elements in a pulse mode were investigated by a numerical modeling using ANSYS finite elements code. The volume of the disc was meshed using the SOLID5 elements. Modeling results were verified by experiments using steel measurement bodies and a good correspondence has been obtained. At the angle θ=36° L and SH waves of similar amplitudes are excited.

11:20

5aEAb2. Radially composite cylindrical piezoelectric ultrasonic transducers. Shuyu Lin, Shuaijun Wang, and Zhiqiang Fu (Applied Acoustics Institute, Shaanxi Normal University, sylin@snnu.edu.cn)

A new type of radially composite cylindrical piezoelectric ultrasonic transducers is presented and its radial vibration is studied. The composite transducer is composed of a radially polarized piezoelectric ceramic short tube with arbitrary wall thickness and a metal tube. The radial vibrations of the radially polarized piezoelectric tube and the metal tube are analyzed and their electro-mechanical equivalent circuits are obtained. Based on the mechanical boundary conditions between the piezoelectric tube and the metal tube, the six-port electro-mechanical equivalent circuit of the radially composite ultrasonic transducer is obtained and the frequency equation is given. The theoretical relationship between the resonance/anti-resonance frequency and the effective electro-mechanical coupling coefficient with the ratio of the inner radius over the outer radius of the composite transducer is analyzed. At the same time, the radial vibration of the composite transducer is simulated by using Finite Element Method. The vibrational modal shape and the harmonic response are given numerically. At last, some radially composite ultrasonic transducers are designed and manufactured; their resonance/anti-resonance frequencies are measured. It is shown that the analytical resonance/anti-resonance frequencies are in good agreement with the numerically simulated and experimental results. It is expected that this type of radially composite ultrasonic transducers can be used in large scale ultrasonic liquid processing, such as ultrasonic extraction, ultrasonic sonochemistry and other applications where large radiation surface and ultrasonic power are needed.

11:40

5aEAb3. Comparisons between various cavity and panel noise reduction control methods in double-panel structures. Jen-Hsuan Ho (University of Twente, Drienerlolaan 5, P.O. Box 217, 7500 AE Enschede, The Netherlands, j.ho@utwente.nl), and Arthur Berkhoff (Department of Electrical Engineering, University of Twente, Drienerlolaan 5, P.O. Box 217, 7500 AE Enschede, The Netherlands; TNO Science and Industry, MON-Acoustics, P.O. Box 155, 2600 AD Delft, The Netherlands)

This paper presents comparisons between various panel and cavity resonance control methods to reduce the transmitted sound in a double-panel structure. The double-panel, which consists of two panels with air in the gap, has the advantages of low weight and effective transmission-loss at high frequency. Therefore, it is widely applied in many areas such as aerospace. Nevertheless, the resonance of the cavity and the poor transmission-loss at low frequency limit its noise control performance. Applying active control forces on the panels or utilizing loudspeakers in the cavity to reduce the noise problem have been discussed in many papers. In this paper, an acoustic-structure coupled model is used to investigate and to compare the transmitted sound reduction of various cavity and panel resonance control methods. The control performance comparison is based on the same stability control margins. Moreover, an adaptive control method is used in the system to further improve the control performance. Finally, experimental results will be presented and discussed.

Contributed Papers

12:00

5aEAb4. Two-dimensional analysis of FBAR by using FDTD method. Xiaoli Yu, Ming Cao, Zhongyong Luo, Xun Gong, and De Zhang (Key Laboratory of Modern Acoustics, Institute of Acoustics, Nanjing University, Nanjing 210093, China, yuxiaoli666666@163.com)

The film bulk acoustic resonator (FBAR) devices are lighter, smaller, cheaper and capable of dealing with larger power than SAW devices. Therefore, the FBAR technology is more competitive in satisfying the commands of modern mobile phone systems. In this paper, the finite-difference time-domain method is applied to analyze the two-dimensional thin-film bulk acoustic wave resonators. The piezoelectric governing equations and Newton's equation of motion are discretized to centered finite-differences in spatial and temporal domain. An electrostatic field simulator named ANSOFT Maxwell 2D is used to calculate the electric field intensity. The frequency domain electrical impedance characteristics of FBAR with different electrode thicknesses are calculated by using the Prony's method. From the simulation results, the optimal electrode thickness is suggested.

12:20

5aEAb5. Modeling and optimization of a shotgun condenser microphone. Ming-Sian Bai (No. 101, Section 2, Kuang-Fu Road, Hsinchu, Taiwan 30013, R.O.C., hoshenhsu@gmail.com)

This paper is focused on modeling and optimization of a shotgun microphone that is known to be a highly directional acoustic pick-up. A lumped-parameter model is established, with the aid of Zuckerwar's approach, to predict the frequency response of the condenser microphone. On the basis of the model, we use the simulated annealing ([SA]) method to optimize the microphone parameters, including the diaphragm radius, the diaphragm thickness, the air gap distance and the volume of back chamber, such that the sensitivity is maximized subject to a desired bandwidth. In our modified approach, the air gap resistance (R_a) and the back chamber compliance (C_{bc}) are used to calculate the D factor in Zuckerwar's model. To model the shotgun tube, T circuit and two-port network are utilized in formulating transfer matrices that is then converted to an impedance matrix representation. In addition, an array model is established to simulate the directional response. The results revealed that the shotgun microphone is highly directional at high frequencies, while the on-axis frequency response is influenced by the acoustic resonances of the tube. The simulation results suggest that the tube length should be greater than half of the wavelength, whereas and the spacing between holes should be less than half of the wavelength.

Session 5aEAc

Engineering Acoustics, Noise, and ASA Committee on Standards: Sound Quality Engineering

Klaus Genuit, Cochair
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Ozawa Kenji, Cochair
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Invited Papers

9:20

5aEAc1. Effects of audio reproduction methods on the sense of presence in audio-visual content. Kenji Ozawa, Masashi Obinata, and Yuichiro Kinoshita (University of Yamanashi, 4-3-11 Takeda, Kofu 400-8511, Japan, ozawa@yamanashi.ac.jp)

The sense of presence is crucial for evaluating audio-visual equipment. To clarify the effects of audio reproduction methods on the sense, two experiments were conducted under audio-only and audio-visual conditions. Twelve scenes were recorded with a high-definition video camera while their sounds were recorded using a dummy head. In the audio-only condition, the recorded audio signals were reproduced with headphones by five methods: binaural reproduction with and without the headphone transfer function calibration, binaural reproduction with the joint stereo coding, stereophonic reproduction in which head-related transfer functions from two loudspeakers to the both ears were convolved to the binaural signals, and diotic reproduction in which the left and right channel signals were superimposed. Twenty subjects evaluated each stimulus using a Likert scale. In the audio-visual condition, the same experiment was performed while video signals were reproduced with a 65-inch display. In the audio-only condition, the effect of reproduction methods was significant, i.e. stimuli with the three binaural reproduction methods were evaluated as being higher presence than those with the other two methods. In the audio-visual condition, however, the effect was less prominent. These results suggest that spatial information of audio signals was compensated by visual information. [Work supported by NICT]

9:40

5aEAc2. An experimental and analytical application of vehicle sound quality target cascading. Koo Tae Kang (Hyundai Motor Company, kanggood@hyundai.com)

To achieve pre-defined sound quality target in a cabin, the interior sound is decomposed into air-borne and structure-borne sound by using transfer path analysis. First, the air-borne sound target is cascaded to the target of each sound package component in the form of the transmission loss mainly in the analytical way. Second, for the structure-bone sound, the experimental noise transfer path analysis is extended to identify the sensitive structural path for specific frequency contents as opposed to the traditional methods with which the critical structural components are hard to identify. As a result, the vehicle sound target proved to be successfully cascaded to the component level target by applying aforementioned methods.

Contributed Papers

10:00

5aEAc3. Assessment of annoyance caused by different types of construction noises. Sung Chan Lee, Pyoung Jik Lee, and Jin Yong Jeon (Hanyang University, sungchan@hanyang.ac.kr)

In the present study, annoyance caused by diverse construction noises was evaluated through surveys and laboratory experiments. A survey with a total of 100 construction workers was carried out to investigate annoyance from construction noises at different construction phases. Then, a number of noises from machinery that were evaluated in the survey as highly annoying were recorded from construction sites in Korea. Recorded construction noises were classified into four groups: stationary, fluctuating, intermittent, and impulsive, according to the temporal, psychoacoustical and spectral characteristics of the noises. A laboratory auditory experiment was then performed in order to quantify the total annoyance caused by individual construction noise and multiple construction noises. From the experiment, synthesis curves were derived for the relationship between noise levels and the percentage of highly-annoyed (%HA) and the percentage of annoyed (%A) for the combined noise sources.

10:20

5aEAc4. A-weighting the equal loudness contours. Jeremy Charbonneau, Colin Novak, Robert Gaspar, and Helen Ule (University of Windsor, charbo6@uwindsor.ca)

The standardized equal loudness contours identify the non-linearities of the human auditory system using simple sinusoidal input signals. The graphical illustration of auditory performance trends provides a visual representation of these non-linearities with respect to both frequency and amplitude across the range of auditory perception. Metrics such as the A-Weighting filter approximate one generalized curve shape, in an effort to quantify measured values in a manner that represents the perception of the measured

sound. With the release of the ISO226:2003 version of the standard, the most recent version of the equal loudness contours provide an improved contour set with more refined shapes and steeper slopes. The purpose of this study is to investigate the performance of the A-weighting function compared to the updated curves of the equal loudness contours. Included is an examination and discussion of the appropriateness of the continued use of the existing A-Weighting filter. Given the overall un-hyperbolic shape and flattening of the equal loudness contours as the amplitude of the noise increases, the A-weighted results visually identify the areas of weakness associated with a constant filter approach. This visual examination easily identifies the strengths of each approach as well as the deviations from anticipated outcomes.

10:40–11:00 Break

11:00

5aEAc5. Comparative study of unsteady loudness models for mechanical and real sounds. Colin Novak, Helen Ule, Jeremy Charbonneau (University of Windsor, 401 Sunset Avenue, Windsor, ON, N8Y 4E5, Canada, novak1@uwindsor.ca), and Tomasz Letowski (U.S. Army Research Laboratory, 520 Mulberry Point Road / R-39, Aberdeen Proving Ground, MD 21005-5425)

While noise levels are most often quantified using physical quantities including A-weighted sound pressure level, these metrics do not adequately represent the human perception of the noise. For this, loudness is a more appropriate acoustic metric as it describes the perceived acoustic intensity of a sound. Given that real sounds are often unsteady, a most useful loudness calculation will also account for the perceptional phenomena of time and temporal masking. Studies have been done which demonstrate the performance of several loudness models for pure tone sounds and compare these results to the ISO 226 equal loudness curves. This investigation goes beyond that and evaluates the performance of two unsteady loudness models, the Glasberg and Moore model and the DIN 45631-A1 method, using mechanical and real life sounds. Through implementation ofa jury, the differences of the two loudness calculation methods are demonstrated by plotting the perceived loudness of the sounds for two different levels compared to two different reference levels.

11:20

5aEAc6. Proposed hybrid multiple look approach for calculating unsteady loudness. Helen Ule, Colin Novak, and Robert Gaspar (University of Windsor, 401 Sunset Ave., Windsor, ON, N9B 3P4, ule@uwindsor.ca)

Experimental studies have shown that for short gaps between 2 to 5 ms, the perceived loudness is higher than for uninterrupted noise presented to the ear. Other studies have also shown that the present temporal integration models for the calculation of time varying loudness do not adequately account for short duration phenomena. It has been proposed that the multiple look approach is a more applicable method for describing these short term circumstances. This approach breaks a sound into very small durations which allows for the intelligent processing of the looks and decision making depending on the nature of the stimulus. However, present technologies, such as the Fast Fourier Transform, are not adequate to deal with short duration sounds across the entire frequency spectra. A compromised approach using a proposed hybrid model is presented to account for perceived loudness levels for sounds in the presence of gaps while using an integration model. This hybrid multiple look model was tested using several sounds including mechanical and speech sounds and was found to perform as intended.

11:40

5aEAc7. Annoyance evaluation of interior vehicle motor noise by impulse response method. Sin-Yeob Lee and Buhm Park (Hanyang University, melonavel@hanyang.ac.kr)

In this study, the annoyance of EPS(Electronic Power Steering system) motor noise in a vehicle was evaluated through the anechoic chamber test. Evaluation during actual operation installed in vehicles allows in-situ annoyance tests, but requires a great deal of labor and time. Therefore anechoic chamber test is required to automate the evaluation process under consistent test condition. However vehicle noise test is significantly different from anechoic chamber test from different sound generation and propagation. Due to these discrepancies, the evaluated results among motors open differ between two test results. For consistent test results compared to those obtained from the actual vehicle environment, impulse response of both the vehicle interior and anechoic chamber was measured, and produced a simulated motor noise by using recorded signal in the anechoic chamber. The important peak noises at 1/3 octave bands to take into account of the tonal components were selected for evaluation of its noisiness. From the multiple regression model, the annoyance of motor noise in vehicle using the anechoic chamber test was evaluated.

12:00

5aEAc8. Development of a high quality wireless sound reinforcement system. Li Liu, Peng Zhang, and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, liuli7700@gmail.com)

This paper describes the development and initial application of a wireless sound reinforcement system. This system consists of wireless communication module, digital signal processor (DSP) and audio codec. Signal processing algorithms are used to achieve high quality wireless voice communications and audio playback. To this end, howling suppression algorithm is implemented using DSP. To date, there are three widely used methods for acoustic feedback control: phase-modulating feedback control (PFC), adaptive feedback cancellation (AFC), and notch-filter-based howling suppression (NHS). In this system, we used an adaptive notch filter (ANF) to perform feedback controlling. Simulation Results indicates the effect of this algorithm in suppressing howling. Furthermore, the control strategy of the algorithm is optimized by experiments in a conference room environment. The experimental results show the performance of this system in the real environment.

12:20

5aEAc9. The complexity of sound quality engineering—only a technical issue? Genuit Klaus (HEAD acoustics GmbH, Ebertstr. 30a, 52134 Herzogenrath, Germany, klaus.genuit@head-acoustics.de)

It is an absolutely essential task to enhance the perceived quality of technical products in order to stand out against competitors in times of a highly competitive market. It is clear that sound quality engineering has changed over time from basic level reduction needs to more complex sound quality design processes taking into account several psychoacoustic quantities. Today, it is very evident that sound level indicators are completely insufficient in describing and predicting perceived sound quality. Nevertheless, it is still an ambitious and challenging task to design product sound adequately, which indicates functionality, high quality, and corporate identity at the same time. To fulfil this task sound quality engineering work must change from a simple technical consideration to an interdisciplinary perspective, where knowledge from engineers, psychologies and sociologists is required. Sound quality is not an inherent product property, it develops when listeners are exposed to and interact with a technical product finally judging on the basis of their experience, expectation and context. Sound quality engineering has to consider these "confounding" variables. The paper will highlight the newest developments in the field of sound quality engineering of technical products. For it, experimental data collected with different test methods and its respective analysis results will be shown and discussed.

Session 5aNSa

Noise and ASA Committee on Standards: Environmental Noise and Regulations I

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Invited Papers

9:20

5aNSa1. Annoyance from railway vibration in residential environments: factors of importance when considering exposure-response relationships. Eulalia Peris, James Woodcock, Gennaro Sica, Andy Moorhouse, and David Waddington (The University of Salford, Acoustics Research Centre, M5 4WT, Salford, UK, E.Peris@edu.salford.ac.uk)

Railway induced vibration is an important source of annoyance in residential environments. Annoyance increases with vibration magnitude. However, these correlations between annoyance and physical ratings are weak. This suggests that vibration-induced annoyance is governed by more than just vibration level, and that simple exposure-response relationships alone sometimes do not provide sufficient information for understanding the wide variation in annoyance reactions. Results of investigations made on factors coming into play when considering an exposure-response relationship between level of vibration and annoyance are presented here. Examples of these factors are time of day, situational factors, personal and attitudinal factors. This was achieved using data from case studies comprised of face-to-face interviews (N=931) and internal vibration measurements collected within the study "Human Response to Vibration in Residential Environments" by the University of Salford. This work will be of interest to researchers and environmental health practitioners involved in the assessment of vibration complaints, as well as to planners and consultants involved in the design of buildings and railways. [Work funded by the Department for Environment, Food and Rural Affairs (Defra) UK]

9:40

5aNSa2. A local noise ordinance for relatively small community, population 16,000. Paul Schomer (Schomer and Associate, 2117 Robert Dr., Chammpaign, IL 61821, schomer@SchomerAndAssociates.com)

This paper presents a local noise ordinance developed for a relatively small community with a population of about 16,000. Key features are ease of enforcement by the use, 95 to 99 percent of the time, of simplified methods, with more detailed, rigorous procedures used only when needed (less than 5 percent of the time). When the more complex procedures are used, they are to be performed with qualified personnel and equipment that are external to the township staff. Use of sound level meters is minimized by employment of noise control techniques that depend on more easily measured factors than sound pressure level. Specifically, the ordinance includes such techniques as speed limits (for noise control), distance criteria, time of day, and the criteria "plainly audible," to reduce reliance on traditional measurements using a sound level meter. The ordinance includes enforcement procedures and qualifications for both the "every day" actions that should be 95 to 99 percent of all of the actions, and the more detailed and precise measurements procedures for the infrequent times that they are needed

10:00

5aNSa3. Characteristics of effective noise regulations. Nancy Timmerman (Nancy S. Timmerman, P.E., 25 Upton Street, Boston, MA 02118-1609, nstpe@hotmail.com)

In considering what is an "effective" noise regulation, the definition used will be that which actually reduces the noise regulated. Parameters which are expected to have an effect on the ability to reduce the noise include control over the noise source regulated, a way to measure results, and consequences or enforceability. Examples will be taken from community noise and airport noise regulations in the United States.

10:20

5aNSa4. Noise regulations and control policy in Taiwan. Li-chung Chou (EPA, eric@oe.com.tw), Chung-ho Yu, and Chien-wei Chen (Victory Scientech)

The number of noise complaint cases increased gradually every year in Taiwan. Refer to the annual report, the major noise sources were such as entertainment and business noise, following with noise from construction sites. The high population density and mixed residential and commercial zones caused a lot of noise complaint cases in urban areas. There were also a lot of neighbor noise complaints. In addition, traffic noise with no effective buffer zone caused complaints from the residents along the wayside. In order to solve the various types of noise problems, the Environmental Protection Administration proposed a noise control policy for short-term, mid-term and long-term. This paper will describe the noise control regulation structure and introduce the control policy in Taiwan.

10:40–11:00 Break

11:00

5aNSa5. A study of the effectiveness of the key environmental protection policies for road traffic noise control. Jiping Zhang (Zhejiang Research & Design Institute of Environmental Protection, 109 Tian Mu Shan Road, Hangzhou 310007, China, jpzhang@mail.hz.zj.cn), Paul D. Schomer (Schomer and Associates, Inc., 2117 Robert Drive Champaign, IL 61821), Maurice Yeung (Hong Kong Institute of Acoustics, China), Anguo Zhou, Hui Ming, Juan Chai, and Lu Sun (Zhejiang Research & Design Institute of Environmental Protection, Hangzhou 310007, China)

This paper introduces the history and current status of key environmental policies implemented for road traffic noise control in China, based in part on lessons learned both from international experiences and from China's own experiences. The general framework for traffic noise control in China is similar to what can be found in many foreign countries, and it has been playing an important role in environmental management of the rapidly-growing Chinese economy. The key international environmental policies for road traffic noise control in developed countries are very mature from the technological, economic, legal, and sociological standpoint, and these policies indicate a sharp conflict between road traffic noise and land uses along the road for distance up to about 200m from the road. But China faces greater limitations in using and enforcing these policies to control road traffic noise completely. This paper reconsiders and evaluates China's environmental policies for road traffic noise by collecting and comparing difficult cases of road traffic noise management in other countries to the Chinese experience so as to learn from these experiences now and minimize problems for the future. The research supported from the State Key Lab. of Subtropical Building Science South China University of Technology.

11:20

5aNSa6. Differences in responses to vibration induced in residential environments by railway and construction activities. James Woodcock, Eulalia Peris, Gennaro Sica, Andy Moorhouse, and David Waddington (University of Salford, Salford, M5 4WT, UK, j.s.woodcock@edu.salford.ac.uk)

This paper summarises the results of the Defra (UK) funded project 'NANR209: Human response to vibration in residential environments'. The main aim of this project was to develop exposure-response relationships for the human response to environmental vibration as experienced by residents in their own homes. The sources of vibration considered were railway, construction, and internal sources outside of the resident's control. In this study, 1431 questionnaires were completed with UK residents in their own homes to determine self reported annoyance. Measurements of vibration inside and outside residences were conducted to determine each resident's vibration exposure. Presented in this paper are the exposure-response relationships derived from these data indicating the percentage of people expressing annoyance above a given threshold for a given vibration exposure. In particular, this paper reports the differences in responses to vibration induced by railway and construction activities. [Work funded by the Department for Environment, Food and Rural Affairs (Defra) UK]

11:40

5aNSa7. Noise policy development in Italy and the EU. Gaetano Licitra (CNR-IPCF and ARPAT Tuscany, Environmental Protection Agency, g.licitra@arpat.toscana.it), and Elena Ascari (IDASC-CNR, Institute of Acoustics "O.M.Corbino")

Since the approval of European Directive 49/2005, in Europe many studies and developments have been done to improve quality and accuracy of noise mapping and action planning. With the end of the first round of agglomerations mapping, a comparison between mapping methods and exposure of cities is available. In Italy, tests have been done to compare procedures and results of different cities to understand how to improve quality and effectiveness of noise maps. In the meantime, EU Community worked to change calculation methods in order to find a procedure adaptable for each country but with a common structure. In fact, we have already seen that each country, also using an interim method has produced very different results. In particular, example within Italian maps will be shown and also between ones of different countries in Europe. The level assignment method to population will change too, in order to move towards a more realistic photograph of health conditions: maximum level was too preventive, when considering large building, so a new method has been taken from the German procedure such that each dwelling/room has his own level. Some example of the effects of this change will be evidenced in this paper.

12:00

5aNSa8. Study on the legislation and management system of community noise. Yiping Chu, Yude Zhou, and Wenying Zhu (Shanghai Academy of Environmental Sciences, 508 Qinzhou Rd., Shanghai, chuyp@saes.sh.cn)

Community noise is very common and numerous in urban life. According to the investigation in Shanghai, most of community noise is accidental, and its impact is different from industrial noise, traffic noise and construction noise. Measured by the one-hour or daily equivalent sound pressure level of current standard, the community noise will not exceed usually. On the other hand, based on the current Law of China on Prevention and Control of Environmental Noise Pollution, 'noise pollution' consists of two factors: one is to impact some person, and the other is to exceed the standard. To this extent, most of the community noise is not 'pollution'. This paper argues that the majority of complaints on community noise should be ranged to the area of social public management, and discusses the legislation and management system of community noise from following aspects: 1) collecting of the existing complaints on community noise; 2) study and investigation on domestic and overseas management system of community noise; 3) the connotative meaning of community noise' impact; 4) the position of community noise in urban management, and the management system be forwarded, including the objects of management, law-enforcing department, and the content of management, etc.

12:20

5aNSa9. Research of aircraft noise evaluation and limits. Wenying Zhu, Yude Zhou, and Yiping Chu (Shanghai Academy of Environmental Sciences, 508 Qinzhou Rd., Shanghai 200233, zwing258@yahoo.com.cn)

Along with the European Union to promote the day - and - night equivalent sound level DENL, a growing number of countries have adopted the equivalent sound level DNL or DENL instead of the traditional evaluation for airport noise evaluation, and in sensitive areas the maximum sound level limits was used, some also use the TALA (Time-above an A-weighted sound level threshold). China promulgated the "GB9660 - 88 measurement of aircraft noise around airport" and "GB9661-88 environment standard of aircraft noise around airport", rating for the weighted equivalent continuous perceived noise level WECPNL, and it is independent of other acoustic environmental standard. The state environmental protection department proposed to revise the criteria. In this paper, based on the large amount of measured data, under the aircraft noise statistical analysis, trying to build the relationship between WECPNL, Leq, Lmax; combined with the research results of relationships between noise level and annoyance; investigating the suitability of the current criteria (evaluation and limits), and explore the establishment of a set of aircraft noise evaluation method, which can not only reflect the characteristics of aircraft noise, but also better associated with living requirements.

FRIDAY MORNING, 18 MAY 2012 — HALL C, 9:20 A.M. TO 12:40 P.M.

Session 5aNSb

Noise: Ship Noise and Vibration I

Lin He, Cochair
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Changgeng Shuai, Cochair
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Contributed Papers

9:20

5aNSb1. Air spring vibration isolation technology for ship propulsion engine. He Lin, Xu Wei, and Shuai Changgeng (Institute of Noise & Vibration, Naval University of Engineering, Wuhan 430033, P.R. China; Science and Technology on Ship Vibration and Noise Laboratory, 430033, P.R. China, helin401@yeah.net)

Propulsion engine (PE) is one of the most dominant noise sources of ship. Due to the imposed requirement of keeping alignment with propulsion shaft during operation, the effective vibration isolation of PE using low frequency mount is difficult to implement as is often adopted by other onboard machinery. In this paper, a low frequency air spring vibration isolation system (ASVIS) with alignment control strategy for PE is conceived and introduced. The application of ASVIS to PE presents both advantages and challenges, which are discussed detailedly in the paper, as well as the feasibility of the ASVIS concept. A systematic design method of ASVIS for PE is established, with focus on the system mechanical behavior optimization and automatic alignment control algorithm development. An ASVIS prototype is designed and manufactured using the proposed method. The performance of the prototype is tested by a series of experiments, including alignment control precision and isolation efficiency. Experimental results show that using ASVIS, the vibration of PE can be attenuated to a satisfactory level, with the alignment between PE and shaft being maintained in the safe range.

9:40

5aNSb2. Experimental investigation of fluctuation characteristics caused by the interaction between turbulent boundary layer and the cavity on the trailing edge of cavity. Yezhen Pang and Mengsa Yu (China Ship Scientific Research Center, No. 222, East Shanshui Road, Wuxi Jiangsu 214082, China, chaos123@sohu.com)

The fluctuation characteristics caused by the interaction between turbulent boundary layer and the cavity on the trailing edge of cavity were investigated. Experiment results showed the fluctuation on the trailing edge of cavity is reinforced. Because the depth of cavity is smaller than the length of opening, the sheartone frequency will not approach the resonance frequency of the cavity, however the sheartone will reinforce the acoustic radiation of the structure downstream the cavity.

10:00

5aNSb3. The study of sono-elasticity of floating bodies in the acoustical environment of shallow sea. Ming-Song Zou (China Ship Scientific Research Center, No. 222, ShanShui East Road, BinHu District, WuXi, JiangSu, zoumingsong1234@yahoo.com.cn)

A concept combining sono-elasticity research field with oceanic sound propagation research field in developed in this paper. The paper formulates in details about how to transfer the sono-elasticity dynamic equations

together with oceanic sound propagation equations in theoretical model into related numerical program by using the PEKERIS neritic acoustic environment as an example. The further verification of this approach and related computing program, and analyses on characteristics of the structural vibration and acoustic radiation in neritic acoustic environment are also provided in the paper by analyzing the problem of a elastic spherical shell as an example. Keywords:Sono-elasticity; Sound propagation; Shallow sea; Spherical shell.

10:20

5aNSb4. Limitations caused by radiation damping and water viscosity on power delivered by ocean wave energy conversion devices. Amadou G. Thiam and Allan D. Pierce (Boston University, 110 Cummington Str., Boston, MA 02215, thiam@bu.edu)

Any generic ocean wave energy conversion device can be suitably "tuned," given various design constraints, to extract the maximum amount of power from a given incoming ocean wave. Just how large this maximum extracted power can be depends to some extent on the magnitudes associated with the various natural damping mechanisms that inhibit the driven oscillations of the device. Such mechanisms include (i) the excitation of ocean waves by the oscillating body and (ii) the friction-induced drag when the body's surface moves relative to the neighboring water. One can generally quantity the joint influence of these mechanisms with a "bobbing test," where an object is initially displaced downward and allowed to freely oscillate, the oscillations dying out exponentially. The authors' general experience is that such a decay typically corresponds to a Q of about 5. Simple analytical models are described to quantify the two mechanisms. For viscous damping, the applicable fundamental model is that which yields a friction force on a flat plate oscillating below a nominally stationary incompressible fluid. For the radiation damping, the theory uses the Green's function solution for a point volume source just below the free surface of an incompressible fluid under the influence of gravity.

10:40–11:00 Break

11:00

5aNSb5. Line-spectra extraction of ship-radiated noise based on harmonic wavelet. Lu Wang and Yuanliang Ma (Institute of Acoustic Engineering, Northwestern Polytechnical University, ylma@nwpu.edu.cn)

The ship-radiated noise line-spectra contain a large number of characteristic information of the ship, which has great significance for detection and classification of underwater targets. This paper presents a method for extraction of the line-spectra of the ship-radiated noise under strong background noise using harmonic wavelet transformation. Harmonic wavelet is a class of wavelet, which has specific expression for the box-like shape of spectrum and frequency-domain continuous distribution. These features make it more accurate in time-frequency positioning with no frequency leakage in the decomposition process which results in ultra-narrow-band and high resolution. Using harmonic wavelet transformation, the ship-radiated noise is decomposed into separate orthogonal frequency bands which are non-redundant and non-leaking. Based on this, the strong background noise can be suppressed and the line-spectra can be extracted in frequency domain and reconstructed in time domain successfully. Simulation and experimental results show that the harmonic wavelet transform is more effective in the background noise suppression, and its ability for weak line-spectra extraction is superior to Fourier analysis and FIR filter methods.

11:20

5aNSb6. Study on effect of dynamic lubricated bearing oil film stiffness on underwater structure acoustic radiation induced by propeller force. Jiayou Yang, Yipeng Cao, and Liaoyuan Li (College of Power and Energy Engineering, Harbin Engineering University, Harbin 150001, China, yangjiayou1986@163.com)

When propeller runs in the unsteady wake flow field, the exciting force can form in the propeller blades. Such propeller exciting force transfers from the shafting, bearing supporting to the structure, causes the underwater structure strong vibration and noise. The vibration of propeller-shafting-structure coupled system is one of the main sources of underwater structure radiated noise now. Different oil film stiffness will affect the input power transmission characteristics of coupled system severely, and then affect the vibration and noise analysis of underwater structure. In this paper, the finite element model of double cylindrical shell structure with shafting is built. The oil film stiffness is got by solving two-dimensional Reynolds equation based on finite difference method. The vibration and underwater noise radiation of the shell structure induced by propeller excite force are calculated by FEM/BEM. Finally, the effect of dynamic lubricated bearing oil film stiffness on underwater structure acoustic radiation induced by propeller force is analyzed.

11:40

5aNSb7. Study on hull vibration and underwater radiation noise induced by propeller excitation considering shafting alignment. Liaoyuan Li (College of Power and Energy Engineering, Harbin Engineering University, Harbin 150001, Heilongjiang, P.R. China, simple.life.lly@gmail.com), Yipeng Cao, and Wenping Zhang (College of Power and Energy Engineering, Harbin Engineering University, Harbin, Heilongjiang)

The recent increases in the number, size, speed and horsepower of commercial ships led the ocean ambient noise levels at low frequencies increased. Accordingly, the analysis of ship hull vibration control and noise reduction needs to be paid more attention now. In this paper, a finite element model of bulk carrier including shafting is built. Considering the shafting alignment, the vibration and underwater radiation noise characteristics of hull structure caused by propeller excite force are calculated. Finite element method (FEM) is employed to simulate the vibration response of the hull structure due to the propeller excitation in consideration of fluid-structure interaction. Then the outer surface of the finite element model is treated as a boundary element model. Finite element solutions of the hull surface vibratory velocity are further used as velocity boundary condition of the hull boundary element model for consequent underwater radiated noise calculation with boundary element method (BEM). In addition, the characteristics of transfer force and radiated sound power of hull structure are obtained. Comparing with the numerical results of considering shafting alignment and without, the effect of shafting alignment on propeller-induced ship hull vibration and underwater radiation noise is analyzed.

12:00

5aNSb8. A method for predicting the vibrations of underwater cylindrical shells by using the equivalent beam model. Tang Rui (Harbin Engineering University, Underwater Acousitc Building, Room 1205, tangrui@hrbeu.edu.cn), Li Qi (Harbin Engineering University, Underwater Acousitc Building, Room 301), and Shang Dejiang (Harbin Engineering University, Underwater Acousitc Building, Room 1206)

The vibrations of cylindrical shells with large length-to-radius radio are similar to those of beams in very low frequency band. One-dimensional beam theoretical model is only considered about the transverse displacement, neglecting the coupling effects of displacements in other directions. Only the first order beam-type modal frequency, not the other higher orders, of cylindrical shells can be predicted precisely by one-dimensional beam theoretical model. To solve this problem, an equivalent beam theoretical model is established, based on Euler-Bernoulli beam theory, in this paper. In this model, a cylindrical shell model is considered as a beam model with the same structural parameters and boundary conditions, the interaction between the structures and water is approximated to added mass. Different Young's modulus values have been searched to make the calculated results identical to those obtained by cylindrical shell theoretical model. The results show that the beam-type modal frequencies are mainly dominated by the length-to-radius radio for shells with length-to-radius radio $L/a>10$ and radius-to-thickness radio $a/h>20$, and the effect of the shell thickness to the modal frequencies can be neglected in such conditions. The equivalent Young's modulus curves for the first five order beam-type modal frequencies of cylindrical shells with different length-to-radius radio have been calculated.

12:20

5aNSb9. Mitigation of low-frequency underwater noise generated by rotating machinery on a mobile work barge using large tethered encapsulated bubbles. Kevin M. Lee, Mark S. Wochner (Applied Res. Labs., The University of Texas at Austin, Austin, TX 78713-8029, klee@physics.utexas.edu), and Preston S. Wilson (Mech. Eng. Dept. and Applied Res. Labs., The University of Texas at Austin, Austin, TX 78712-0292)

Collections of bubbles cause significant dispersion and attenuation of underwater sound at frequencies near the individual bubble resonance and can potentially be used to mitigate low-frequency anthropogenic underwater

5a FRI. AM

noise. Such effects have been reported for large encapsulated bubbles with resonance frequencies below 100 Hz [J. Acoust. Soc. Am. **130**:3325-3332 (2011)] and significant attenuation due to bubble resonance phenomena and acoustic impedance mismatching was observed in experiments using a compact electromechanical acoustic source [J. Acoust. Soc. Am. **128**:2279 (2010); J. Acoust. Soc. Am. **129**:2462 (2011)]. In the present study, screens of tethered resonant encapsulated air bubbles were used to surround a mechanically-vibrated mobile work barge in a lake experiment to demonstrate their potential as a mitigation strategy for such noise sources. Conventional screens of freely-rising bubbles were also deployed for comparison. The radiated acoustic pressure was measured at various water depths and ranges to determine the effect of the bubble screens on the noise field. Compared to the freely rising bubbles, the tethered encapsulated bubbles yielded a significant increase in noise reduction below 1 kHz, demonstrating their efficacy for abatement of low-frequency underwater rotating machinery noise. [Work supported by Shell Global Solutions.]

FRIDAY MORNING, 18 MAY 2012 HALL B, 9:20 A.M. TO 12:40 P.M.

Session 5aNSc

Noise and Architectural Acoustics: Noise Effects on Occupant Comfort and Performance in Buildings I

Lily Wang, Cochair
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C. M. Mak, Cochair
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Invited Papers

9:20

5aNSc1. The long-term effects of aircraft noise exposure on children's cognition: findings from the UK RANCH follow-up study. Charlotte Clark (Barts & the London School of Medicine, Queen Mary University of London, London, EC1M 6BQ, UK, c.clark@qmul.ac.uk), Jenny Head (University College London 1-19 Torrington Place London, WC1E 7HB, UK), and Stephen Stansfeld (Barts & the London School of Medicine, Queen Mary University of London, London, EC1M 6BQ, UK)

Many studies have demonstrated that environmental noise exposure at school is associated with poorer reading ability. However, studies have yet to determine the long-term consequences of environmental noise exposure during school for later developmental outcomes. This paper presents a longitudinal follow-up of the UK RANCH cohort assessing whether aircraft noise exposure in primary school is associated with poorer reading comprehension in secondary school. The original RANCH study found a relationship between aircraft noise exposure at primary school and children's reading comprehension for 9-10 year old children attending schools around London Heathrow, Amsterdam Schiphol and Madrid Barajas airports. Six years after the original study, 461 participants aged 15-16y (48% of the UK baseline sample) completed a standardised reading test during group testing in their classroom. Aircraft noise exposure at school was assessed using Leaq16 contour data. Analyses found no significant effect of aircraft noise exposure at primary school or secondary school on reading comprehension assessed at secondary school, although there was a trend for both types of noise exposure to be associated with poorer reading comprehension. The study is limited by its small sample but the findings indicate that aircraft noise exposure at school could have long-term implications for children's cognitive development.

9:40

5aNSc2. The effects of intelligibility and variability on cognitive performance and acoustic comfort. Andreas Liebl (Fraunhofer Institute for Building Physics, Nobelstrasse 12, 70569 Suttgart, Germany, andreas.liebl@ibp.fraunhofer.de)

Open-plan offices are very popular due to some assumed economic and organizational advantages. These benefits are confronted with typical drawbacks, i.e. office noise and lack of speech privacy. Ambient speech caused by conversations of colleagues is assumed to impair office workers' individual task performance at silent, concentrated work. The Speech Transmission Index (STI) is proposed to be a predictor of how much performance is reduced due to ambient speech in dependency of its intelligibility (Hongisto, 2005). Target values for the acoustic quality of open-plan offices were defined which account for the special importance of speech intelligibility (e.g. Virjonen et al., 2009). Fluctuation strength, a hearing impression due to slow modulations of amplitude or frequency, has also been shown to predict the loss of individual task performance (Schlittmeier et al. 2011). The relevance of both variables is systematically explored with regard to cognitive performance and acoustic comfort. Based on these results target values are discussed.

10:00

5aNSc3. Temporal and spatial characteristics of noise in train stations. Yoshiharu Soeta (Health Research Institute, National Institute of Advanced Industrial Science and Technology (AIST), 563-8577, y.soeta@aist.go.jp), and Ryota Shimokura (Nara Medical University, 634-8522, Japan)

The train noise in station (TNIS) can annoy passengers and reduce the speech intelligibility of the public address system in a station. Thus clarifying the acoustical characteristics of TNIS is important for the comfort and safety. Noise level was evaluated the equivalent continuous sound pressure level (LAeq). The sound diffuseness was evaluated by the interaural cross-correlation coefficient (IACC)

from the interaural cross-correlation function. The pitch and pitch strength was evaluated by the delay time (t1) and amplitude (f1) of the first peak of the autocorrelation function. The LAeq and IACC of the TNIS in the underground station were higher and lower, respectively, than those in the ground station. The t1 and f1 of the TNIS in the underground station were stable and lower, respectively, than those in the ground station. When the train came into the station, the LAeq of the TNIS at the island platform were higher than that at the side platform. The result suggests that the underground island platform, which is one platform at the center and the rail tracks on the both sides, needs improvement in particular although it is most frequently adopted as newly constructed stations in Japan.

10:20

5aNSc4. The quiet office—noise abatement in office buildings by means of smart structures. Thilo Bein (Fraunhofer LBF, Bartningstr. 47, 64289 Darmstadt, thilo.bein@lbf.fraunhofer.de), and Joachim Bös (TU Darmstadt, Chair of System Relibaility and Machine Acoustics, Magdalenenstr. 4, 64289 Darmstadt)

Noise is a serious form of environmental pollution believed to affect the lives of some 100 million European citizens. The cost of the associated damage is estimated at more than ten billion euro per year. Noise leads to serious health problems, limits the capability to learn and effects the occupant comfort and performance in buildings. In this context, advanced noise abatement concepts are being developed in the demonstration project "Quiet Office" within the LOEWE-Center AdRIA (Adaptronics – Research, Innovation, Application), a large interdisciplinary research project funded by the German federal state of Hessia. As underlying principle for noise reduction concepts, Active Structural Acoustic Control (ASAC) is primarily being considered. Applying ASAC concepts, the noise radiation is controlled either by controlling the structural vibration of the radiating structure or by controlling the structure borne sound path. This paper will present the most recent concepts and results from the "Quiet Office". Starting from more generic concepts, concepts specifically designed for office applications including smart windows will be discussed. Examples presented will include distributed absorbers, shunt technologies with piezoelectric ceramics, smart Helmholtz resonators and fully active noise and vibration control.

10:40–11:00 Break

11:00

5aNSc5. Noise reduction by eaves/louvers attached on façade of high-rise buildings. Shinichi Sakamoto (Institute of Industrial Science, The University of Tokyo, sakamo@iis.u-tokyo.ac.jp), and Takumi Asakura

The authors have been investigating noise reduction effects of several types of eaves/louvers attached on façade of high-rise buildings. The eaves are originally used for the aim of shading of solar radiation or as structures for fire-proofing in Japan, but the devices can also provide high effectiveness of noise reduction against road traffic. The authors have revealed such effects through wave-based numerical analyses, scale model experiments and in-situ experiment. The eaves attached on the façade work as either noise shielding materials or noise reflectors, so inclined eaves at higher rooms have more effectiveness of noise reduction. The effectiveness varies with protrusion length of the eave, inclination angle, height of the room and so on. In this paper, such results of our investigation on noise reduction by eaves and louvers will be presented.

11:20

5aNSc6. On site performances of ventilation windows. H.K. Wong, C.N. Yung (Housing Department, Hong Kong SAR Government, hungkeung.wong@housingauthority.gov.hk), and S.K. Tang (Department of Building Services Engineering, The Hong Kong Polytechnic University)

The present study investigates the performance, in term of sound attenuation, of ventilation windows when applied to a real situation. A full scale model consisted of two identical modular public housing residential flats was built up next to a very busy truck road with traffic noise level LA10 exceeding 80 dBA in general. One of these model flats was equipped with two ventilation windows, while the other with conventional windows having side-hung window panes. Noise measurements were carried out simultaneously inside both flats and at the façades of these model flats. The total conventional window opening size was kept at that of minimum prescribed area for ventilation (1/16 of floor area in Hong Kong). The measurements at the façades outside provided the necessary adjustments if necessary whereas the sound attenuation is the difference between the insertion loss of the ventilation window and the conventional window. The present results show that the current setup of the ventilation window can provide sound attenuation of 6.6 dB without any sound absorption treatment inside the window. The sound attenuation rose up to 8.1dBA when the two vertical interior sides and the interior top surface of each ventilation window were lined with such materials.

11:40

5aNSc7. The acoustic and ventilation performance of new ventilated window design. Michelle Wang (Dept. of CSE, H.K. Polytechnic University, Humghom, Hong Kong, michelle.wong@connect.polyu.edu.hk), Catherine Hui, and C.F. Ng

Noise and air pollution problems become significantly in dense city such as Hong Kong, since buildings are usual located close to the heavy traffic lines. Traditional openable window cannot fulfill all the functions of noise reduction, lighting and natural ventilation. A new ventilated window, which combines the multiple quarter-wave resonators (silencer) with the new wing wall design, is designed to balance between acoustic and ventilation performances. Furthermore, the use of multiple-wave resonators to replace absorption material can enhance the durability; avoid small particle emission and toxic gas due to fire. The acoustic and ventilation performance of new ventilated window were examined in this study. Noise attenuation of the new ventilated window design is improved significantly by combining flexible absorber and quarter-wave resonator effects. The test methods in acoustics are in accordance with relevant ISO 140 procedures.Transmission loss of 10 dB to 22 dB can be achieved in the frequency range of 500 Hz to 4k Hz band. The best ventilation performance of the wing wall is at an incident angle of 45°. Outlet air flow velocity of ventilated window design is double of the velocity of "open window". Thus, both the acoustics and ventilation performance of the new ventilated window is effective. Wind-driven natural ventilation is an effective way in maintaining the comfort and health of indoor environment.

Contributed Papers

12:00

5aNSc8. Influence of music performance on Sun Yat-sen Memorial Hall. Linqiang Gu and Weixin Lin (Department of Architecture and Urban Planning, Guangzhou University, 510006, China, gulinqiang@gmail.com)

Currently, many heritage buildings have been converted into music performance venues. Performing at ultra high sound pressure has induced coupled vibration of building elements which should be of protected, then caused irreversible damage of them after a certain time, such that application requirements and cultural relic protection formed an inevitable contradiction. Research on a case study of Guangzhou's Sun Yat - sen Memorial Hall in China demonstrated that music performance has little impact on the structure safety but a passive influence on some heritage building elements like colored glass and some murals, gave a reference for selection and design of damping measures, and at the same time pointed out that some issues required further study in this area.

12:20

5aNSc9. Acoustic design criteria in green building rating systems. May Han Grace Kwok (Allied Environmental Consultants Ltd. 19/F, Kwan Chart Tower, 6 Tonnochy Road, Wanchai, Hong Kong, gk@aechk.com)

Acoustic design aspect has been considered in various green building rating systems as part of the sustainable building design approach in an attempt to ensure a desirable acoustic environment for the future occupants. Primarily, the focus is on speech privacy, background noise control and noise isolation in a general building setting. In this paper, the acoustic design criteria adopted by different green building rating systems, including LEED in the United States, China Green Building Label in China, BREEAM in Britain, Green Star in Australia, CASBEE in Japan, BEAM Plus in Hong Kong, are evaluated in a comparative study with pros and cons highlighted. Selected criteria are investigated against respective regulatory requirements and common acoustic design standards and practices. The potential implications on building design and construction as well as the associated benefits to building users are also discussed. Possible conflicts and synergy with other sustainable design aspects in the green building rating systems are addressed. Lastly, acoustic design strategies and post-occupancy acoustic surveys for green buildings are recommended.

FRIDAY MORNING, 18 MAY 2012 S224 + S225, 9:20 A.M. TO 12:40 P.M.

Session 5aPA

Physical Acoustics: Sound Generation, Propagation, and Scattering (Lecture/Poster Session)

Constantin Coussios, Cochair
constantin.coussios@eng.ox.ac.uk

Contributed Papers

9:20

5aPA1. Formulation and selected applications of a regularized elastodynamics integral equation approach to large scale scattering problems involving inhomogeneous objects. Elizabeth Bleszynski, Marek Bleszynski, and Thomas Jaroszewicz (Monopole Research, Thousand Oaks, CA 91360, elizabeth@monopoleresearch.com)

Elements are presented of the formulation and selected applications of the combined surface and volume integral-equation approach for finding pressure, displacement, and traction fields in composite objects consisting of piece-wise homogeneous regions characterized by different Lame's material parameters, as well as of highly inhomogeneous regions. The proposed elastodynamics surface/volume-based integral equations, in which the unknowns are volumetric pressure or displacement variables and surface displacement and traction fields on the interfaces, involve only weakly singular integral-equation kernels with at most 1/r singularity. The proposed method is found to provide well-convergent and accurate solutions for a large spectrum of problems, in particular those involving high-contrast material interfaces. The proposed method's accuracy and its capability (achieved through the (non-lossy FFT-based) Adaptive Integral Method matrix compression) of handling large-scale problems involving several millions of unknowns, is demonstrated through comparison with analytically constructed solution for a layered material sphere and in selected numerical simulations of propagation of acousto-elastic waves in human head and inner-ear region through air- and bone-conduction pathways. The proposed approach can be used in assessing effectiveness of noise-protection devices. This work is supported by a grant from the Air Force Office of Scientific Research.

9:40

5aPA2. Diffuse ultrasonic backscatter in two-phase media. Dalie Liu (Dept. of Mechanical Engineering, Zhejiang A&F University, 88 Huancheng North Road, Lin'an, Hangzhou, Zhejiang Province, 311300, China, dalieliu@msn.com), and Joseph Turner (Dept. of Mechanical & Materials Engineering, W 317.4 Nebraska Hall, Univ. of Nebraska-Lincoln, Lincoln, NE 68588-0526)

The microstructure of sintered metals and ceramics may be modeled as two phases consisting of the particles and a surrounding matrix during certain processing stages. Diffuse ultrasonic backscatter has been widely used to extract the microstructural parameters and also to detect flaws in such materials. In this research, a singly-scattered response model is used to predict the grain noise for the two-phase model. The expression of backscatter coefficient is derived and found to be dependent on the spatial correlations of the material properties. The geometric two-point correlation function plays an important role in this model and can be determined from numerical correlation statistics. Good agreement is found between the results determined from the model and the experimental results.

10:00

5aPA3. Mass flow and surface wave in a vertically vibrated granular system. Hui Cai, Weizhong Chen, and Guoqing Miao (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, mg0723001@smail.nju.edu.cn)

We report an experimental finding on coexistence of the horizontal mass flow and the vertical surface wave in a vertically vibrated granular system. The container is an annular channel with a sawtooth-shaped base which is excited and controlled by a vibration system. The particles are copper spheres with diameter of 3 mm. The driving frequency f and dimensionless acceleration amplitude $\Gamma=4\pi^2 f^2 A/g$ (A is the driving amplitude and g the gravitational acceleration) are used as two control parameters. A high-speed image recording system is used to record the movements of all spheres, and an image processing technique is used to track the locations of all particles. As Γ increases beyond a critical value (about 1.8), the granular layer fluidizes from top to lower parts, and the horizontal mass flow appears. The velocity of the mass flow increases with Γ, and from bottom to top decreases monotonously. In the meanwhile, if Γ exceeds 2.5, the vertical surface wave occurs. The amplitude of the surface wave increases with driving amplitude. The mechanism of the mass flow is the ratchet effect of the sawtooth, while the correlation between the mass flow and the surface wave is to be explored.

10:20

5aPA4. Acoustic-gravity waves in ocean and atmosphere generated by an underwater source. Iosif Fuks (NOAA Earth System Research Laboratory and Zel Technologies LLC, Mail Code R/PSD-99, 325 Broadway, Boulder, CO 80305, Iosif.Fuks@noaa.gov), and Oleg A. Godin (NOAA Earth System Research Laboratory, Physical Sciences Division and CIRES, University of Colorado at Boulder, Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305)

Air-water interface becomes anomalously transparent, and the power flux in the wave transmitted into the air increases dramatically, when a compact sound source in water approaches the interface within a fraction of wavelength [O.A. Godin, Phys. Rev. Lett. **97**, 164301 (2006)]. The anomalous transparency of the ocean-atmosphere interface has important implications for detection of underwater explosions and monitoring of compliance with the Comprehensive Nuclear Test Ban Treaty. At wave frequencies below 0.1 Hz, it becomes necessary to take gravity into account. Then fluid buoyancy and compressibility simultaneously serve as restoring forces, and mechanical perturbations in the water and in the air propagate as acoustic-gravity waves (AGW). It was previously shown [I. Fuks and O.A. Godin, Proc. OCEANS'11, MTS/IEEE, Kona, HI, Sept. 2011] that, in the case of a shallow source in an ocean of an infinite depth, a sharp peak in the power flux into air appears at frequencies close to a cutoff frequency of about 4mHz of a surface acoustic-gravity wave. In this paper, we extend these results to the ocean of a finite depth where the AGWs interact with an elastic bottom.

10:40–11:00 Break

11:00

5aPA5. Micorcrack localization in pipelines using nonlinear guided waves combined with time reversal. Xiasheng Guo, Di Yang, and Dong Zhang (Key Laboratory of Modern Acoustics (MOE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, guoxs@nju.edu.cn)

A three-dimensional (3D) micro-crack imaging technique is proposed for pipeline inspections. With non-classical nonlinear guided waves generated from micro-cracks recorded, the third harmonic waves are used to image the fatigued crack, with the help of time reversal process. A finite-difference time-domain (FDTD) code is programmed to solve the wave equations under cylindrical coordinates, and simulate the wave propagation process. A defect with hysteretic stress-strain behavior is embedded in the model; its interaction with guided waves generates nonlinear components, the harmonic components are time reversed and played back into the pipe, hence produce the image of the cracked area. The results show excellent spatial retrofocusing capability. The accuracy of localization depends on crack orientation angle and an adopted guided wave mode. [This work is supported by the National Natural Science Foundation of China (11104140) and Natural Science Foundation of Jiangsu Province (BK2011543)]

11:20

5aPA6. Characteristics of thickness-shear modes excited by two layers of piezoelectric films in acoustic sensors. Hui Zhang, Shu-yi Zhang, Li Fan, and Yu-ran Wang (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, paslabw@nju.edu.cn)

The thickness-shear mode (TSM) excited by two layers of (112(–)0) textured hexagonal piezoelectric films are studied to improve the performances of acoustic sensors. The corresponding researches are carried out by the acoustic wave propagation model which considers the effects of the acoustic attenuations. When the acoustic field and electric field satisfy to a special match condition, i.e. the phase variation of TSM at each film equal to π, both piezoelectric layers with opposite polarization directions reduce the first TSM and generate the second TSM with higher frequency and higher quality factor than the first TSM excited by two-layers of piezoelectric films with identical polarization direction. Furthermore, the second enhanced TSM holds a moderate the ratio of the operating frequency to the quality factor. These properties of the TSM are appropriate for improving the mass sensitivity and sensitive resolution of the acoustic sensors in liquid medium. Acknowledgment: This work is supported by National Natural Science Foundation of China (No. 11004099 and 11174142), State Key Laboratory of Acoustics of Chinese Academy of Sciences, and also PAPD of Jiangsu Higher Education Institutions.

11:40

5aPA7. Theoretical study of interaction of the acoustic waves in different crystals. Xiaozhou Liu, Tingting Zhang, and Xiufen Gong (Key Lab of Modern Acoustics, Institute of Acoustics, Nanjing University, Nanjing 210093, China, xzliu@nju.edu.cn)

In isotropic solids, transverse acoustic waves propagate undistorted and the interaction of collinear transverse waves is forbidden. In this study, It is found that transverse and longitude waves behave differently in anisotropic solids and the different formulas are given for the interaction of transverse and longitude acoustic waves propagating in different kinds of crystals including cubic, trigonal, tetragonal, hexagonal crystals. The formulas are expressed in terms of the derivatives of the strain energy. The formulas are used to calculate the interaction coefficients in the general anisotropic medium emphasizing the influence of nonlinearities. The evolution equations are derived for the wave amplitudes and find analytical formulas for all interaction coefficients of quadratically nonlinear interacting waves. The equations of motion show that transverse and longitude waves are distort as they propagate and interact. It is observed that two acoustic waves with frequency can produce harmonic waves with sum and difference frequency. This shows evidence that the intrinsic nonlinearities of an anisotropic solid can induce an interaction of acoustic waves. This project is supported the National Basic Research Program of China (Grant Nos. 2012CB921504, 2011CB707902), financial support of the National Natural Science Foundation of China (Grant No. 11074122), fundamental Research Funds for the Central Universities (Grant Nos. 1113020403, 1101020402), State Key Laboratory of Acoustics, Chinese Academy of China (Grant No. SKLOA201005) and A project funded by the priority academic program development of Jiangsu higher education institutions.

The following abstracts will be presented in poster format. The posters will be on display and the authors will be at their posters from 12:00 noon to 12:40 p.m.

5aPA8. Optimization and limitations of a preconditioned multi-level fast multipole algorithm for acoustical calculations. Ralf Burgschweiger, Martin Ochmann (Beuth Hochschule für Technik Berlin, University of Applied Sciences, Luxemburger Str. 10, 13353 Berlin, Germany, burgi@beuth-hochschule.de), Ingo Schäfer, and Bodo Nolte (Federal Armed Forces Underwater Acoustics and Marine Geophysics Research Institute, Klausdorfer Weg 2-24, 24148 Kiel, Germany)

The Multi-Level Fast Multipole Method (MLFMM) allows the computation of acoustical problems based on the Boundary Element Method (BEM) where the discretized models of the corresponding structures may consist of a huge number of elements. The required calculation time and the memory requirements are much less when compared with conventional methods because the algorithm uses a level-based composition of the potentials from different point sources to acoustic multipoles, which highly accelerates the computation of the matrix-vector-products required for iterative solvers. A multi-level single-order variation of the algorithm was extended to a multi-level adaptive-order version, which was analyzed and optimized with respect to quality, performance and parallelization issues. The iterative solvers used with the MLFMM will be combined with appropriate preconditioners for reducing the number of iterations and improving the performance. The insights gained will be presented using different test cases and the results achieved will be compared with analytical solutions and results of conventional BEM- and FEM-based calculations.

5aPA9. Reduced acoustic cloaks: theoretical analysis and numerical simulation. Yuxian Fan (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, fanyx@mail.ioa.ac.cn), Wenlin Hu, Peifeng Ji, and Jun Yang

Acoustic cloaking is a new technique which can make an object invisible in acoustic waves. This method of controlling and directing sound has a promising prospect in application. Rigorous physical parameters are required for perfect cloaking, which is unpractical for experimental realization. Reduction is an alternative when imperfect cloaking performance is acceptable. In this paper, theoretical analysis of reduced acoustic cloaks based on transformation acoustics is presented. Fundamental features of acoustic cloaking are derived and discussed. Layered medium theory is employed to design reduced acoustic cloaks. Reasonable physical parameters and tolerable cloaking performance are expected in the prerequisite of the work. The properties of the designed reduced acoustic cloaks are studied. Numerical simulation shows that the reduced acoustic cloaks have effective cloaking performance.

5aPA10. Research on experiments for high frequency sound propagation with rough surface in shallow water. Xudong An (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, anxudong@mail.ioa.ac.cn)

A experiment was carried out to research high frequency underwater sound propagation with rough surface in shallow water. A wave-maker was used to simulate the sine wave surface with different wavelength, amplitude and cycle in a pool (108m$\times$7m$\times$3.5m). The multiple propagation tracks and the multiple reflections at surface and bottom as well as focusing and defocusing due to reflection from surface waves were measured and analyzed. The accuracy and efficient of experiment are verified by the agreement between the numerical simulation and the analysis results of the measured experimental data.

5aPA11. Acoustical emission of jet-edge systems using a vibrating element in water. C Li, Jingjun Deng, and Lixin Bai (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing 100190, China, lichao@mail.ioa.ac.cn)

The jet-edge system using a vibrating element (liquid whistle) was investigated experimental using high-speed camera. The vibrating element vibrated periodically with a few millimeters displacement and cavitaion bubbles were observed near the tip of the vibrating element. The acoustic emission of the system was measured with hydrophones in some place. The low frequency and shock signals were observed in the pressure signal, and whistle can be heard at the same time. In frequency field, the line and broadband spectrum exist. The foundation frequency f0 of the vibrating element and harmonics compose the line spectrum. The former 10 order harmonics can be seen obviously. As the flow velocity increasing, the frequency f0 and the harmonic do not change, but the intensity of the high order harmonics and broadband spectrum become strong. Base the experimental result, we consider that the f0 is the foundation frequency of the vibrating element, and the harmonics and the continuous spectrum is the bubble cavitation emission. Acknowledgement: This work was supported by the National Natural Sciences Foundation of China (10804118)and he National Science and Technology Major Project of China (No. 2011ZX05032-003)

Session 5aPP

Psychological and Physiological Acoustics: Auditory Function, Mechanisms, and Models (Poster Session)

Michael Heinz, Cochair
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Contributed Papers

All posters will be on display from 9:20 a.m. to 12:40 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:20 a.m. to 10:40 a.m. and contributors of even-numbered papers will be at their posters from 10:40 a.m. to 12:40 p.m.

5aPP1. Electrophysiologic assessment of (central) auditory processing disorder in children with non-syndromic cleft lip and/or palate. Xiaoran Ma (B092, 5F, Division of Speech and Hearing Sciences, 34 Hospital Road, The Prince Philip Dental Hospital, Sai Ying Pun, Hong Kong, xiaoran@hku.hk), Lian Ma (School of Stomatology, Beijing University), and Bradley McPherson (5F, Division of Speech and Hearing Sciences, 34 Hospital Road, The Prince Philip Dental Hospital, Sai Ying Pun, Hong Kong)

Cleft of the lip and/or palate is a common congenital craniofacial malformation worldwide, particularly non-syndromic cleft lip and/or palate (NSCL/P). Though middle ear deficits in this population have been universally noted in numerous studies, other auditory problems including inner ear deficits or cortical dysfunction are rarely reported. A higher prevalence of educational problems has been noted in children with NSCL/P compared to craniofacially normal children. These high level cognitive difficulties cannot be entirely attributed to peripheral hearing loss. Recently it has been suggested that children with NSCLP may be more prone to abnormalities in the auditory cortex. The aim of the present study was to investigate whether school age children with (NSCL/P) have a higher prevalence of indications of (central) auditory processing disorder [(C)APD] compared to normal age matched controls when assessed using auditory event-related potential (ERP) techniques. School children (6 to 15 years) with NSCL/P and normal controls with matched age and gender were recruited. Auditory ERP recordings included auditory brainstem response and late event-related potentials, including the P1-N1-P2 complex and P300 waveforms. Initial findings from the present study are presented and their implications for further research in this area — and clinical intervention — are outlined.

5aPP2. Change deafness with recognizable and unrecognizable sounds. Vanessa Irsik (UNLV- 6151 Mountain Vista #527 Henderson, Nv 89014, irsikv@unlv.nevada.edu), Melissa Gregg, and Joel Snyder (UNLV 4505 S. Maryland Parkway, P.O. Box 455030, Las Vegas, Nevada 89154-5030)

Change Deafness with Recognizable and Unrecognizable Sounds Change deafness is the remarkably frequent inability of listeners to detect changes occurring in their auditory environment. In this study, we used behavioral measures and event-related potentials (ERPs) to determine if change deafness is a fundamental auditory sensory process, rather than simply a reflection of verbal or semantic memory limitations. Change detection performance was examined for scenes composed of four recognizable or unrecognizable sounds. Listeners completed a change detection task by making a same/different judgment for two consecutively presented scenes that were either the same (Same trials) or had one sound replaced by another sound (Change trials). The behavioral data indicated substantial change deafness for both recognizable and unrecognizable sounds, indicating that change deafness is not the result of verbal or semantic memory limitations. In Change trials, N1 ERPs were less smaller in the post-change scene on trials in which the change was not detected, suggesting that change deafness is associated with less robust sensory encoding. P3 ERPs to the post-change scene were also smaller for non-detected changes, which may reflect lack of memory updating, attention, and/or awareness during change deafness. Overall, the results provide novel information regarding the stages of processing involved in change deafness in natural auditory scenes.

5aPP3. Attentional modulation of EEG signals. Inyong Choi (Center for Computational Neuroscience and Neural Technology, Boston University, 677 Beacon St., Boston, MA 02215, iychoi@bu.edu), Hari Bharadwaj (Department of Biomedical Engineering, Boston University), and Barbara Shinn-Cunningham (Center for Computational Neuroscience and Neural Technology, Boston University, 677 Beacon St., Boston, MA 02215)

Physiological measures revealing effects of selective attention are needed to develop insights into how auditory attention is controlled. Here, we used electroencephalography (EEG) to measure neural activity when listeners attended one of two competing auditory streams. Listeners identified whether a target three-tone melody had a rising, falling, or non-monotonic pitch contour while ignoring a competing, simultaneous melody. To enhance stream segregation, the streams differed in timbres (oboe and cello) and perceived lateral position (to the left, to the right). We “tagged” each of the streams by adding amplitude modulation (AM) at different gamma-band frequencies. We found that the evoked gamma activity at the AM frequencies was statistically above the noise floor, and depended on the spatial configuration and the modulation frequency, as might be expected. Importantly, for identical acoustic inputs, the relative strength of the AM frequencies depended on which stream a listener attended. Moreover, the pattern of responses depended on what cue directed the listener to attend to a particular target. Results suggest that the control of attention engages different neural networks, depending on the feature directing attention. These results will be discussed in light of efforts to develop brain-machine interfaces that monitor auditory attention.

5aPP4. What is so hard about selectively attending? Ross K Maddox, Willy Cheung, and Adrian KC Lee (University of Washington, Portage Bay Building, P.O. Box 357988, 1715 Columbia Road North, Seattle, WA 98195, rkmaddox@uw.edu)

As the number of elements that make up an auditory scene increases, it becomes harder to selectively attend just one of those elements. Previously, the limit of listeners’ abilities to attend a target stream of repeating letters in an overcrowded scene was tested. In that experiment, each stream consisted of a repeated monotonized and localized spoken letter (an “item”), with a

repetition period of 1 s. Among streams, item onset times were distributed across each repetition. Listeners were asked to detect when the attended target letter changed to an oddball "R" for a single repetition, ignoring such occurrences in the non-target streams. With a constant repetition period, adding streams to the stimulus meant that the number of items per second increased proportionally. The decrease in performance could thus be a result of having more streams in the scene, or because of the increased item rate. Here, a similar experiment was performed, holding the item rate constant, rather than the repetition period. The results allow us to disentangle the effects of the number of distractor streams and the item rate, yielding insight into the specific reasons for the diminished ability to selectively attend. Funded by USA-NIH-T32DC009975 (RKM) and R00DC010196 (AKCL).

5aPP5. Effects of spatial attention on across-frequency grouping in speech. Miguel Cepeda, Virginia Best, and Barbara Shinn-Cunningham (Boston University 677 Beacon Street Boston, MA 02215, mdcepeda@bu.edu)

To understand speech, listeners integrate information across disparate frequency regions. Such integration may depend not only on low-level grouping cues (e.g., correlations in envelope across frequency) but also on volitional attention (e.g., integrating only frequency bands from a desired direction). Here, listeners reported the content of a target sentence conveyed by two spectrally separated, narrow bands of speech presented to the left ear. Two additional speech bands, centered between the other bands, were presented either to the left or the right ear. These additional bands were either from the target speech (matched) or an independent sentence (conflicting). In the key experimental block, listeners were instructed to attend to the speech on the left while either conflicting bands or, infrequently, matched bands were presented on the right. Splitting the target across the ears degraded intelligibility, showing that spatial separation interferes with grouping. However, directed spatial attention had no effect: intelligibility was equivalent when listeners explicitly were told to attend to the target bands split across both ears and equivalent trials in which listeners volitionally attended only the left ear. Results show that for structure-rich speech, directed attention does not overcome automatic grouping processes: across-frequency grouping of related speech bands is obligatory.

5aPP6. Functional subnetwork structure in auditory cortex for stream segregation. Takahiro Noda, Ryohei Kanzaki, and Hirokazu Takahashi (University of Tokyo, 153-8904, noda@brain.imi.i.u-tokyo.ac.jp)

Perceptual integration and segregation of alternating tone sequence differing in frequency (ABA-ABA-...) depend on the frequency differences (Delta-Fs) between A and B tones and the inter-tone intervals (ITIs) between successive tones. In the auditory cortex, tonotopic separation, forward suppression and multisecond habituation have been considered as possible neural correlates of this perceptual phenomenon. This model, however, cannot completely account for the van Noorden's perceptual boundary and the temporally continuous perception of auditory streaming. Here we examined the temporal changes of the functional network properties in auditory cortex to tone sequences with different Delta-Fs and ITIs. Specifically, we recorded local field potentials using microelectrode arrays from anesthetized and awake rats and constructed the functional network based on phase synchrony in gamma-band oscillation. As the results, the networks consisted of sub-networks highly correlated with tonotopy, and the sub-network selective to B tones lasted for a prolonged period at large Delta-F. Such characteristic substructures of functional network are a possible candidate of neural mechanisms of auditory stream segregation.

5aPP7. Probing the cortical dynamics involved in ignoring a low-salience distracting sound. Eric Larson and Adrian KC Lee (University of Washington, Portage Bay Building, Room 204, 1715 Columbia Road North, Seattle, WA 98195, larsoner@uw.edu)

Verbal communication in many everyday settings requires listeners to ignore distracting auditory events. However, how the brain coordinates the suppression of even low-salience auditory distractors remains unclear. In studies of visual attention, it has been observed that the frontal eye fields (FEF) are involved in orienting and maintaining attention to targets of interest. With multiple fMRI studies implicating FEF in orienting auditory attention, we sought to examine the temporal dynamics of auditory attentional control in the presence of a low-salience distractor. We utilized a task that required listeners to attend to and report one of two simultaneous, but spatially-separated, spoken digits. A visual arrow-cue prompted listeners to report the stimulus originated from that hemifield. Shortly following the visual cue, we presented listeners a noise burst that either originated from the same hemifield or the opposite side as the visual cue. We utilized simultaneous magneto- and electro-encephalography to investigate the cortical dynamics involved in suppressing the potentially distracting noise burst. Using source-space analysis, we observed significantly stronger activation in left FEF when the noise burst was presented in the to-be-attended hemifield, providing further support that left FEF is specifically involved in maintaining auditory spatial attention. Funded by USA-NIH T32DC000018(EL) and R00DC010196(AKCL).

5aPP8. Effects of changes in the depth feeling of the visual target on the simultaneity perception of an auditory-visual event. Hiroshi Hasegawa, Tomoharu Ishikawa, Masao Kasuga, and Miyoshi Ayama (Utsunomiya Univ., hasegawa@is.utsunomiya-u.ac.jp)

In this study, we investigated the simultaneity perception between a visual stimulus and its associated sound. We carried out experiments of an auditory-visual stimulus presentation using an audio-video clip of a man beating a drum on a road. The visual stimulus had a feeling of depth with a perspective view of the road. There were four kinds of distance between the visual target of a man beating a drum and the video camera to capture the target of 5, 10, 20, and 40m, and we called these distances as "the presentation distances." We presented the auditory-visual stimuli to experimental subjects at each presentation distance of 5, 10, 20, and40 m under various conditions, where we varied the feeling of depth of the visual stimulus from -40% to 40% and the time delay between the auditory and visual stimulus from -8F to 8F (1F = 1/30s). We analyzed the experimental results and calculated the point of subjective simultaneity (PSS) of the auditory-visual event. As a result, the PSS intended to increase as the feeling of depth increased of the visual stimulus. This result shows that changes in the depth feeling of the visual stimulus could influence the simultaneity perception.

5aPP9. Robustness of audio-visual spatial disparity in peripheral field. Ryota Miyauchi (Japan Advanced Institute of Science and Technology, 1-1 Asahidai, Nomi, Ishikawa, 923-1292, Japan, ryota@jaist.ac.jp), Dea-Gee Kang, Yukio Iwaya, and Yôiti Suzuki (Research Institute of Electrical Communication, Tohoku University, 2-1-1 Katahira, Aoba-ku, Sendai 980-8577, Japan)

In our previous studies, a perceptual phenomenon, audio-visual peripheral spatial disparity (AVPSD), has been found. The spatial coincidence of perceived locations of a sound and flash presented in the peripheral field varies from their coincident physical location. Since the stimulus condition used in the previous experiments was limited, three new experiments were conducted to investigate the robustness of the spatial disparity between the perceptual and physical locations of the sound and flash. In Experiments 1 and 2, the frequency characteristics of the sound and the repetition of stimulus appearance were respectively changed from those used in the previous experiments. In Experiment 3, the presentation timing of the sound or the flash was delayed for one second. The results of Experiments 1 and 2 show that the spatial disparity was robustly appeared. In contrast, the spatial disparity disappeared in Experiment 3. These results suggest that the features of the stimulus pattern do not affect the spatial disparity, but the simultaneity of the sound and flash plays an essential role. Acknowledgment: This research was supported by a Grant-in-Aid for Specially Promoted Research (No. 19001004) and Grant-in-Aid for Young Scientists (B) (No. 21730584).

5aPP10. Visual deprivation improves auditory scene analysis. Marie-Soleil Houde, Marianne Bélanger, Douglas Shiller, and François Champoux (Université de Montréal, C.P. 6128, Succursale Centre-Ville, mariesoleilhoude@alumni.uottawa.ca)

The present study aims to examine the effect of temporary visual deprivation on a process central to auditory segregation, namely harmonicity. A harmonicity discrimination task was administered twice, with an interval of 90 minutes, in two groups of individuals. One group was temporary deprived of visual information during that interval period. Control conditions revealed that auditory capabilities were similar across groups. However, results in the

subsequent testing session revealed that temporarily deprived individuals showed a remarkable improvement in the auditory task in comparison with the non-visually-deprived group (i.e. control group). These results suggest that temporary blindness can improve auditory segregative processing.

5aPP11. Acoustical dummy head for Chinese adults. Na QI and Zihou MENG (Communication University of China, qina@cuc.edu.cn)

An acoustical dummy head is designed based on the Chinese national code of adult head geometry. From a survey of more than four hundreds Chinese adult auricle structure, a typical physiological structure of auricle is selected for the acoustic dummy head. Eyes, nose and other head structure are simplified. The HRTF of the acoustical dummy head is measured. The cues relating to sound source localization, including interaural time difference (ITD),interaural level difference (ILD), and spectral features introduced by pinna, nose, hair and other detail structure of head, are analyzed. And then the analytical data is compared with a full-scale dummy head. The result shows that the acoustical dummy head retains the main acoustic characteristics attributed to auditory localization. A listening test is carried out to verify this property.

5aPP12. Effects of spatial localization on sentence recognition in noise. Tatsuo Nakagawa (Yokohama National University, Faculty of Education and Human Sciences, 79-2 Tokiwadai, Hodogaya-ku, Yokohama, Japan, nakagawa@edhs.ynu.ac.jp)

Speech recognition in noise improves when speech and noise are separated in space. Interaural level and time difference cues are used in spatial localization. Most people with hearing loss have better hearing in the low frequencies. Interaural cues for sound localization are also more important in the low frequencies. An objective of the current experiment was to assess the effects of spatial separation of speech and noise on the recognition of low-pass filtered speech in persons with normal hearing and hearing loss. Speech levels corresponding to 50% correct recognition of sentences were measured in a 65dB SPL multi talker noise. Spatial benefits for listeners with and without hearing loss are discussed.

5aPP13. Effect of similarity between target speech and time-reversed masker on speech intelligibility. Bin Jiang and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, binjiang@mail.ioa.ac.cn)

The similarity between target speech and interfering signal may affect masking. Since time-reversed masker is an unintelligible speech-like signal, the influence of intelligibility can be avoided. This paper describes a sentence intelligibility test with target speech masked by different time-reversed maskers to evaluate the similarity effect. The time-reversed masker is processed by locally time-reversal segments of speech uttered by same-talker, same- and different-sex talker. The results show that the amount of masking is highly dependent on the similarity between target speech and time-reversed masker: time-reversed masker of target speech gives a rise in average speech reception threshold (SRT) of about 3 dB and 9 dB compared to that of another talker with same- and different-sex, respectively. Assuming the energetic masking effect for time-reversed maskers of different talkers to be similar, the amount of similarity with same-sex is larger than 6 dB and the amount of similarity with same-talker is larger than 9 dB.

5aPP14. Comparison of frequency lowering algorithms on Mandarin speech recognition. Yunyi Zhang (Institute of Acoustics, Chinese Academy of Sciences, zyy780@sina.com), Jie Cui, and Ling Xiao

This paper examined several frequency lowering algorithms for improving speech recognition in Chinese listeners with high-frequency sensorineural hearing loss, including both frequency compression and frequency transposition algorithms. The frequency lowering algorithms were evaluated by a subjective listening test. Monosyllabic words in Mandarin spoken by a female and a male talker were used as speech materials. Speech materials processed by frequency compression and frequency transposition were presented to normal-hearing listeners in the listening tests, as well as the control condition of unprocessed speech. More benefits in speech recognition were observed with speech spoken by female talker than by male talker. It was showed that frequency lowering made little difference for some words while words with dental consonant were most infected by frequency lowering. The results indicate that Mandarin recognition for high-frequency hearing loss may benefit from frequency lowering in some cases, while whether choosing frequency lowering or not and which algorithm to choose depends on individuals.

5aPP15. The effect of spectral asynchrony on speech recognition in demanding listening conditions. Magdalena Wojtczak, Sachin Rai, Jordan A. Beim, and Andrew J. Oxenham (University of Minnesota, N218 Elliott Hall, 75 East River Rd., Minneapolis, MN 55455, wojtc001@umn.edu)

Speech recognition is remarkably robust to overall spectral asynchronies as large as 160-180 ms when performance is measured in quiet. One reason for such robustness may be substantial redundancy in the speech stimulus. This study investigated the effect of spectral asynchrony on speech recognition in listening conditions yielding 80% or less correct recognition score for synchronized speech (baseline condition). IEEE sentences mixed with a speech-shaped noise or two-talker babble were divided into 1/3-octave bands. Successive bands were progressively delayed from low to high, or from high to low, center frequency. The overall delay between the lowest and highest-frequency channels was varied between 0 and 160 ms. The difficulty of the task in the synchronous condition was determined by different signal-to-noise ratios and by presenting either the full stimulus or a lowpass-filtered version. The results show that performance is less robust to spectral asynchrony when speech is presented with an interfering background, although it remains relatively unchanged for overall delays up to 40-ms, independent of the direction of the delay as a function of frequency. The results were analyzed using the speech transmission index. [Supported by NIH grant R01DC010374].

5aPP16. Auditory and cognitive factors in speech and environmental sound perception of cochlear implant listeners. Valeriy Shafiro (Rush University Medical Center, valeriy_shafiro@rush.edu), Stanley Sheft, Sejal Kuvadia, Brain Gygi, and Kim Ho

This study examined the role of acoustic pattern discrimination abilities and higher-order cognitive factors on the perception of spectrally degraded speech and environmental sounds. Over the course of four testing sessions and a week of environmental sound training, cochlear implant and normal-hearing listeners (tested with a 4-channel implant vocoder simulation), received a 160-item environmental sound test, and isolated word and sentence tests. Listeners were also tested with a battery of cognitive tests which included measures of inhibition, closure, working memory and executive function, and a battery of psychoacoustic tests of spectral and temporal pattern processing. A great deal of overlap in the processing abilities for speech and environmental sounds was found in both groups of listeners. Centrally, speech and environmental sounds were associated with working memory and executive function measures (i.e., digit-span-backward, letter-number sequence). Psychoacoustically, the two sound classes were associated with measures of spectral resolution and temporal fine-structure processing. These findings indicate that successful perception of acoustically complex meaningful patterns under conditions of spectral degradation involves joint contributions from specific peripheral factors and central processes, and suggest directions for improving listener performance through training [Support provided by NIH/NIDCD].

5aPP17. Word retrieval process uses phonemic cues in memory-impaired epilepsy analyzed by Rey's auditory verbal learning test. Keiko Asano (Juntendo University, School of Medicine, 1-11-2905 Kinko-cho, Kanagawa-ku, Yokohama-City, Japan, keiko_asano@sakura.juntendo.ac.jp), Hidehiro Okura (Juntendo University, School of Medicine), Hidenori Sugano, Mitsuko Nakano, Keiko Fusegi, and Hajime Arai (Juntendo University, School of Medicine)

This study investigated what kinds of phonemic strategies memory-impaired patients with diagnoses such as frontal lobe epilepsy compared to the strategies healthy people use in order to memorize and retrieve words in the aspects of Rey's Auditory Verbal Learning Test (AVLT). This Auditory Verbal Learning Test is widely used in the field of clinical and neuropsychological assessments to examine word learning and memory. This test has two stages: A list of 16 unrelated, concrete nouns is presented over five learning trials with immediate recall tested following each presentation. After a delay interval of 30 minutes with no further presentations of the list, delayed recall is assessed. The number of words correctly recalled is commonly adopted as quantitative information in clinical assessment. After the tests, participants are asked what kinds of strategies they used to memorize the words in terms of examining the process of decoding, recall and retrieval. Spontaneous memorization strategies used by memory-impaired patients are visual-oriented and episode-making, whereas the most healthy people use the method of grouping a few words as one chunk. Contrary to self-monitoring strategies, especially among the epilepsy patients, phonemic cues are unconsciously used to retrieve the words. The implication on the function of different brain areas activated between patients and healthy people will also be discussed.

5aPP18. Acoustic cues used by blind travelers. Helen Simon (Smith-Kettlewell Eye Research Inst, San Francisco, CA, helen@ski.org), Deborah Gilden (Smith-Kettlewell Eye Res Inst. San Francisco, CA), John Brabyn (Smith-Kettlewell Eye Res Inst, San Francisco, CA), Al Lotze (Smith-Kettlewell Eye Res Inst., San Francisco, CA), and Harry Levitt (Advanced Hearing Concepts, Bodega Bay, CA)

People with vision loss rely heavily on subtle environmental sound cues for safe and efficient travel (“Wayfinding”). Using our laboratory-developed instruments, the acoustic cues available to blind individuals, with and without hearing loss, during real-life pedestrian travel were recorded. Acoustic signals picked up by electret condenser microphones in the ear canals were fed to a wearable digital audio recorder. Head and body movements were monitored by accelerometers and gyroscopes mounted on the heads and torsos of subjects during typical travel situations such as walking along a corridor with an open doorway. A skilled wayfinder can detect the presence of an open doorway from the acoustic characteristics of the ambient sound field (the “acoustic signature”). The salient characteristics of the acoustic signature when passing an open door were found to be below 1500 Hz. These data confirm previous work regarding ambient room noise near a wall and an opening (Ashmead, 1999). Unlike previous work, the current study also measured the interaural characteristics of the acoustic signature. The results of this investigation will be used to develop the design requirements of a special-purpose hearing aid for people with both vision and hearing loss. (Funded by NIDRR and Smith-Kettlewell Eye Research Institute.)

5aPP19. Contribution of pitch-accent information to lexical decision in Japanese. Ikuyo Masuda-Katsuse (Kinki University, katsuse@fuk.kindai.ac.jp)

Word intelligibility, latency in shadowing, and brain activations when listening to spoken Japanese with incorrect pitch accents compared to normal, were investigated. Japanese is one of pitch-accent languages. In a pitch-accent language, “where” the pitch falls is important, whereas in Mandarin Chinese “how” pitch rises and falls is important. The word intelligibility scores were examined based on the adequacy rating of the accent types. Results reveal that under noisy conditions, the higher the adequacy of the accent type was rated, the higher the word intelligibility was. Under the clean condition, a significant negative correlation was shown between latency in shadowing and adequacy rating of accent types. In the event-related fMRI experiment, brain activities were investigated when listening to words with incorrect pitch accents (INCORRECT condition) and words with normal pitch accents (NORMAL condition). The contrast between INCORRECT and NORMAL revealed the significant activation in the bilateral inferior frontal, the bilateral precentral, and the left supplementary motor area, and right superior temporal gyrus. The result support the view that speech perception is a sensory-motor process, suggesting when a pitch accent of the stimulus was mismatch to the template, silent rehearsal might be repeated to make a lexical decision.

5aPP20. A cross-language study of breathing rhythm related to different linguistic structures in Chinese and Korean. Hanna Oh (Dept. of Chinese Language and Literature, Peking University, Beijing 100871, China, hanna.pku@gmail.com)

This paper investigates correlations between respiration patterns and different linguistic structures in Chinese and Korean.The study includes comparative analysis between the Korean and Chinese scripts of poetry and news with same meaning and different languages, and the subjects included 4 native speakers (2 males, 2 females) for each language. It can be made certain by measuring physiological and acoustic parameters by means of two respiratory sensors for chest and abdomen and electroglottalgraphy. Preliminary results show that due to the poetry has fixed frame, therefore regardless of language and sex, the respiratory rhythm of the poetry is quite similar in two languages. In contrast, for news scripts, the data show that breathing groups of the Chinese quite correspond to intonation units. In contrast,the Korean is more flexable in planning breathing rhythm,generally breathing resets appears not only in intonation phrase but in accentual phrase,and chest respiration is mostly used for the accentual phrase. It suggests that different linguistic structures operate powerfully upon preplanning and respiratory patterns. Furthermore, the results show the correlation of voice quality variation and respiratory rhythm in different languages.

5aPP21. Boundary tones in production and comprehension of in pro-drop language: the case of Korean. Hyunah Ahn (University of Hawaii at Manoa, Department of Second Language Studies, 1890 East-West RD, Honolulu, HI 96822, hyunah1112@gmail.com)

This paper investigates if boundary tones have either a gradient or a categorical indication of structural phrasing. The role of prosody in syntactic disambiguation has been well documented and a recent study showed Korean speakers use the presence and absence of boundary tones to disambiguate a null-argument sentence borne out of double nominative construction in Korean (Ahn, 2011). Two experiments show that boundary tones play an important role both in comprehension and in production in spoken Korean. In Experiment 1, speakers read a paragraph including contexts and the critical sentence given in a latin-square design. The critical sentences in this production experiment include both null-argument and non-null-argument sentences to see if the use of boundary tones is for addressees or for the speaker himself. In Experiment 2, participants judge the naturalness of null-argument sentences given in contexts and answer comprehension questions probing if they got the correct reading of the two ambiguous interpretations. The results show a clear interaction effect of prosody and structure on the accuracy to the comprehension questions. A gradient-categorical dichotomy of prosody in sentence processing will be discussed and suggestions for a further study will be made to investigate the relationship of boundary tones and comprehension.

5aPP22. Estimating listeners’ internal noise from intensity and spectral-shape discrimination tasks. Huanping Dai (University of Arizona, Tucson, AZ 85721, hdai@email.arizona.edu), Feng-Yi Chuang, and Andrew Lotto

Within the framework of signal-detection theory, listeners’ ability to process stimulus information is partly limited by their internal noise. Knowing the internal noise of listeners will allow the researcher to predict their performance in various detection or discrimination tasks. For the purpose of estimating internal noise, in this study we measure the ability of listeners to discriminate the intensities of pure tones (Exp. 1) and two-tone complexes (Exp. 2), and to discriminate the spectral shapes of two-tone complexes (Exp. 3). The results from five listeners were fitted with an internal-noise model. The analysis showed that a major portion of the internal noise is correlated across frequency channels. This outcome is consistent with the conclusions of previous studies that estimated internal noise using different methods. Under the dominance of channel-correlated internal noise over channel-specific internal noise, listeners’ ability to integrate information over frequency should be reduced, and their performance in detection and discrimination should be less affected by uncertainty for the signal parameters.

5aPP23. High-frequency complex pitch: a search for temporal cues and for a role of spectral indices. Sébastien Santurette, Torsten Dau (Centre for Applied Hearing Research, Technical University of Denmark, DTU Bygning 352, Oersteds Plads, 2800 Kongens Lyngby, Denmark, ses@elektro.dtu.dk), and Andrew J. Oxenham (Department of Psychology, University of Minnesota, 75 East River Parkway, Minneapolis, MN 55455)

Harmonics in a complex tone are typically considered unresolved when they interact with neighboring harmonics in the cochlea and cannot be heard out separately. Recent studies have suggested that the low pitch evoked by unresolved high-frequency harmonics may be coded via temporal fine-structure cues. However, these conclusions rely on the assumptions that combination tones were properly masked and that the ability of listeners to hear out individual partials provides an adequate measure of resolvability. Those assumptions were tested by measuring the audibility of combination tones and their effects on pitch matches, the effects of relative component phases and of dichotic presentation, and listeners' ability to hear out individual partials. The results confirmed that combination tones affected pitch, but pitch remained salient when they were masked. The lack of dependence of pitch on relative component phases or dichotic presentation provided no evidence in favor of temporal cues. Moreover, similar trends were observed between pitch salience and the listeners' ability to hear out individual partials. The results are consistent both with the use of place information and with a temporal code based on the combination of information across auditory channels. [Supported by the Danish Research Foundation and by NIH grant R01DC05216.]

5aPP24. Envelope pitch at different stimulation sites of cochlear implant. Qinglin Meng, Meng Yuan, Hongyu Mou, and Haihong Feng (Shanghai Acoustics Laboratory, Chinese Academy of Sciences, No. 456, Xiaomuqiao Road, Shanghai, China, mengqinglin08@gmail.com)

Pitch perception researches showed that temporal modulation plays an important role in pitch perception from unresolved harmonics. Fundamental frequency (F0) enhancement through temporal modulation of envelope was utilized in some cochlear implant (CI) signal processing strategies [e.g., Laneau, et.al., 2006] to improve the pitch perception of CI. In this study, Temporal Modulation Transfer Function (TMTF) and the relationship between Amplitude Modulation Rate (AMR) and pitch perception at different stimulation sites (apical, middle, and basal) were investigated. The stimuli are sinusoidal-amplitude–modulation pulse trains. Two psychophysical experiments were carried out: Two-Alternative-Forced Choice on modulated/un-modulated stimuli and AMR-Pitch Ranking. Experimental results showed that some of the CI subjects exhibited distinct performance at different stimulation sites, which implies that issues of stimulation sites and the subject's individual performance should be considered for F0 enhancement strategies on CI. More experiments are in progress and whole results will be shown on the conference.# This work is supported by National Natural Science Foundation of China (11104316), and Shanghai Natural Science Foundation (11ZR1446000).

5aPP25. Loudness of sounds with a subcritical bandwidth. Jan Hots (Department of Experimental Audiology, Otto-von-Guericke University, Leipziger Strasse 44, D-39120 Magdeburg, Germany, jan.hots@med.ovgu.de), Jan Rennies (Project Group Hearing, Speech and Audio Technology, Fraunhofer IDMT, Marie-Curie-Strasse 2, D-26129 Oldenburg, Germany), and Jesko Lars Verhey (Department of Experimental Audiology, Otto-von-Guericke University, Leipziger Strasse 44, D-39120 Magdeburg, Germany)

The predicted loudness for narrowband signals with bandwidths smaller than the bandwidth of the auditory filter, i.e., with a subcritical bandwidth, depends on the type of loudness model. While models analyzing the long-term spectrum (stationary loudness models) predict loudnesses for these subcritical signals which are independent of the bandwidth of the sound, models sensitive to the temporal structure of the signal (dynamic loudness models) predict higher loudnesses for narrowband noise signals than for tones due to the inherent level fluctuations of the noise. The present study measures the loudness of subcritical sounds for center frequencies of 750, 1500, and 3000 Hz. For all three center frequencies, the level of the narrowband noise had to be up to 5 dB higher than the level of the equally-loud tone at the center frequency. This experimental finding is at odds with both predictions, i.e., the loudness is neither the same for all signals nor is the level of the narrowband noise smaller than that of the equally-loud tone. The data indicate a special character of sounds with a distinct tonal character which is not accounted for in current loudness models.

5aPP26. Amplitude modulation detection by human listeners in reverberant sound fields: carrier bandwidth effects and binaural versus monaural comparison. Pavel Zahorik (Department of Psychological and Brain Sciences, University of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu), Duck O. Kim, Shigeyuki Kuwada (Department of Neuroscience, Univ. of Connecticut Health Center, Farmington, CT 06030-3405), Paul W. Anderson, Eugene Brandewie, Regina Collecchia, and Nirmal Kumar Srinivasan (Department of Psychological and Brain Sciences, University of Louisville, Louisville, KY 40292)

Previous work [Zahorik *et al*., POMA, **12**, 050005 (2011)] has reported that for a broadband noise carrier signal in a simulated reverberant sound field, human sensitivity to amplitude modulation (AM) is higher than would be predicted based on the broadband acoustical modulation transfer function (MTF) of the listening environment. Interpretation of this result was complicated by the fact that acoustical MTFs of rooms are often quite different for different carrier frequency regions, and listeners may have selectively responded to advantageous carrier frequency regions where the effective acoustic modulation loss due to the room was less than indicated by a broadband acoustic MTF analysis. Here, AM sensitivity testing and acoustic MTF analyses were expanded to include narrowband noise carriers (1-octave and 1/3-octave bands centered at 4 kHz), as well as monaural and binaural listening conditions. Narrowband results were found to be consistent with broadband results: In a reverberant sound field, human AM sensitivity is higher than indicated by the acoustical MTFs. The effect was greatest for modulation frequencies above 32 Hz and was present whether the stimulation was monaural or binaural. These results are suggestive of mechanisms that functionally enhance modulation in reverberant listening. [Work supported by the NIH/NIDCD.]

5aPP27. Behavioral and electrophysiological measures of stimulus envelope and fine structure contributions to the binaural masking level difference. Ann Clock Eddins, Makenzie Kline, and David A. Eddins (Dept of Communication Sciences & Disorders, University of South Florida, 4202 E Fowler Ave., PCD 1017, Tampa, FL 33620, aeddins@usf.edu)

The goal of this study is to establish electrophysiological methods for estimating the relative contribution of stimulus envelope and fine-structure cues across a range of clinical populations using a binaural masking level difference (BMLD) paradigm. Stimulus envelope cues were manipulated by the using narrowband noise maskers (50 Hz) with the inherent envelope fluctuations of Gaussian noise (GN) or the reduced envelope fluctuations of low-noise noise (LNN). Fine-structure cues were manipulated by choosing signal frequencies and masker center frequencies of 500 and 4000 Hz. The availability of fine structure cues to the auditory system differs markedly between these two frequencies. Electrophysiological measures consisted of far-field evoked potentials targeting the P1-N1-P2 cortical components. Stimulus maskers were presented continuously at a spectrum level of 60 dB SPL while signal level was varied in descending steps. Threshold was estimated as the lowest signal level that generated a reliable N1-P2 response. Validation of cortical indices was achieved via behavioral measures of the BMLD. Signal level was varied using an adaptive three interval, three-alternative forced choice procedure with a three-down, one-up adaptive tracking rule. Raw thresholds and derived BMLD values using both measurement methods will be reported for a baseline population of 10 young, normal-hearing listeners.

5aPP28. Examining enhancement conditions with an auditory nerve model. Erica Hegland and Elizabeth Strickland (Purdue University, SLHS Dept, 500 Oval Drive, West Lafayette, IN 47907, ehegland@purdue.edu)

Psychophysical studies have shown that the ability to detect a signal in a masker may be improved by presenting a preceding sound (precursor) identical to the masker, an effect called overshoot. When the masker has no spectral notch around the signal frequency, overshoot has been predicted using a physiologically-realistic auditory nerve (AN) model in which gain is reduced by the precursor. This may simulate the medial olivocochlear reflex (MOCR). When there is a spectral notch in the masker at the signal frequency, the resulting improvement in signal detection is sometimes called enhancement. It has been suggested that enhancement may be due to adaptation of suppression, which could be related to gain reduction in the cochlea, or to adaptation of inhibition, which could occur more centrally. Physiological studies of the AN have shown either no adaptation of suppression [Palmer et al., J. Acoust. Soc. Am. 97, 1786-1799] or a reduction in suppression when the MOCR was elicited [Kawase et al., J. Neurophys. 70, 2533-2549]. The purpose of the present experiment is to use the AN model to investigate the relationship between gain reduction and suppression with stimuli used in previous studies of enhancement. [Research supported by NIH(NIDCD) R01 DC008327]

5aPP29. Modeling the effects of sensorineural hearing loss on temporal coding in the auditory nerve. David Axe (Weldon School of Biomedical Eng., Purdue Univ., 206 S. Martin Jischke Dr., West Lafayette, IN 47907, davidraxe@gmail.com), and Michael Heinz (Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907)

Recent psychoacoustical studies have suggested a temporal-fine-structure (TFS) deficit in listeners with sensorineural hearing loss (SNHL). Physiological studies generally have not found a reduction in the strength of TFS coding in auditory-nerve responses; however, a number of other effects on temporal coding have been found that may be related to these perceptual temporal processing deficits [e.g., Kale and Heinz, JARO (2010); Scheidt et al., Hear. Res. (2010)]. These physiological effects of SNHL include enhanced envelope (ENV) coding, changes in relative TFS/ENV coding that depend on stimulus bandwidth, broadened tuning, loss of tonotopicity for complex sounds, reduced latencies and traveling-wave delays, and altered temporal dynamics such as enhanced onset responses, faster adaptation, and slower recovery from stimulation. The current study evaluated which of these physiological effects were accounted for by an existing computational model of auditory-nerve responses [Zilany et al., JASA (2009)], and the extent to which these properties are predicted to depend on outer- vs. inner-hair-cell dysfunction. The degree to which existing models account for these effects of SNHL on temporal coding will help to elucidate the responsible mechanisms, and is important for future diagnostic and rehabilitative applications. [Work supported by NIH Grant No. R01DC009838.]

5aPP30. A modified channel model for the auditory peripheral system. YongHee Oh, Evelyn M. Hoglund, and Lawrence L. Feth (The Ohio State University / OH 43210, oh.172@osu.edu)

The correlated channel model (Zhang, 1995) was proposed as a modification of the (Durlach, et al., 1986) channel model by assuming that the internal noise in each peripheral channel is correlated. However, both models are limited because signals resolved into N channels by the peripheral processing unit are assumed to be statistically independent. Based on the results of several recent studies, the signals themselves may be correlated due to nonlinear properties of the basilar membrane, even before the addition of interchannel noise. In this study, the correlated channel model was extended using correlated and weighted signals, and the sensitivity index d' for a multi-tone detection task was derived. The performance of the modified model was characterized by parameter estimation, and compared with the results of a spectrotemporal integration experiment. [Research supported by a grant from the Office of Naval Research # N000140911017.]

5aPP31. Medial olivocochlear influence on stimulus-frequency otoacoustic emission input-output functions. James Dewey and Sumitrajit Dhar (Roxelyn & Richard Pepper Department of Communication Sciences and Disorders, Northwestern University, Frances Searle Building, 2240 Campus Drive, Evanston, IL 60208, jbdewey@u.northwestern.edu)

Modulation of cochlear mechanics by the medial olivocochlear efferent system is characterized by a reduction in active, outer hair cell-mediated amplification of basilar membrane motion. This increases cochlear thresholds and linearizes basilar membrane input-output functions for low-to-moderate stimulus levels. Significant efferent effects have also been observed for responses to higher stimulus levels, potentially reflecting changes in the mechanical properties of the cochlear partition. In humans, sound activated changes in stimulus-frequency otoacoustic emissions have been used as a tool for investigating the dynamics of the medial olivocochlear reflex. However, the degree to which the amplitude and phase of otoacoustic emissions are related to those of basilar membrane motion is not entirely clear. For the purposes of comparison with invasive physiological measurements in animals, stimulus-frequency otoacoustic emission input-output functions were obtained from human subjects in the presence and absence of contralateral acoustic stimulation. Medial olivocochlear effects were quantified in terms of the absolute change in emission level as well as the vector change, which incorporates changes in emission phase. The extent to which efferent modulation of emissions in humans reflects that observed in previous reports of basilar membrane motion will be discussed.

5aPP32. Comparison of pure tone thresholds obtained via automated audiometry and standard pure tone audiometry. David A. Eddins (Department of Communication Sciences & Disorders, University of South Florida, PCD 1017, 4202 East Fowler Avenue, Tampa, FL 33620, deddins@usf.edu), Joseph P. Walton (Department of Communication Sciences & Disorders, University of South Florida, PCD 1017, 4202 East Fowler Avenue, Tampa, FL 33620), Adam E. Dziorny (University of Rochester Medical Center, 601 Elmwood Ave., Rochester, NY 14642), and Robert D. Frisina (Department of Biomedical Engineering, University of South Florida, PCD 1017, 4202 East Fowler Avenue, Tampa, FL 33620)

It is likely that the role of automated audiometry will expand in both clinical and research settings in the next few years. A novel method for measuring pure tone thresholds using an automated threshold measurement method is reported here. The Automated Audiometry for the NIH Toolbox (AANT) test was developed for use in the NIH Toolbox multi-disciplinary evaluation battery which contains over 20 tests of sensory function and cognition. Development goals included low system cost, high accuracy, test administration time under 10 minutes, and automated calibration and measurement procedures suitable for use by evaluators without specialized training. Here we report the results of a validation study in which pure tone thresholds obtained using the AANT algorithm and hardware are compared to pure tone thresholds measured using the gold standard clinical method, standard audiometric hardware, and experienced examiners. Air-conduction thresholds will be reported for test frequencies of 500, 1000, 2000, 4000, 6000, and 8000 Hz using both methods. Additional measures for each subject include otoscopy, screening tympanometry, and the Hearing Handicap Inventory for Adults or Elderly. Data will be reported for 100 subjects between the ages of 4 and 85 years in age ranges.

5aPP33. Development of an evaluation system for cochlear implant. Zhenya Yang, Sheng Wang, Meng Yuan, and Haihong Feng (Shanghai Acoustics Laboratory, Chinese Academy of Sciences, No. 456, Shanghai Xu Hui Qu Xiao Mu Qiao Rd., Shanghai, China, yzy@mail.ioa.ac.cn)

The cochlear implant (CI) stimulation parameters such as pulse width, phase gap, pulse amplitude, and stimulation rate are the key and fundamental features in the objective evaluation of a CI device. In this study, an electronic evaluation system was developed to measure the above CI parameters for fully understanding the performance of a CI device. This system is multi-functional, powerful and flexible, which contains several modules, e.g. source signal generation, I/O synchronization, pulse sequence acquisition, data processing, and pulse/waveform display. A user-friendly graphic interface was developed including many useful functions and options to make different measurement and evaluation. The measurement results and original data can be easily exported from the evaluation system for further analysis. The accuracy and limitation of the system will also be discussed in this work. This system is suitable for CI parameter verification, speech processing research strategy evaluation, multi-electrode streaming, and CI quality examination. This work is supported by National Natural Science Foundation of China (11104316), and Shanghai Natural Science Foundation (11ZR1446000).

5aPP34. Sound source localization in the horizontal plane through the bilaterally applied bone-conducted ultrasonic hearing aids. Takuya Hotehama and Seiji Nakagawa (National Institute of Advanced Industrial Science and Technology (AIST), MOL105, 1-8-31 Midorigaoka, Ikeda, Osaka 563-8577, Japan, t-hotehama@aist.go.jp)

Bone-conducted ultrasound (BCU) can be perceived not only by normal hearing but also by the profoundly hearing impaired who cannot make use of a conventional hearing aid. BCU hearing aid (BCUHA), in which an ultrasonic carrier is amplitude-modulated (AM) by collected external sound and presented through a bone-conduction vibrator onto the mastoid portion, has been developed for the profoundly hearing impaired. To obtain useful information in order to realize accurate sound source localization through the BCUHAs. In this study, localization performance in the horizontal plane through bilaterally applied BCUHAs was investigated by psychological experiments. Results show that subjects hardly lateralized when the

BCUHAs were simply applied bilaterally with the double-sideband modulation method. On the other hand, the localization performances were improved when the inter-lateral intensity and time differences of presenting signals of BCUHAs were enhanced on the basis of those of the collected external sounds and when the "transposed modulation" was employed as the modulation method. Our results suggest the necessity of the inter-lateral co-operative system to modify the spatial parameters of the output signal for accurate sound localization through the BCUHAs.

FRIDAY AFTERNOON, 18 MAY 2012 HALL A, 2:00 P.M. TO 2:40 P.M.

Session 5pAAa

Architectural Acoustics and Psychological and Physiological Acoustics: Objective and Subjective Parameters of Spatial Impression in Performing Arts Spaces II

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Contributed Papers

2:00

5pAAa1. Subjective experiment on preferable reverberation of a musical practice room by professional players. Akihiro Nakajima (Graduate School, the University of Tokyo, Ce402, 4-6-1 Komaba, Meguro-ku, Tokyo, Japan, nakajima@iis.u-tokyo.ac.jp), Sakae Yokoyama, Sohei Tsujimura, Shinichi Sakamoto (I.I.S. the University of Tokyo, Ce401, 4-6-1 Komaba, Meguro-ku, Tokyo, Japan), Ami Tanaka, and Yoshihide Shiba (Nikken Sekkei LTD., 2-18-3 Iidabashi, Chiyoda-ku, Tokyo, Japan)

A musical practice room is very important space for players to improve their skills. However, no architectural design guideline of the musical practice room exists in Japan and the individual room is being designed based on architect's intuition and experiences. For a basic investigation aiming to establish the guideline, we are now investigating appropriate acoustic conditions of musical practice rooms for players by subjective experiments. In this study, we focused on reverberation time of the room and subjective experiments using a three dimensional sound field simulation were conducted for professional players including woodwind instrument players, string players and singers. In the experiment, a room having 6 conditions of reverberation time was virtually simulated in the sound field simulation system established in an anechoic chamber. As a result, a clear tendency of preferable reverberation time dependent on the difference of types of the instruments was found and a range of preferred reverberation time for a practice room was obtained.

2:20

5pAAa2. Measurement of acoustic characteristics in a traditional Korean Buddhist temple complex. Edwin Stephen Skorski III (Keimyung University, College of Architectural Studies, 1095 Dalgubeoldaero, Dalseo-Gu, Daegu, 704-701, Korea, sskorski@yahoo.com)

Traditional Korean Buddhist Temple complexes, typically found in rural, mountainous settings, hold great cultural, historical, and architectural significance in the Republic of Korea. These buildings, constructed primarily of wood, with little or no mechanical fasteners, create unique acoustic environments used for chanting, spoken word, music, and quiet meditation. Within the temple complex there are multiple halls and shrines of varying size and geometry that are utilized for worship. This presentation documents the room acoustic characteristics of a variety of worship spaces within a prototypical traditional Korean Buddhist Temple complex. These spaces include the main Buddha Hall as well as surrounding halls dedicated to additional Buddhas and Bodhisattvas. Using a laptop based measurement system; field measurements were conducted to document reverberation time (RT), clarity (C-50), speech transmission index (STI), and background noise levels. Data gathered is discussed and analyzed in regards to the primary use of each space. This research was conducted utilizing BISA Research Grant funds provided by Keimyung University.