Keynote Lecture

Invited Paper

8:20

Acoustic diode. Jianchun Cheng, Dong Zhang, Bin Liang, Xiasheng Guo, and Juan Tu (Key Laboratory of Modern Acoustics (Nanjing University), Ministry of Education, Nanjing, Jiangsu 210093, P.R. China)

Usually, waves can travel just as easily in either direction along a given path. The invention of electric diode, which acts as a one-way filter for the current flux, has marked the beginning of modern electronics and eventually led to worldwide revolutions in many aspects. Similar devices also exist for light and heat transmission. However, it is much more difficult to make such one-way devices for sound waves, another important form of classical wave with even longer research history than electric waves, because of the way sound waves move though a material. Recently, the first model of “acoustic diode” has been demonstrated both theoretically and experimentally to allow the acoustic energy to flow in only one direction. This device was fabricated by coupling a superlattice with a layer of ultrasound contrast agent microbubble suspension. A significant rectifying effect could be observed within two frequency bands at locations that agreed well with theoretical predictions. The development of the “acoustic diode” prototype will inspire the interests and investigations in the more practical and efficient acoustic rectifiers, which should have substantial significance for the applications of ultrasound devices in many practical areas such as medical ultrasound therapy and high resolution imaging.

TUESDAY MORNING, 15 MAY 2012 HALL A, 9:15 A.M. TO 12:00 NOON

Session 2aAAa

Architectural Acoustics, Noise, and ASA Committee on Standards: Classroom Acoustics in Asia

Kenneth Roy, Cochair
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Xiang Yan, Cochair
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Chair’s Introduction—9:15

Invited Papers

9:20

2aAAa1. Binaural room impulse response database acquired from a variable acoustics classroom. Zhao Peng, Siu-Kit Lau, Lily M. Wang (University of Nebraska – Lincoln, Durham School of Architectural Engineering and Construction, 1110 S. 67th St., Omaha, NE 68182-0816, zpeng@unomaha.edu), Sean Browne, and Kenneth P. Roy (Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604-3001)

Room measurements were conducted in a variable acoustics classroom mockup space (epod) at Armstrong World Industries in Lancaster, Pennsylvania, USA. Binaural room impulse responses were measured in the epod using a head and torso simulator. Five reverberation time (RT) scenarios were achieved with various combinations of absorptive wall panels and acoustical ceiling tile. These provided a range of mid-frequency RTs from 0.4 to 1.1 seconds. Three of the five RT scenarios were achieved using two different material configurations to also consider material location effects. For each of the eight material configurations, two student-teacher orientations were tested. One orientation had all desks facing the front of the room across the long dimension of the space, while the other had all furniture facing the front of the room along the short dimension of the space. Binaural impulse responses were measured at nine student positions in the longitudinal orientation and ten student positions in the transverse orientation, both using two teacher positions. This database of 252 binaural impulse responses from a variable acoustics classroom provides realistic test cases rather than simulations. These are being used for acoustic investigations on various topics including effects of classroom furniture orientation or acoustic material arrangements.

9:40

2aAAa2. Linking HVAC type and student achievement. Ana Jaramillo and Michael Ermann (Virginia Tech. Blacksburg, VA, anaja@vt.edu)

Research has long-linked HVAC system type to room noise levels, and it has long-linked room noise levels to student learning. Uninformed design decisions and the absence of policy combine with ineffective execution and insufficient enforcement to produce noisy mechanical systems for schools. Some school HVAC systems place both fans and compressors in the classroom itself, as part of DX units; others utilize remote compressors as part of chillers, but in-room fan motors as part of fan coil units; still other systems use remote AHUs and remote chillers so that neither compressor nor fan motor is exposed to the classroom. This pilot study intends to search for a broader link between HVAC system type and student achievement exams in elementary schools.

10:00

2aAAa3. Acoustic evaluation of classrooms in China—green campus workgroup. Sean Browne, Kenneth P. Roy (Armstrong World Industries, 2500 Columbia Avenue, Lancaster, PA 17604, sdbrowne@armstrong.com), and Jerry Li (Armstrong World Industries, 22 Floor, Cross Tower, 318 Fu Zhou Road, Shanghai 200001, China)

Armstrong, as a member of the China Green Building Council, has been working with the special subcommittee on Green Campus associated with Tongji University in Shanghai. The goal of this working group is to develop a green rating system for schools including acoustic design/performance in classrooms. In support of this rating development we have been pursuing research on classroom performance relative to meeting the current GB 50118-2010 and future GB 50099-2011 codes for schools. In conjunction with the Green Campus work group 10 schools were identified across China for study with pre/post architectural interventions. In this case a suspended acoustical ceiling was installed and both objective (sound) measurements, and subjective (student & teacher perception) surveys were conducted. Additionally it was desired to develop an objective assessment of student learning performance with changes in acoustical design and this was addressed through a joint development program with the University of Nebraska. Current results from these school evaluations are being presented.

10:20

2aAAa4. Music rehearsal room acoustics: comparisons of objectives and performance measures. Ron Freiheit (Wenger Corporation, 555 Park Drive, Owatonna, MN 55060, ron.freiheit@wengercorp.com)

A comparison of the acoustic performance criteria for high school music education rehearsal rooms to standard classrooms in the United States and Asia. High school music rehearsal rooms have acoustical requirements that are very different from traditional classrooms, where academic subjects are typically taught in a lecture-based setting. Due to the extended frequency range and dynamics of music rehearsals, most standard classroom acoustic treatments will not provide effective results for music rehearsal rooms. A number of acoustic measurements will be discussed – comparing rehearsal rooms and standard classrooms. Potential problems and solutions will be identified.

10:40–11:00 Break

11:00

2aAAa5. Tsinghua University student design competition—Schools. Jia Luo (Armstrong (China) Investment Co., Ltd., Shanghai 200001, China, armstrong.thinghua@gmail.com), Xiufang Zhao, Shuo Zhang, Kun Li, and Xiangdong Zhu

Armstrong (China) Investment Co., Ltd., with support from Tsinghua University sponsored a school design competition, and this represents the winning student entry. This competition was intended to encourage students to express their knowledge of architectural acoustics and noise control in the design of a school in which acoustical considerations are of significant importance. Design included considerations of 1) the general architectural building design including both shape and location of the building relative to transportation noise sources, and 2) the specific architectural acoustic design of rooms, including noise control both within and between rooms. The design scenario was for a new primary school designed to replace an older school building at a site near Tsinghua University. The primary school will include grade levels from 1 to 6, with approximately 1000 pupils. This school building will include normal classrooms (number and size based on student population and design standards), cafeteria, auditorium, library, music rooms, gym, dancing room, etc, as are normally found in such a primary school. The school will consist of a single building, which may be a multi-level building.

11:20

2aAAa6. Tongji University student design competition—Schools. Jia Li (Armstrong (China) Investment Co., Ltd, 318 Fuzhou Road, 22F Cross Tower, Shanghai 200001, China, armstrong.tongji@gmail.com), Gong Lv, Beiyang Duan, and Liyao Hu

Armstrong (China) Investment Co., Ltd., with support from Tongji University sponsored a school design competition, and this represents the winning student entry. This competition was intended to encourage students to express their knowledge of architectural acoustics and noise control in the design of a school in which acoustical considerations are of significant importance. Design included considerations of 1) the general architectural building design including both shape and location of the building relative to transportation noise sources, and 2) the specific architectural acoustic design of rooms, including noise control both within and between rooms. The design scenario was for a new primary school designed to replace an older school building at a site near Tongji University. The primary school will include grade levels from 1 to 6, with approximately 1000 pupils. This school building will include normal classrooms (number and size based on student population and design standards), cafeteria, auditorium, library, music rooms, gym, dancing room, etc, as are normally found in such a primary school. The school will consist of a single building, which may be a multi-level building.

Contributed Paper

11:40

2aAAa7. Sound absorption of periodically arranged flat surface and uneven surface. Jingjing Wang (Fraunhofer-Institute for Building Physics, Nobelstr. 12, D-70569 Stuttgart, sikalite@163.com)

Nowadays, thermally active concrete ceilings are frequently used in new buildings due to the requirement of thermal comfort and energy saving. But architectural design requires for planar surfaces so that the sound absorbers should not be installed below the ceiling even for guaranteeing the thermal efficiency. Thus a periodic arrangement of sound-absorbent strips or patches in concrete ceilings will be a good solution to this problem. The sound absorption of these two structures will be introduced in this report. Furthermore, a periodically uneven surface with rectangular profile is also commonly used at present, for example, as wooden floor. The sound absorbers can be hided in the grooves to control the acoustic environment. This configuration satisfies people's requirement of seeing the wood ceiling. The sound absorption of this groove structure will be introduced in this report as well. Besides, a parametric survey has been down on the performance of the periodically groove structure and porous absorber. Based on this survey, optimized examples for application are suggested.

TUESDAY MORNING, 15 MAY 2012 HALL B, 9:35 A.M. TO 1:00 P.M.

Session 2aAAb

Architectural Acoustics and Signal Processing in Acoustics: Multiple-Microphone Measurements and Analysis in Room Acoustics II (Lecture/Poster Session)

Boaz Rafaely, Cochair
br@ee.bgu.ac.il

Sam Clapp, Cochair
clapps@rpi.edu

Chair's Introduction—9:35

Invited Papers

9:40

2aAAb1. Precise and efficient localization of room reflections using compact microphone arrays. Walter Kellermann (Multimedia Communications and Signal Processing, wk@lnt.de), Edwin Mabande, Haohai Sun, and Konrad Kowalczyk

Recent works have shown that precise localization of dominant reflections in acoustic environments can be achieved even by relatively small arrays if advanced beamforming concepts are employed. To this end subspace-based and steered beamformers-based localization techniques are implemented in the so-called eigenbeam domain leading to eigenbeam (EB)-ESPRIT, EB-MUSIC, and EB-Minimum Variance Distortionless Response (EB-MVDR) beamforming. The resulting algorithms can be deployed with small off-the-shelf spherical microphone arrays and can then lead to acoustic maps using only a single multichannel recording. They can furthermore be used as a tool for efficiently inferring the geometry of a room. In this contribution we will present a comparison of several methods in real-world scenarios and illustrate their distinctive properties, including robustness issues and known limitations.

10:00

2aAAb2. Sound field diffusivity estimation using spherical microphone arrays. Craig Jin and Nicolas Epain (Bldg, J03, The University of Sydney, Sydney, NSW 2006, Australia, craig.jin@sydney.edu.au)

During the last decade, spherical microphone arrays have become increasingly popular in the acoustic community. These microphone arrays can be used for various applications such as beamforming, sound field reproduction and spatial sound field analysis. Within the context of sound field analysis and room acoustics, estimating the diffusivity of the sound field at a particular time instant is useful. In this paper we present an algorithm for evaluating the diffusivity of sound fields recorded by spherical microphone arrays. Results of numerical simulations show that this algorithm is more robust to the presence of sound sources in opposite directions to each other, as compared to previously proposed methods. In addition the algorithm provides an upper bound for the number of dominant sound sources.

10:20

2aAAb3. The use of multi-microphone measurements of directional and random incidence acoustical coefficients. Peter D'Antonio and Brian Rife (RPG Diffusor Systems, Inc., 651C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

This paper will review the use of multiple microphones in the measurement of the normal incidence absorption coefficient, ISO 10534-2, the random incidence absorption coefficient, ISO 354, the scattering coefficient, ISO 17497-1 and the diffusion coefficient, ISO 17497-2. Multiple microphones are used to quadruple the plane wave frequency range providing a bandwidth from 63 Hz to 4000 Hz in a single 150 mm square impedance tube. Random incidence measurements are accelerated by simultaneously measuring 12 impulse responses from two speakers and 6 distributed microphones. Scattering coefficient measurements can be very time consuming, due to the fact that at least 72 MLS averages are required per rotating table revolution. By using the dual source and simultaneous 6 microphone measurement technique, measurement times are reduced from 43 minutes to roughly 4 minutes for a 3 sec MLS stimulus. Diffusion coefficient measurements are greatly accelerated by simultaneously measuring 32 scattered impulse responses for a given angle of incidence. The process utilizes 32 fixed microphones 5 degrees apart, 32 MOTU preamps and an MLS stimulus. The 32 recorded scattered signals are deconvolved with the MLS stimulus to determine the 32 impulse responses from which polar responses and the diffusion coefficient are calculated.

10:40–11:00 Break

Contributed Papers

11:00

2aAAb4. Evaluation method of sound field diffusion using sound intensity measurements. Kohta Sugiura and Akira Omoto (Kyushu University, Shiobaru 4-9-1, Minami-Ku, Fukuoka-City, Fukuoka 815-8540, Japan, kohta.sugiura.175@s.kyushu-u.ac.jp)

Diffuseness of sound field is an essential item for evaluating the characteristics of the sound field. Up to now, several methods for evaluating the degree of diffusion have been proposed. In this study, the method called VSV (Virtual Source Visualizer) is proposed, which effectively visualize direct and reflected sound waves by using the instantaneous intensities in three orthogonal directions obtained from the impulse responses. The VSV casts the projection of dominant reflection onto the virtual sphere, which has a panoramic image of the sound field as a texture. Also, the parameter UAD (Uniformity of Arrival Direction) is introduced that is calculated by using the histogram of the arriving directions of intensities. The effective examination of the degree of diffusion would be expected by the combination of these visualized information from VSV and quantitative evaluation with UAD.

11:20

2aAAb5. Effects of coupling between loudspeaker and wall on impulse response measurement. Di Wu, Fangshuo Mo, and Jianmin Ge (Tongji University, No. 1239, Siping Road, Shanghai 200092, China, wdi0225@163.com)

Impulse response measurement constitutes a major part in room acoustical measurement. To measure an impulse response, the loudspeaker issues a relatively long signal, which is counteracted by the reflected sound field, and then its radiation impedance would undergo a change. Volume velocity generated by the loudspeaker is determined not only by excitation signal and loudspeaker parameters, but also by the coupling between the loudspeaker and the walls. Calculating the impulse response by deconvolution of sound pressure and excitation signal introduces a bias. The paper adopts an analogy of electric, force and acoustics, fully analyzing the effect of change in radiation impedance on impulse response measurement, and runs an experiment to verify the effects.

11:40

2aAAb6. Objective sound field analysis based on the coherence estimated from two microphone signals. Martin Kuster (Phonak AG, Laubisrütistrasse 28, 8712 Stäfa, Switzerland, martin.kuster@phonak.com)

The coherence estimate function represents the coherence function for discrete-time and finite-length time signals. In order to avoid bias error in the estimation of the required spectral densities, there is always an averaging mechanism in either time, frequency or space involved. This averaging has important consequences in room acoustics because diffuse field equations are then applicable to reverberant fields. It will be shown how the coherence estimates from diffuse and reverberant fields differ as a function of the averaging constants. For reverberant fields it will further be shown that the dependence between coherence estimate and averaging constants is defined by the direct-to-reverberant energy ratio and the reverberation time. Finally, the existing analytical expression for the coherence estimate as a function of the direct-to-reverberant energy ratio is extended to several primary sources.

12:00

2aAAb7. Design of three-dimensional microphone array on polyhedrons. Jaehyung Lee and Jong-Soo Choi (Chungnam National University 305-746, aerojhl@cnu.ac.kr)

The localization of sound sources in a three dimensional space has been recognized as an important research topic in acoustics. Many acoustic measurement techniques in three-dimensional space have used spherical shape of array to optimize sensor configuration and increase its performance and spatial resolution. In this study, the design of microphone array on polyhedral surfaces is proposed to introduce an easy way of building arrays and to achieve the measurement capability for all direction as a spherical array does. Five symmetrical polyhedrons are used and the simulation results are compared. MEMS microphones are used in arrays and the tests are made in an anechoic environment to validate the performance of arrays. MEMS sensor makes it easy to build a three-dimensional array and enables to implement large number of sensors in a small area. Sensors are built on a printed circuit board (PCB) which becomes a sub-array of polyhedrons. The array output is processed using conventional beamforming method to localize a source's position. The influence of sound wave diffraction around the polyhedron corner on beamformer output is discussed. [Work supported by National Research Foundation of Korea (NRF) grant funded by the Korea government (MEST) (No. 2010-0014978)]

12:20

2aAAb8. On the measurement of directivity index for adaptive directional hearing aids. Buye Xu, Ivo Merks, and Tao Zhang (Starkey Labs., Inc., 6600 Washington Ave. S., Eden Prairie, MN 55344, buye_xu@starkey.com)

Adaptive directional microphone arrays have been widely utilized in hearing aids to help wearers understand speech in noisy environments. However, a practical and objective way of measuring the benefit of this technique is not obvious. One commonly used measure for directional microphone arrays is directivity index (DI), which evaluate the arrays' capability to attenuate a diffuse noise field. However, the DI has rarely been considered for the Adaptive directional microphone arrays because a truly diffuse noise field is required for the measurement. This paper investigates the feasibility of measuring the DI of adaptive directional hearing aids in a diffuse noise field generated by a loudspeaker array playing uncorrelated noise signals simultaneously. The requirement for the degree of diffuseness

of the noise field is studied based on numerical simulations for first-order and second-order adaptive microphone arrays. The possibility of measuring the DI in both anechoic and non-anechoic conditions is investigated. The accuracy of the proposed approach is further discussed.

The following abstract will be presented in poster format. The poster will be on display and the author will be at the poster from 12:40 p.m. to 1:00 p.m.

2aAAb9. A two-microphone method for localization of multiple speech sources using complex exponential transform of phase differences. Xiaoyan Zhao, Jie Tang, Lin Zhou (School of Information Science and Engineering, Southeast University, Nanjing 210096, P.R. China, xiaoyanzhao205@yahoo.com.cn), and Zhenyang Wu (Key Laboratory of Underwater Acoustic Signal Processing of Ministry of Education, Southeast University, Nanjing 210096, P.R. China)

This paper proposes a two-microphone method for localization of multiple speech sources by utilizing speech's sparse attribute in time-frequency domain. The proposed method estimates time-delay of each source by applying complex exponential transform to the inter-channel phase differences (IPDs). In order to improve the performance in noisy environment, the proposed method selects time-frequency points with high SNR. The method obtains the initial time-delay estimate of each speech source by utilizing the IPDs at low frequencies, and then iteratively updates the time-delay by utilizing the whole selected points. With the complex exponential transform on IPDs, the proposed method takes full advantage of the high-frequency phase information, not requiring phase compensation for IPDs at high frequencies. Experiments have been conducted to study the effect of the exponential factor on the performance of the proposed method and to compare the performance of the proposed method with the generalized hard clustering algorithm. Experimental results show that the proposed method achieves an optimal performance when the exponential factor ranges between 0.8 and 1.2. Comparisons of the results show that the proposed method outperforms the GHCA algorithm under different noise and reverberation conditions, and the performance improvement increases as the SNR is decreased.

TUESDAY MORNING, 15 MAY 2012 THEATRE FOYER, 9:20 A.M. TO 12:40 P.M.

Session 2aAAc

Architectural Acoustics and National Council of Acoustical Consultants: Student Design Competition

Philip Robinson, Cochair
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David Woolworth, Cochair
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The Technical Committee on Architectural Acoustics of the Acoustical Society of America with support from the Robert Bradford Newman Student Award Fund and the National Council of Acoustical Consultants are sponsoring the 2012 Student Design Competition that will be professionally judged at this meeting. The purpose of this design competition is to encourage students enrolled in architecture, engineering, physics, and other university curriculums that involve building design and/or acoustics to express their knowledge of architectural acoustics in the acoustic design of a mixed use building planned for the Mong Kok district of Hong Kong, to include offices on the lower floors, luxury hotel rooms in the middle floors, and a nightclub on the top floor.

Submissions will be poster presentations that demonstrate room acoustics, noise control, and acoustic isolation techniques in building planning and room design. The submitted designs will be displayed in this session and they will be judged by a panel of professional architects and acoustical consultants. An award of $1250.00 US will be made to the entry judged "First Honors." Four awards of $700.00 US will be made to each of four entries judged "Commendation." Awards are made possible through a grant from the Wenger Foundation and by the Newman Student Award Fund.

Session 2aAO

Acoustical Oceanography, Animal Bioacoustics, and Underwater Acoustics: New Technologies for Monitoring Fish—Active and Passive

Orest Diachok, Cochair
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K. Sawada, Cochair
ksawada@fra.affrc.go.jp

Contributed Papers

9:20

2aAO1. A comparison of community structure from the southeastern and central Bering Sea shelf: insights gained from acoustic backscatter. Jennifer L. Miksis-Olds, Samuel L. Denes (Applied Research Laboratory, Penn State, PO Box 30 Mailstop 3510D, State College, PA 18604, jlm91@psu.edu), and Joseph D. Warren (Stony Brook University, 239 Montauk Hwy, Southampton, NY 11968)

A two year time series of acoustic backscatter was acquired from the Middle Domain of the southeastern and central Bering Sea shelf starting in 2009. A three-frequency echosounder system was integrated into NOAA oceanographic moorings at these locations and provided information for the classification of backscatter into 4 biological categories: small (1-5mm), medium (5-15mm), and large (15-30mm) crustaceans, and resonant scatters. The seasonal pattern of backscatter intensity was tightly coupled to sea ice dynamics at both mooring sites, but the community structure and timing of zooplankton blooms differed between sites. Winter 2009 was a light ice year on the southeastern shelf compared to heavy ice presence in 2010. Comparison of backscatter intensity and structure between years at this location provides information about how sea ice extent impacts upper trophic level dynamics. Insights gained on the relationship between ice and community structure through analysis of acoustic backscatter between these two sites provides important information for predicting the ecosystem response in this area to variable and potentially decreasing seasonal ice extent associated with global climate change. [Work supported by ONR]

9:40

2aAO2. Effects of multiple scattering, attenuation, and dispersion in waveguide sensing of fish. Purnima Ratilal, Mark Andrews, and Zheng Gong (Northeastern University, Department of Electrical and Computer Engineering, 360 Huntington Ave, Boston, MA 02115, purnima@ece.neu.edu)

An ocean acoustic waveguide remote sensing system can instantaneously image and continuously monitor fish populations distributed over continental shelf-scale regions. Here it is shown theoretically that the areal population density of fish groups can be estimated from their incoherently averaged broadband matched filtered scattered intensities measured using a waveguide remote sensing system with less than 10% error. A numerical Monte-Carlo model is developed to determine the statistical moments of the scattered returns from a fish group. It uses the parabolic equation to simulate acoustic field propagation in a random range-dependent ocean waveguide. The effects of (1) multiple scattering, (2) attenuation due to scattering, and (3) modal dispersion on fish population density imaging are examined. The model is applied to investigate population density imaging of shoaling Atlantic herring during the 2006 Gulf of Maine Experiment. Multiple scattering, attenuation and dispersion are found to be negligible at the imaging frequencies employed and for the herring densities observed. Coherent multiple scattering effects, such as resonance shifts, which can be significant for small highly dense fish groups on the order of the acoustic wavelength, are found to be negligible for the much larger groups typically imaged with a waveguide remote sensing system.

Invited Paper

10:00

2aAO3. Acoustic scattering measurements by an ultra-broadband transducer using multilayer piezoelectric elements. Kazuo Amakasu (Res. Center for Advanced Sci. and Technol., Tokyo Univ. of Marine Sci. and Technol., 4-5-7 Konan, Minato-ku, Tokyo 108-8477, Japan, amakasu@kaiyodai.ac.jp), Yuka Mishima (Tokyo Univ. of Marine Sci. and Technol., Konan, Minato-ku, Tokyo 108-8477, Japan), Tohru Mukai (Hokkaido Univ., Minato, Hakodate, Hokkaido 041-8611, Japan), Kouichi Sawada (Natl. Res. Inst. of Fisheries Eng., Fisheries Res. Agency, Hasaki, Kamisu, Ibaraki 314-0408, Japan), and Toyoki Sasakura (Fusion Inc., Daiba, Minato-ku, Tokyo 135-0091, Japan)

A Langevin-type broadband transducer has been built using multilayer piezoelectric elements. The resonance frequency of this element was 138 kHz, but the quality factor was very low and the transducer had broadband sensitivities due to Langevin structure. The useful frequency band was 20-150 kHz and the beam widths at 38 and 120 kHz were 21.2 and 6.6 degrees, respectively. An echo-sounding system has been constructed using commercially available equipments and the system calibration and acoustic scattering measurements have been conducted using a 2-ms linear-frequency-modulated signal. The system response has been successfully determined using a 20.6-mm-diameter tungsten carbide sphere as a standard target. Furthermore, the target strength (TS) spectrum of a 38.1-mm-diameter tungsten carbide sphere has been measured and the measured TS spectrum was in good agreement with the predicted TS spectrum based on the exact modal series solution. [Work supported by Grant-in-Aid for Scientific Research and TUMSAT.]

10:20

2aAO4. Prediction of acoustic properties of juvenile salmon, *Oncorhynchus keta*, for acoustic monitoring. Kouichi Sawada (National Research Institute of Fisheries Engineering, FRA, Hasaki 7620-7, Kamisu, Ibaraki 314-0408, Japan, ksawada@fra.affrc.go.jp), Tadahide Kurokawa (Ohoku National Fisheries Research Institute, FRA, Sinhama 3-27-5, Shiogama, Miyagi 985-0001, Japan), and Akihiko Hashiba (Sanriku Yamada Fisheries Cooperative Association)

To monitor the juvenile salmons (*Oncorhynchus keta*) swimming near surface, ventral aspect target strength (TS) of juvenile salmons were predicted by the prolate spheroid modal series model (PSMM). Soft X-rayimages of juvenile salmons (34.0-75.7 mm, in standard length, n=46) were taken from lateral and dorsal sides. The outlines of the swimbladders were digitized. TS pattern, which is a function of tilt angle, was calculated based on the morphological parameters of swimbladder and was averaged by assumed tilt-angle distributions. Normal distributions with nine different mean values (-20° – +20° at 5° step) and a constant s.d. of 20° were selected for mean TS calculations. The mean ratio of swimbladder height and the width was almost unity (s.d. 0.16). The maximum difference of the predicted mean TS by the difference of tilt-angle distributions became 1.4 dB, 2.3 dB, and 3.1 dB at 38 kHz, 70 kHz, 120 kHz, and 200 kHz, respectively. Using the model, TS=20logL-b, yielded TS=20logL-65.4 at 70 kHz, when a normal distribution with mean 0° and s.d. 20° was assumed.

10:40–11:00 Break

11:00

2aAO5. Evaluation of playback sounds by a newly developed dolphin-speaker. Yuka Mishima (Tokyo Univ. of Marine Sci. and Technol., 4-5-7 Konan, Minato-ku, Tokyo 108-8477, Japan, thank_you_for_email_5yuka@yahoo.co.jp), Keiichi Uchida, Kazuo Amakasu, Yoshinori Miyamoto (Tokyo Univ. of Marine Sci. and Technol., Konan, Minato-ku, Tokyo 108-8477, Japan), and Toyoki Sasakura (Fusion Inc., Daiba, Minato-ku, Tokyo 135-0091, Japan)

It is important to playback broadband sounds for the research of communication and echolocation of dolphins. Acoustic studies of dolphins mainly focus on recording of vocalizations and hearing abilities, but relatively few playback experiments have been conducted. To improve our understanding of their communication and echolocation abilities, an extremely broadband speaker, which is able to project communication sounds, whistles and burst-pulse sounds, and echolocation clicks, is anticipated. We developed a prototype Dolphin-speaker covering the frequency range from 30 kHz to 150 kHz within +/- 15 dB. Although the transmitting sensitivity of the prototype was almost flat at frequency band higher than 30 kHz, the sensitivity lower than 30 kHz was worse and it had some dips. Using two techniques, composition of the prototype and a Langevin-type element that has resonance at about 10 kHz and equalization for the ripples, the newly developed Dolphin-speaker has been improved having flat transmitting sensitivity from 5 kHz and 150 kHz within +/- 6 dB. A variety of sounds from captive dolphins were played back by the Dolphin-speaker. Visual and numerical comparisons of the playback sounds and originally generated sounds will be presented.

11:20

2aAO6. Classification of three tuna species in enclosures by using target strength spectra measured by a broadband split-beam system. Tomohito Imaizumi, Koki Abe (National Research Institute of Fisheries Engineering, Fisheries Research Agency, 7620-7, Hasaki, Kamisu, Ibaraki 314-0408, Japan, imat@affrc.go.jp), Yong Wang (Furuno Electric Co., Ltd., 9-52 Ashihara-cho, Nishinomiya, Hyogo, Japan), Masanori Ito, Ikuo Matsuo (Department of Information Science, Tohoku Gakuin University, 2-1-1 Tenjinzawa, Izumi-ku, Sendai, Japan), Yasushi Nishimori (Furuno Electric Co., Ltd., 9-52 Ashihara-cho, Nishinomiya, Hyogo, Japan), and Tomonari Akamatsu (National Research Institute of Fisheries Engineering, Fisheries Research Agency, 7620-7, Hasaki, Kamisu, Ibaraki 314-0408, Japan)

The selective capture of tunas; bigeye tuna (*Thunnus obesus*), skip jack tuna (*Katsuwonus pelamis*) and yellow fin tuna ((*Thunnus albacares*) is important for Japanese seine net fisheries. Classification of the species composition by using acoustic information before catching them will significantly contribute for the selective catch. Target strength spectra (TSSP) of each species were measured by a broadband split beam system. Each tuna species was separately kept in a enclosure, which sized 8 × 8 m square and approximately 5 m in depth. This system was able to measure not only TSSP but also a swimming track of individual in 3D. The differences of TSSP among three species were confirmed. For example, the target strength value of begeye whose tilt angle was 0 degree about 10 dB higher than skip jack tuna one. Swimbladder sizes were measured for each species by soft X-ray because skip jack tuna are physoclist species, and others are physostome species. There were differences of the swimbladder shape between bigeye tuna and yellow fin tuna. The TSSP seemed to be depend on not only body size (fork length), but also swimbladder shape. The TSSP could be useful information to classify among tunas.

11:40

2aAO7. Discrimination of *Diaphus theta* and *Euphausia pacifica* using underwater irradiance. Matsuura Tomohiko (Tokyo University of Marine Science and Technology, 4-5-7, Konan, Minato-ku, Tokyo 108-8477, Japan, mtsr@affrc.go.jp), Sawada Kouichi (National Research Institute of Fisheries Engineering, Fisheries Research Agency, 7620-7, Hasaki, Kamisu, Ibaraki 314-0408, Japan), and Uchikawa Kazuhisa (Japan Sea Fisheries Reasearch Institute, Fisheries Research Agency, 1-5939-22, Suido-cho, Niigata 951-8121, Japan)

This study proposes scatterers discrimination method by using underwater irradiance in addition to the usual acoustic method. Mean volume backscattering strengths (MVBS) at 38 and 120 kHz and underwater irradiance were measured at fixed locations in the north-western Pacific from 24 to 27 August, 2008. Two sound scattering layers (SSLs) conducting diel vertical migration (DVM) at different depths were observed and were identified as *Diaphus theta* and *Euphausia pacifica* by net samplings, respectively. During DVM, both species followed mostly the constant light level. The mean irradiance at the lower and the upper outlines of SSLs for *D. theta* were ranged from -81.0 to -57.8 dB re μ mol/m^2/nm and that for *E. pacifica* were ranged from -50.5 to -42.1 dB re μ mol/m^2/nm, respectively. The mean and standard deviation of $\triangle$ MVBS corresponding to the scattering layers of *D. theta* and *E. pacifica* were -4.7 ± 2.7 dB and 7.7 ± 3.2 dB, respectively. Scatterers discrimination using irradiance parameter in addition to usual $\triangle$ MVBS parameter was found to be more effective to distinguish both species from other scatterers.

12:00

2aAO8. A simple resonator technique for determining the acoustic properties of fish schools. Craig N. Dolder (Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78712, dolder@utexas.edu), and Preston S. Wilson (Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713)

Acoustic resonators have been used for decades to measure material properties, but only recently have they been applied to determine the effective medium properties of largely inhomogeneous materials. One-dimensional resonators can be used in both a laboratory and field setting to determine the speed and attenuation of acoustic waves through fish schools. Artificial arrays of fish schools are placed in a one-dimensional resonator. After correcting for elastic waveguide effects, the resonances give effective phase speeds and attenuations. The application of this technique to artificial arrays of fish, and how it can be applied to live fish in both a laboratory setting and deployed in the sea will be discussed.

12:20

2aAO9. Force estimation and prediction from time-varying density images. Srinivasan Jagannathan (MIT, 5-435, 77 Mass. Ave., Cambridge, MA 02139, srini.jag@gmail.com), Berthold Horn (MIT), Purnima Ratilal (Northeastern University), and Nicholas Makris (MIT, 5-212, 77 Mass. Ave., Cambridge, MA-02139)

We present methods for estimating forces which drive motion observed in density image sequences. Using these forces, we also present methods for predicting velocity and density evolution. To do this, we formulate and

apply a Minimum Energy Flow (MEF) method which is capable of estimating both incompressible and compressible flows from time-varying density images. Both the MEF and force-estimation techniques are applied to experimentally obtained density images, spanning spatial scales from micrometers to several kilometers. Using density image sequences describing cell splitting, for example, we show that cell division is driven by gradients in apparent pressure within a cell. Using density image sequences of fish shoals, we also quantify 1) intershoal dynamics such as coalescence of fish groups over tens of kilometers, 2) fish mass flow between different parts of a large shoal, and 3) the stresses acting on large fish shoals. IEEE Transactions on Pattern Analysis and Machine Intelligence 33(6), pp. 1132-1146 (2011)

TUESDAY MORNING, 15 MAY 2012 S224 + S225, 9:20 A.M. TO 12:40 P.M.

Session 2aBA

Biomedical Acoustics: Biomedical Ultrasound Imaging Instrumentation

Lei Sun, Cochair
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Qifa Zhou, Cochair
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Invited Papers

9:20

2aBA1. High frame rate velocity-coded speckle imaging platform for coherent blood flow visualization. Alfred C. H. Yu and Billy Y. S. Yiu (Medical Engineering Program, The University of Hong Kong, alfred.yu@hku.hk)

Non-invasive imaging of blood flow at over 100 fps (i.e. beyond video display range) is known to be of clinical interest given that such a high frame rate is essential for coherent visualization of complex hemodynamic events like flow turbulence. From a technical standpoint, getting into this frame rate range has became possible with the advent of broad-view ultrasound imaging paradigms that can track motion over an entire field-of-view using few pulse-echo firings. Leveraging on an imaging paradigm known as plane wave excitation, a novel high-frame-rate flow visualization technique has been developed to depict both blood speckle motion (using B-flow imaging principles) and flow velocities (using conventional color flow imaging principles). Experimental demonstration of this method has been carried out using a channel-domain research platform that supports real-time pre-beamformed data acquisition (SonixDAQ) and a high-throughput processing engine that is based upon graphical processing unit technology (developed in-house by the authors). In a case with a 417 fps frame rate (based on 5000 Hz pulse repetition frequency and slow-time ensemble size of 12), results show that high-frame-rate velocity-coded speckle imaging can more coherently trace fast-moving blood flow than conventional color flow imaging. Acknowledgement: Research Grants Council of Hong Kong (GRF 785811M)

9:40

2aBA2. An open system for intravascular ultrasound imaging. Weibao Qiu, Yan Chen, Wang Fai Cheng, Yanyan Yu, Fu Keung Tsang, Jiyan Dai (The Hong Kong Polytechnic University, qiu.weibao@connect.polyu.hk), Qifa Zhou (University of Southern California), and Lei Sun (The Hong Kong Polytechnic University)

Cardiovascular disease is the main causes of morbidity and mortality due to lumen stenosis and atherosclerosis. Intravascular ultrasound (IVUS) is able to delineate internal structures of vessel wall with fine spatial resolution. However, IVUS is insufficient to identify the fibrous cap thickness and tissue composition of atherosclerotic lesions, the key factors to stage atherosclerosis and determine appropriate treatment strategies. Currently, novel techniques have been developed to determine tissue composition, which require an open IVUS system to accommodate these techniques for comprehensive plaque characterization. This paper presents the development of such an IVUS system with reconfigurable hardware implementation, programmable image processing algorithms, and flexible imaging control to support an easy fusion with other techniques to improve the diagnostic capabilities for cardiovascular diseases. In addition, this IVUS utilized a miniaturized ultrasound transducer constructed by PMN-PT single crystal for better piezoelectric constant and electromechanical coupling coefficient than traditional PZT ceramics. Testing results showed that the IVUS system could offer a minimum detectable signal of 25μV, allowing a 51dB dynamic range at 47dB gain, with a frequency range from 20MHz to 80MHz. Finally, phantom imaging and in vitro vessel imaging were conducted to demonstrate the performance of the open system for IVUS applications.

10:00

2aBA3. A dual-modality system for imaging anatomy and vasculature in live mouse embryos. Parag V. Chitnis (F. L. Lizzi Center for Biomedical Engineering, Riverside Research, 156 William St., 9th Floor, New York, NY 10038, pchitnis@riversideresearch.org), Orlando Aristizábal (Skirball Institute of Biomolecular Medicine, New York University School of Medicine, 540 First Avenue, New York, NY 10016), Ashwin Sampathkumar, Erwan Filoux, Jonathan Mamou (F. L. Lizzi Center for Biomedical Engineering, Riverside Research, 156 William St., 9th Floor, New York, NY 10038), Daniel H. Turnbull (Skirball Institute of Biomolecular Medicine, New York University School of Medicine, 540 First Avenue, New York, NY 10016), and Jeffrey A. Ketterling (F. L. Lizzi Center for Biomedical Engineering, Riverside Research, 156 William St., 9th Floor, New York, NY 10038)

An imaging system that combined high-frequency ultrasound (HFU) with photoacoustics (PA) for in vivo visualization of anatomical and vascular maps in transgenic mouse embryos is presented. The system consisted of a five-element, 40-MHz, poly-vinylidenefluoride-co-trifluoroethylene (PVDF-TrFE) annular array with a hole in the center to facilitate coaxial delivery of light produced by a 532-nm pulsed laser. A phosphate buffer-filled petri-dish with a central hole was placed on the abdomen of an anesthetized mouse and a laparotomy was performed to expose an intact uterus. The probe was raster scanned to achieve 3-D imaging. The central element of the array was excited with a high-voltage impulse synchronized with the light pulse. The resulting ultrasound echo and PA signal from each scan location were digitized from all five array channels and synthetically focused in post-processing. The embryonic vasculature depicted by the PA image was co-registered with anatomical features represented in the HFU image. The head and the abdomen of embryos were imaged; feasibility of real-time, spatially co-registered, dual-modality in vivo imaging of mouse embryos was demonstrated.

10:20

2aBA4. Comparison of conventional multiple line acquisition with plane wave and diverging wave imaging for cardiac applications: a simulation study. Ling Tong, Hang Gao, Hon Fai Choi, and Jan D'hooge (Lab. on Cardiovascular Imaging & Dynamics, Dept. of Cardiovascular Diseases, Catholic University of Leuven, UZ Herestraat 49 - box 7003, 3000, Leuven, Belgium, ling.tong@uzleuven.be)

When imaging the heart, multiple line acquisition (MLA) is commonly used to increase the frame rate. For a typical phased array, frame rate can be increased by a factor of 4 using 4MLA with a less focused transmit beam. Alternatively, plane wave or diverging wave could be used allowing for 16MLA. However, transmit compounding is required in order to keep the spatial resolution acceptable resulting in a gain in frame rate similar to the one of a 4MLA system. The aim of this study was therefore to directly contrast the performance of a conventional 4MLA system to a plane wave and diverging wave imaging system by computer simulation. The performance of different imaging systems were investigated by quantitatively evaluating the characteristics of their two-way beams (i.e. -6dB beam width, side lobes to main lobe energy ratio, main lobe centralization and side lobes asymmetry). The results showed that the conventional 4MLA and plane wave imaging were very competitive imaging strategies while operating at a similar frame rate. 4MLA outperformed in the near field (i.e. 10mm-50mm), while plane wave imaging had better beam profiles in the far field (i.e. 50mm-90mm). The optimal settings of the diverging wave imaging system requires further study.

10:40–11:00 Break

11:00

2aBA5. Pulse Wave Imaging (PWI) of the human carotid artery: An in vivo feasibility study. Jianwen Luo (Department of Biomedical Engineering, Columbia University, New York, NY, USA and Department of Biomedical Engineering, Tsinghua University, Beijing, China, luo_jianwen@tsinghua.edu.cn), Ronny Li (Department of Biomedical Engineering, Columbia University, New York, NY), and Elisa Konofagou (Departments of Biomedical Engineering and Radiology, Columbia University, New York, NY)

Noninvasive measurement of the pulse wave velocity (PWV) is of high clinical importance. Pulse Wave Imaging (PWI) has been previously developed by our group to visualize the propagation of the pulse wave and to estimate the regional PWV. The objective of this study was to determine the feasibility of PWI in the human carotid artery in vivo. The left common carotid artery of 8 healthy human subjects (27 ± 4 y.o.) was scanned in a long-axis view. The beam density of the 10 MHz linear array was equal to 16 beams so as to increase the frame rate to 1127 Hz for an imaging depth of 25 mm and width of 38 mm. The RF signals were acquired to estimate the velocity of the arterial wall using a 1D cross-correlation technique. Sequential wall velocity frames depicted the propagation of the pulse wave in the carotid artery within the field of view. Regional PWV was estimated from the spatiotemporal variation of the wall velocities and ranged from 4.0 to 5.2 m/s, in agreement with findings in the literature. PWI was thus proven feasible in the human carotid artery.

11:20

2aBA6. Accuracy of kidney stone size in conventional ultrasound Bmode imaging. Barbrina Dunmire (University of Washington, Applied Physics Lab, 1013 NE 40th St, Seattle, WA 98105, mrbean@u.washington.edu), Mathew Sorensen (University of Washington, Dept. of Urology, 1959 NE Pacific St., Seattle, WA 98195), John Kucewicz, Michael Bailey, Bryan Cunitz (University of Washington, Applied Physics Lab, 1013 NE 40th St., Seattle, WA 98105), Jonathan Harper (University of Washington, Dept. of Urology, 1959 NE Pacific St., Seattle, WA 98195), Oleg Sapozhnikov, and Lawrence Crum (University of Washington, Applied Physics Lab, 1013 NE 40th St., Seattle, WA 98105)

The objective of this study was to determine the accuracy of conventional ultrasound imaging in sizing kidney stones, since this can be a determining factor in the treatment protocol for fragments on the order of 5 mm. Ex-vivo human kidney stones 3 to 12 mm were imaged in a water bath at depths from 6 to 10 cm using a Philips HDI5000 and Versonics software-based ultrasound system with the C4-2 transducer. Stone sizes were estimated offline a) manually by a sonographer and b) through an automated contrast based edge detection algorithm. Stone size was consistently overestimated with both instruments and by both estimation methods. On average, size was overestimated by 1 to 2 mm for stones 6 cm deep, and the overestimation increased with increasing depth and system gain. The overestimation was independent of actual stone size. These results suggest there is an inherent error in conventional ultrasound that leads to overestimation of stone size. These results also validate the software-based instrument for future work toward 1) investigating the sources of overestimation and 2) testing new methods for improving the accuracy of size estimation. Work supported by NIH DK43881 and DK092197, and NSBRI through NASA NCC 9-58.

11:40

2aBA7. Through-transmission medical imaging using phase-insensitive pyroelectric ultrasonic detectors. Bajram Zeqiri, Christian Baker (Acoustics and Ionising Radiation Division, National Physical Laboratory, Hampton Road, Teddington, TW11 0LW, United Kingdom, bajram.zeqiri@npl.co.uk), Haidong Liang (Department of Medical Physics and Bioengineering, St Michael's Hospital, Southwell Street, Bristol, BS2 8EG, United Kingdom), Giuseppe Alosa (Acoustics and Ionising Radiation Division, National Physical Laboratory, Hampton Road, Teddington, TW11 0LW, United Kingdom), and Peter Wells (Institute of Medical Engineering and Medical Physics, Cardiff University, Cardiff, CF24 3AA, United Kingdom)

Ultrasonic Computed Tomography (UCT) has been unable to rival its x-ray counterpart in terms of reliably distinguishing different tissue pathologies. Conventional piezoelectric detectors deployed in UCT are phase-sensitive and it is well established that their use can give rise to phase-cancellation artefacts that mask true tissue structure. In contrast, phase-insensitive detectors are more immune to this effect, although sufficiently sensitive devices for clinical use are not yet available. This paper establishes proof-of-concept for a novel phase-insensitive transducer for UCT. The detector employs an acoustic absorber to convert received acoustic intensity into heat that is subsequently detected using the pyroelectric response of a thin piezoelectric membrane bonded intimately to the absorber. The paper explores UCT application of the phase-insensitive detectors, comparing with traditional detection methods. Results are presented for a range of detector apertures; tomographic reconstruction images being compared using stable two-phase phantoms containing inserts as small as 3 mm. The project has demonstrated that the new detectors are significantly less susceptible to refraction and phase-cancellation artefacts, generating realistic images in situations where conventional techniques were unable to do so. The novel detector holds promise as the basis of a new type of clinical UCT system.

12:00

2aBA8. Pulse compression in time-varying systems with applications in ultrasonic vibrometry and tissue elastography. James Martin (Georgia Institute of Technology, School of Mechanical Engineering, 771 Ferst Drive, Atlanta, GA, 30332-0405, james.martin@me.gatech.edu), Peter Rogers, and Michael Gray (Georgia Institute of Technology, School of Mechanical Engineering, 771 Ferst Drive, Atlanta, GA, 30332-0405)

Pulse compression is normally applied only to time invariant systems, as the variation of a system's properties during its interrogation violates assumptions of the compression process. However, there is an exact solution to the pulse-compression problem when the time variance satisfies specified criteria. These are the same criteria that are required for the operation of an ultrasonic vibrometer in the context of a tissue elastography system. They are the requirement that the variations be very small in comparison with the wavelength of the interrogating ultrasound signal and that they be within a single Nyquist band as sampled at the periodicity of that signal. The solution to this problem involves a step-wise interpolation of the static pulse-compression transfer function in the frequency domain. It is possible to show that this technique offers significant advantages in terms of measurement time and/or measurement resolution when the limiting source of noise is any other than ambient ultrasonic noise or thermal noise at the receiving transducer, because the acoustic energy in the region of interest is limited by regulatory standards for diagnostic ultrasound. The technique has been demonstrated both analytically and with numerical models. It has also been tested in laboratory experiments on tissue phantoms.

12:20

2aBA9. Sub-surface elastography based on the dynamic shear strain analysis. Kenbu Teramoto and Mahbub Hasan (SAGA University, 1-Honjo, Saga, Japan, miroku.teramoto@nifty.com)

Quantitative acoustical imaging methods estimate the characteristics of the tissue elasticity of a region of interest having somewhat inhomogeneity contrasted to the surrounding soft tissues. The aim of this research is to propose a novel near-field imaging method for the shear wave elastography and sub-wavelength imaging. The proposed imaging methodology utilizes the determinant of a covariance matrix which is composed of the orthogonal pair of the shear strain variations. The image reconstruction theory can be summarized as follows. 1) The distributions of the normal displacement of the skin surface is governed by the 2-dimensional wave equation in the Lamb-wave field. 2) When the single propagating wave front exists on the surface of the tissue without any stiffer regions, therefore, the orthogonal pair of the out-of-surface shear strains are linearly dependent each other. Therefore the determinant of a covariance matrix which is composed of the orthogonal pair of the shear strains becomes zero. 3) When a region of interest having somewhat inhomogeneity exists, scattered wave field arises in the Lamb-wave field. Consequently, the determinant becomes larger than zero because of the independency between the orthogonal pair over the region of the scattered and incident wave field.

TUESDAY MORNING, 15 MAY 2012 — S221, 9:20 A.M. TO 12:40 P.M.

Session 2aEA

Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials I

Michael Haberman, Chair
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Invited Papers

9:20

2aEA1. Acoustic metamaterials with negative parameters: a multiple scattering approach with examples. José Sánchez-Dehesa, Victor M. García-Chocano, Rogelio Gracià-Salgado, Francisco Cervera, and Daniel Torrent (Wave Phenomena Group, Universitat Politècnica de València, Camino de vera s.n. (Edificio 7F), E-46022 Valencia, Spain, jsdehesa@upvnet.upv.es)

A homogenization method is here developed in the framework of multiple scattering. The method will be described and the resulting semi-analytical formulas are employed to solve several examples in which the effective parameters of acoustic metamaterials are negative. We also present the experimental realization of a quasi two-dimensional acoustic metamaterial with negative bulk modulus. The metamaterial consists of a hexagonal array of cylindrical boreholes. Experiments are performed using a two-dimensional waveguide where a slab of seven layers has been fabricated and characterized to extract the effective dynamical parameters. It is demonstrated that, at the frequency region were the bulk modulus is negative, the impinging wave is totally reflected and the pressure amplitude exponentially decreases inside the slab. The skin-depth effect has been also studied as a function of the frequency. The data are well supported by band structure calculations and by the homogenization method developed in the framework of the multiple scattering theory. Work supported by ONR and MICNN (Spain).

9:40

2aEA2. Dynamic effective medium theory for periodic structures with application to acoustic cloaking metamaterials. Andrew Norris (Rutgers University, Mechanical and Aerospace Engineering, 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

Expressions are presented for the fully dynamic effective material parameters governing the spatially averaged fields in three dimensional periodic systems. The results, which are valid at any frequency and wavenumber, are obtained by using the plane wave expansion (PWE) method. The effective equations are of Willis form with coupling between momentum and stress. Applications to layered fluids are first illustrated, showing that the effective density must be anisotropic and frequency dependent. A metamaterial proposed for cloaking - Metal Water - will be considered in detail as a function of frequency.

10:00

2aEA3. Vibration energy of metamaterials with negative effective parameters. Yuri I. Bobrovnitskii (Mechanical Engineering Research Institute, 4, Griboedov Str., Moscow 101990, Russia, yuri@imash.ac.ru)

Metamaterials, having unusual wave properties and offering promising applications, received much attention in recent years. However there are many questions begging for answer. Some of them concerning energy characteristics of metamaterials with negative inertial and elastic effective parameters are examined in this paper. The main result of the paper represent simple equations that express the vibration (acoustic) energy of a metamaterial through its effective density and elastic modules and their derivatives with respect to frequency. Special attention is paid to negative values of these parameters. Revealed are rather severe restrictions on their possible values that follow from the derived equations. The results are illustrated on metamaterials known from literature.

10:20

2aEA4. Acoustic diode. Bin Liang, Xiasheng Guo, Juan Tu, Dong Zhang, and Jianchun Cheng (Institute of Acoustics, Department of Physics, Nanjing University, Hankou road 22, Nanjing 210093, China, liangbin@nju.edu.cn)

Usually, waves can travel just as easily in either direction along a given path. The invention of electric diode, which acts as a one-way filter for the current flux, has marked the beginning of modern electronics and eventually led to worldwide revolutions in many aspects. Similar devices also exist for light and heat transmission. However, it is much more difficult to make such one-way devices for sound waves, another important form of classical wave with even longer research history than electric waves, because of the way sound waves move though a material. Recently, the first model of "acoustic diode" has been demonstrated both theoretically and experimentally to allow the acoustic energy to flow in only one direction. This device was fabricated by coupling a superlattice with a layer of ultrasound contrast agent microbubble suspension. A significant rectifying effect could be observed within two frequency bands at locations that agreed well with theoretical predictions. The development of the "acoustic diode" prototype will inspire the interests and investigations in the more practical and efficient acoustic rectifiers, which should have substantial significance for the applications of ultrasound devices in many practical areas such as medical ultrasound therapy and high resolution imaging.

10:40–11:00 Break

Contributed Papers

11:00

2aEA5. Performance optimization of acoustic concentrator. Yu-ran Wang, Hui Zhang, and Shu-yi Zhang (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, wangxx1986@gmail.com)

Acoustic concentrator, focusing acoustic field and enhancing acoustic energy in a region, is presented. The concentrating performances and the scattering properties of the acoustic concentrator with multilayered alternative homogeneous materials at different frequencies are investigated by the frequency response analysis with finite-element method (FEM). The calculation results show that it has an optimized relation between the acoustic concentrating performances in the inner region and the scattering properties in the outer region of these concentrators. A kind of acoustic concentrators constructed by an array of cylindrical rigid scatterers in fluid is proposed. The multiple scattering theory (MST) is introduced to describe the superposition of the external incident acoustic field and the radiative scattered field induced by the array, by which the structure of the array and the performance of acoustic concentrator can be optimized. By the way, the MST can be used to estimate and optimize not only the acoustic concentrators but also other potential acoustic metamaterials. 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.

11:20

2aEA6. Effective dynamic constitutive parameters of acoustic metamaterials with random microstructure. Mihai Caleap, Bruce W. Drinkwater, and Paul D. Wilcox (Department of Mechanical Engineering, University of Bristol, Queen's Building, University Walk, Bristol BS8 1TR, U.K., Mihai.Caleap@bristol.ac.uk)

A multiple scattering analysis in a non-viscous fluid is developped in order to predict the effective constitutive parameters of certain suspensions of disordered particles or bubbles. The analysis is based on an effective field approach, and employs suitable pair-correlation functions in order to account for the essential feautures of densely distributed particles. The effective medium that is equivalent to the original suspension of particles is a medium with space and time dispersion, and hence, its parameters are functions of the frequency of the incident acoustic wave. Under the quasi-crystalline approximation, novel expressions are presented for the effective constitutive parameters, which are valid at any frequency and wavelength. The emerging possibility of designing fluid-particle mixtures to form acoustic metamaterials is discussed. Our theory provides a convenient tool to test ideas *in silico* in search for new

metamaterials with specific desired properties. An important conclusion of the proposed approach is that negative constitutive parameters can also be achieved by using suspensions of particles with random microstructures with properties similar to those shown in periodic arrays of microstructures.

11:40

2aEA7. Theory of sound propagation in porous media allowing for spatial dispersion. Navid Nemati, Denis Lafarge, and Aroune Duclos (LAUM, UMR6613, Avenue Olivier Messiaen, 72085 Le Mans, France, navid.nemati.etu@univ-lemans.fr)

We present here a new nonlocal theory of long-wavelength sound propagation in rigid-framed porous media saturated with a viscothermal fluid. For unbounded macroscopically homogeneous media, isotropic or having a preferred wave-guide axis; the symmetry of the problem suggests that the wave propagation should be described in terms of an Equivalent-fluid having frequency- and wavenumber-dependent density and bulk modulus. Based on considerations borrowed from electromagnetic theory, a definite procedure is proposed to compute these two quantities from microstructure. Using the finite element method to implement the computation procedure, the possible relevance of the new theory is tested in two simple types of 2D geometries: that of the so-called ultrasonic metamaterials made of an array of Helmholtz resonators, and that of an array of cylindrical circular solid inclusions.

12:00

2aEA8. Seismic wave attenuator made of acoustic metamaterials. Sang-Hoon Kim (Division of Liberal Arts and Sciences, Mokpo National Maritime University, Mokpo 530-729, R. O. Korea, shkim@mmu.ac.kr)

We suggest a new method of an earthquake-resistant design to support conventional aseismic designs using acoustic metamaterials. Our device is an attenuator of a seismic wave. Constructing a spherical shell-type surface waveguide that creates a stop-band for the seismic wave, we convert the wave into an evanescent wave for some frequency range without touching the building we want to protect. It is a simple and practical method to reduce the amplitude of a seismic wave exponentially. Controlling the width and refractive index of the waveguide, we can upgrade the aseismic range of the building as needed in order to defend it. It may applicable for social overhead capitals such as power plants, dams, airports, nuclear reactors, oil refining complexes, long-span bridges, express rail-roads, etc.

12:20

2aEA9. Mode hybridization at subwavelength scale in acoustic metamaterials. Ying Cheng and Xiaojun Liu (Laboratory of Modern Acoustics, Nanjing University, Nanjing 210093, China, chengying@nju.edu.cn)

The artificial acoustic metamaterials consisting of subwavelength resonator elements can exhibit properties beyond those found in nature. These unique properties were described by effective media approximation theory, which treat the response as the averaged effects of the individual element's resonance response and ignore the coupling interactions between the elements. This paper reports the mode hybridization at subwavelength scale in acoustic metamaterials composed of single-slit Helmholtz resonator arranged in two-dimensional square lattice with twist angle between adjacent elements in ΓX direction. The dispersion curves and the transmission spectra demonstrate that the strong interactions between elements are not negligible when phi=180 degree and could lead to novel coupled resonance modes which do not exist in uncoupled metamaterials of phi=0 degree. The adjunct elements oscillate in-phase for the symmetric hybridization mode and out-of-phase for the anti-symmetric mode. In addition, the hybridizations are very sensitive to the twist angle, which could be indispensable in tuning the transmission. The results may be used to develop novel metamaterials and functional acoustic devices in the future. Acknowledgements: This work was supported by the National Basic Research Program of China (2012CB921504), National Natural Science Foundation of China (11074124, 11104139, and 10904052), and Jiangsu Provincial Natural Science Foundation (BK2011542).

TUESDAY MORNING, 15 MAY 2012 S226, 11:00 A.M. TO 12:20 P.M.

Session 2aED

Education in Acoustics: Engaging in Effective Teaching Methods in Acoustics

Wendy K. Adams, Cochair
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S. K. Tang, Cochair
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Invited Papers

11:00

2aED1. Measuring the effect of instruction. Wendy K. Adams (University of Northern Colorado, CB 127, Greeley, CO 80639, wendy.adams@colorado.edu)

Teaching as a science: As part of the endless pursuit of teaching excellence faculty develop and each year modify course materials in an attempt to help students learn as much science as possible in a semester. The goals include teaching the content as well as an appreciation for the science and how it connects to our everyday lives. At the same time we expect our students to learn what it is to do science and be a scientist. How can we measure the effectiveness of our teaching methods and compare them from year to year and from university to university? Since the mid-90's physics instructors have been using the FCI (Force Concept Inventory) to measure something, conceptual understanding maybe, in first semester introductory physics. Many other conceptual inventories have followed. In 2005 the CLASS was developed to measure student's perceptions and beliefs about learning physics. Quite often faculty are surprised and disappointed in the outcomes of these measures. As a colleague so eloquently stated, "Students have a way of disappointing you." In this presentation, I will briefly present my approach to teaching with interactive engagement and how I have attempted to measure the results using a combination of measures.

11:20

2aED2. An interactive method for teaching circuit model construction for complex interconnected acoustic systems. Preston S. Wilson (Mech. Eng. Dept. and Applied Res. Labs. The University of Texas at Austin, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

Complex, highly interconnected acoustic systems can be difficult to model for students and inexperience practitioners. The systematic lumped-element circuit model construction method presented here is easy to learn and teach, and allows for rapid, error-free circuit model construction. The present author discovered the method in a book by Mario Rossi [Acoustics and Electroacoustics, Artech House Publishers (1988)] and has included it in the electroacoustic transducers course taught at the University of Texas at Austin since 2003. This method has been effective in this course and utilizes two types of effective instruction discussed in the scientific education literature: the use of interactive engagement and visual models.

11:40

2aED3. Animations and visualizations of teaching and learning building acoustics for civil engineers. Zainal Abidin Akasah and Chiew Siah Ng (Faculty of Civil and Enviromental Engineering Universiti Tun Hussein Onn Malaysia, Zainal59@uthm.edu.my)

Building acoustics is becoming one of the important considerations in the design of a building. Thus it is of utmost importance to have competent designers who are can integrate acoustics needs in the design stage of a building. However, building acoustics is quite a challenging subject to teach and learn. The traditional method of teaching and learning is not the best method to promote a good understanding of the subject matter. Therefore, alternative methods must be found to improve effectiveness in teaching of this subject matter. The purpose of this study is to review existing animations and visualizations tools that have the potential to be used in the teaching and learning of building acoustics. The scope of this study is limited to education of building acoustics for civil engineers. Selected applications were surveyed on their usefulness to engineering educators. The result indicate that Mediacoustic is one of the most suitable animation and visualization tools for teaching and learning of building acoustic. A sample of animation and visualization module was created by using Xara Xtreme 3.2 as addition added-value to existing Mediacoustic. The sample animations provide support in the teaching and learning of building acoustics.

12:00

2aED4. The blended approach for teaching architectural acoustics—a preliminary study. S. K. Tang (Department of Building Services Engineering, The Hong Kong Polytechnic University, Hong Kong, China, besktang@polyu.edu.hk), and Roy Kam (Educational Development Centre, The Hong Kong Polytechnic University, Hong Kong, China)

Teaching architectural acoustics in tertiary education is always a challenge not only because the acoustical effects are invisible but because these effects could be difficult to experience. This paper reports a preliminary study of exploring the blended approach that combines traditional classroom deliveries with online resources for teaching this topic in a Hong Kong university. The online resources of the proposed blended approach comprise (i) measurement of binaural pulse decays at many locations inside the university's auditorium; (ii) selected examples of pulse decays at locations of considerably difference in acoustical properties (RT, Clarity, etc.); and (iii) the mix between the decays and different music for demonstrating the influences of acoustical properties. Apart from traditional classroom deliveries, the students are free to experience the online resources as many times as they want and to reinforce their understanding of acoustical effects after class. The students' evaluation was generally positive about the proposed blended approach, with over 90% of them particularly indicating that the online resources (i) helped them understand the principles of architectural acoustics; (ii) have strengthened their learning skills in approaching the topic; (iii) helped them better relate the concepts learnt in class, and (iv) stimulated their interests in the topic.

Session 2aHT

Hot Topics: Community Noise Policy Development I

Marion Burgess, Cochair
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Chair's Introduction—9:15

Invited Papers

9:20

2aHT1. Updating the WHO guidelines on community noise. Rokho Kim (European Centre for Environment and Health Herrmann-Ehlers-Str. 10, 53113 Bonn, Germany World Health Organization Regional Office for Europe, rki@ecehbonn.euro.who.int)

Adverse health effects of community noise are a growing concern in many countries. To provide evidence-based policy guidance to the member states, the World Health Organization published Guidelines for Community Noise (1999), Night Noise Guidelines for Europe (2009), and Burden of Disease from Environmental Noise (2011). The Parma Declaration on Environment and Health (2010) urged WHO to develop guidelines on noise suitable to reduce children's exposure to noise, including that from personal electronic devices, recreation and traffic, especially in residential areas, at child care centres, kindergartens, schools and public recreational settings. Accordingly, the Noise Guideline Development Group (NGDG) was convened for renewing the WHO guidelines on community noise. The new guidelines, to be finalized by 2013, will reflect newly available evidence on adverse health effects of community noise from various sources. Newly emerged issues such as windmill noise and neighbourhood noise will be addressed. The whole process from formulation of the topics and choice of the relevant outcomes, evidence retrieval, assessment and synthesis through systematic review, to formulation of the recommendations will follow the standard WHO guidelines for guidelines development to eliminate any potential conflict of interests by ensuring the highest level of transparency and accountability.

9:40

2aHT2. Environmental noise management and sustainable urbanization. Dietrich Schwela (Stockholm Environment Institute, Environment Department, University of York. Heslington, York, United Kingdom, dietrich.schwela@york.ac.uk)

Each day 180,000 people are moving into urban areas. As a society develops, it increases its level of urbanization and industrialization and the extent of its transportation system. Each of these developments brings an increase in noise load. A major contribution to noise exposure comes from the sound emissions of vehicles, which are commuting over large distances every workday. There is a direct relationship between the level of development in a country and the level of noise impacting on its people. Without appropriate intervention, the noise impact on communities will escalate. Sustainable urbanization means the application of the concept of sustainable development to the field of urban planning. Six basic principles are being applied in order to achieve sustainable urbanization: Compactness, completeness, conservation, comfort, co-ordination and collaboration. However, sustainable urbanization is more than the application of these basic principles. It has also to do with resource limits, avoidance of their exhaustion and mitigation of environmental pollution. Environmental noise management can contribute to achieve these goals. In this paper the linkage between environmental noise management and sustainable urbanization is elaborated. The paper develops a framework on how environmental noise management can contribute to sustainable urbanization.

10:00

2aHT3. Purpose of the international consortium on noise issues in emerging and developing countries. Lawrence Finegold (Finegold & So, Consultants; 1167 Bournemouth Court, Centerville Ohio 45459, LSFinegold@earthlink.net), and Dieter Schwela (Stockholm Environment Institute, University of York, York, UK)

As part of the growing interest in developing appropriate concepts and approaches for a "Global Noise Policy", consideration needs to be given to how effective and affordable noise policies might need to vary based on factors which differ depending on the "state of development" of individual countries. Although there is no standard manner to distinguish between "developed", "developing" and "emerging" countries, it is obvious that countries do differ in terms of their level of technological development, their financial capabilities and the availability of other resources required for adequate management of community noise. They also differ in their level of knowledge about the effects of noise, their views about the proper role of national and local governments, and the availability of engineering techniques to control exposure to community and occupational noise. This paper describes the current International Consortium on Noise Issues in Developing and Emerging Countries as a forum to facilitate discussions and share relevant information among the Consortium participants and other interested acoustics professionals.

10:20

2aHT4. Future environmental noise and health research needs for policy. Stephen Stansfeld and Charlotte Clark (Queen Mary University of London, Old Anatomy Building, Charterhouse Square, London EC1M6BQ, United Kingdom, s.a.stansfeld@qmul.ac.uk)

There is increasing evidence of the effects of environmental noise on human health with studies linking noise exposure to higher risk of hypertension, stroke and even mortality. However, there are gaps in the evidence and a lack of robust exposure-response relationships. There is also debate about whether much of the health effects of road traffic are attributable to noise or air pollution. There is a need for research that quantifies the effect of noise on health and assesses the total burden of disease attributable to environmental noise. Recommendations are reported based on the findings of the European Network on Noise and Health, funded by the EU 7th Framework Programme, to show how this could be achieved.

10:40–11:00 Break

11:00

2aHT5. The environmental noise directive as a catalyst for change in noise policy. Simon Shilton (Acustica Ltd, Trident One, Styal Road, Manchester, M22 5XB, United Kingdom, simon.shilton@acustica.co.uk)

The Environmental Noise Directive entered into EU legislation in 2002, and subsequently into the national legislation within the 27 Member States. The Directive sets out a strategic framework for a consistent approach to the management of environmental noise within the EC through a cycle of strategic noise mapping, public consultation and action planning. The activities and deliverables required under the END have lead to noise mapping and noise action planning activities on a previously unprecedented scale in many countries, both within Europe and beyond. With the Directive approaching its tenth anniversary it is an appropriate time to look back at whether it has been a catalyst for change in the approach to noise policy within Europe; and how the current ongoing review of the Directive, and proposed development of a common method of assessment, may affect noise policy in the future.

11:20

2aHT6. Some challenges in developing community noise policy. Marion Burgess (University of NSW, Canberra, Australia, m.burgess@adfa.edu.au)

There are a number of measures of success for environmental noise policy. For the regulatory or enforcement agency the measure of success is the lack of (or the reduction in) complaints about the noise in the area. This is a clearly quantifiable measure. For those responsible for the source of the noise a successful environmental policy is one that has clear and specific criteria for compliance which can be met in a cost effective manner. Again this is a quantifiable assessment. For the community the measure of success is satisfaction with the aural environment. Such satisfaction is a subjective measure. When establishing a noise policy the regulatory agency must match an understanding and knowledge of the community expectations with a quantifiable measure of noise. This measure then becomes the basis for the implementation of amelioration measures for the noise generator as well as the mechanism for the enforcement agency to verify compliance. Measuring the noise levels in the community has become relatively easy with modern instrumentation. Establishing the appropriate criteria remain the major challenges and will be discussed in this paper.

11:40

2aHT7. Brief review of legal framework on environmental noise in Japan. Ichiro Yamada (Airport Environment Improvement Foundation, K5 Bld., 1-6-5, Haneda Kuhkou, Ohta-ku, Tokyo 144-0041, Japan, i-yamada@center.aeif.or.jp)

This paper makes a brief review of legal framework for the assessment of environmental noise in Japan and discusses issues of noise policy to be improved and needs to change or modify noise evaluation methods. For example, road traffic noise and aircraft noise are now evaluated using Leq metrics, but high speed railway noise is still evaluated using LASmax. It may cause a difficulty when evaluating the impact of compound noise exposure due to simultaneous road and railway traffic. On the other, level magnitude and frequency of C-weighted sound levels of noise events are still used as metrics for assessment and improvement of sound environment at public buildings such as schools, hospitals and so on near airfields and maneuvering grounds. Needs and requirements for sound environment may have greatly changed with the times. The author looks back over the way to use such metrics.

12:00

2aHT8. Noise policy in Germany. Christian Fabris (Umweltbundesamt, Wörlitzer Platz 1, D-06844 Dessau, christian.fabris@uba.de)

This paper is a summary of the principle and main noise policy instruments in Germany. It attempts to show a simple model of these instruments, embedded both in European and German federal state legislation. German legislation on noise is divided into several laws, ordinances and other regulations concerning the various sources of noise (traffic, industry, mobile machinery, sports grounds, etc.) Noise emissions are generally governed by European legislation. Examples are the so-called “Outdoor Directive” and the “Energy-using-Products Directive”. Other laws limit the noise exposure from noise sources. Another example is the implementation of the Environmental Noise Directive into German noise policy. This contains the principles to create feasible noise abatement plans considering public concerns. The planning of traffic routes as the most annoying noise sources in Germany is regulated in particular laws and ordinances for the respective sources. Noise exposure of the most stationary noise sources is limited by a national instrument of legislation, the “Technical Instructions on Noise Abatement – TA Laerm”. There are also some governmental economic development schemes which are related to noise criteria. Last but not least there is the environmental label “Blue Angel”, which awards several products which are outstanding quiet in their product family.

Session 2aMU

Musical Acoustics: Asian Wind Instruments

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Invited Papers

9:40

2aMU1. The resonance hole with membrane; a distinctive feature of East Asian transverse flutes. Akiko Odaka (Tokyo University of the Arts; 12-8, Ueno koen, Taito-ku, Tokyo 110-8714, Japan, odaka@ms.geidai.ac.jp)

The East Asia region, except for mainland Japan, has transverse flutes which have resonance holes with a membrane. The *dizi* in China, the *taegum* in Korea and the *fansō* in Okinawa are good examples of this. Generally, a reed caliber epidermis is used as the membrane. Chinese *dizi* players point out that a reed epidermis creates a louder and clearer resonance than other materials. In China, transverse flutes with a resonance hole appeared in the Song dynasty, when Chinese theatrical music became popular. A transverse flute was played as the main accompaniment with other stringed and percussion musical instruments. Ordinarily, theatrical plays were performed outside. It was under these circumstances that the resonance hole was added, creating a louder, clearer sound. The Korean *taegum* is thought to have existed since the Three Kingdoms period (57B.C.-668). Its resonance creates variegated timbre combined with *taegum*'s unique vibrato techniques. The Okinawan *fansō* has its origin in the Chinese *dizi* of the Ming dynasty. However, currently, musicians play an improved *fansō*, which has no resonant holes. This presentation will show the historical and musical background of the resonance hole and will include music recordings.

10:00

2aMU2. Vibro-acoustic analysis of wind instruments with membranes on resonance holes. Toshiya Samejima, Shiori Ide, and Yozo Araki (Kyushu University, 4-9-1, Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, samejima@design.kyushu-u.ac.jp)

Some Asian wind instruments have a membrane glued over a special hole. A Japanese flute "Shino-bue" has a special hole called "resonance hole" between the mouth-hole and the first tone-hole. The resonance hole is covered with a bamboo paper membrane called "Chikushi". "Dizi" in China and "Taegum" in Korea also have a similar structure. The vibration of the membranes gives the wind instruments their characteristic bright timbre, thereby making them distinctive from comparable Western instruments. To investigate the influence of the membrane upon acoustical properties of such a wind instrument more qualitatively, this paper develops a numerical method for calculating the sound field around a wind instrument with a membrane on its resonance hole, as a vibro-acoustic system. This method couples, the integral equation derived from the normally differentiated Kirchhoff-Huygens formula for the sound field around the thin obstacle, with a theoretical solution of the vibration equation for the membrane. The formulation of the developed method is confirmed by comparing numerical results with measured results for a simple model of the wind instrument. Effects of the tension of the membrane, the size and location of the resonance hole, are discussed through numerical calculations using the developed method.

10:20

2aMU3. Relating the harmonic-rich sound of the Chinese flute (dizi) to the cubic nonlinearity of its membrane. Chen-Gia Tsai (National Taiwan University, tsaichengia@ntu.edu.tw)

Among the flute-type instruments all over the world, only the Chinese flute (dizi) and the Korean taegum have a membrane covering a hole in the wall of the instrument between the embouchure hole and the uppermost finger-hole. Nonlinear vibration of the dizi membrane endows dizi tones with a bright quality, which is due to the harmonics in the frequency range of 4–7 kHz. We provided a Duffing model of the dizi membrane, finding good agreement between this model and experimental results. Furthermore, we suggest that wrinkling of the membrane may be critical to its cubic nonlinearity.

10:40–11:00 Break

11:00

2aMU4. Changes in acoustical design from ancient shakuhachi to modern shakuhachi. Shigeru Yoshikawa (Graduate School of Design, Kyushu University, 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, shig@design.kyushu-u.ac.jp)

The shakuhachi was originally introduced from China in the Tang dynasty around 750. Since this ancient shakuhachi has been preserved in the Shousouin of the Toudaiji temple, it is called the "Shousouin shakuhachi". This shakuhachi, which has six tone holes to play a Chinese diatonic scale (e.g., A-B-Db-D-E-Gb-A, D-E-Gb-G-A-B-D), was adapted to play a Japanese pentatonic scale (D-E-G-A-B-D) by removing the second tone hole (Gb) around early 16th century. Moreover, the positions of five tone holes were modified to make effective use of the pitch bending (e.g., Eb) by drawing down player's jaw and half-covering the tone hole(s) around the 17th century, and a scale pattern D-F-G-A-C-D was established. Since this shakuhachi (made from the root end of bamboo) was played exclusively by a group of wondering priests ("Komusou"), it is called the "Komusou shakuhachi" and regarded as the origin of the modern shakuhachi. Changes in acoustical design from the Sousouin shakuhachi to the Komusou shakuhachi are considered based on the input admittance calculated from the inner geometry. The blowing conditions of the Shousouin shakuhachi are estimated from the investigation carried out during 1948 to 1952. Some problematic points in cross-fingerings of the Shousouin shakuhachi are also discussed.

11:20

2aMU5. Sound production in Asian free reed mouth organs. James P. Cottingham (Physics Dept., Coe College, Cedar Rapids, IA 52402, jcotting@coe.edu)

The Asian free reed wind instruments typically employ a free reed strongly coupled to a pipe resonator. In these reed-pipes the same reed often operates on both directions of airflow and behaves as a blown-open or outward striking reed, with playing frequency above both the resonant frequency of the pipe and the natural frequency of the reed. The Asian instruments were known in Europe when the Western free reed instruments were developed about 200 years ago, but in the European instruments a free reed of fundamentally different design was used. This paper summarizes the important acoustical properties of the Asian free reed mouth organs, contrasting them with the free reed instruments of European origin. Instruments considered include the khaen and other free reed mouth organs with multiple pipes as well as instruments consisting of a single free reed pipe in which the effective acoustical length is varied by the use of tone holes. Acoustical measurements made on these instruments include studies of reed vibration and impedance measurements of the pipes, with particular attention to the coupling of the reed vibration with the pipe resonator.

11:40

2aMU6. Bamboo pipe wall vibrations in Asian free reed instruments. Miles Faaborg (Coe College 1220 First Ave NE, Cedar Rapids, IA 52402, milsivich@gmail.com), and James Cottingham (Coe College 1220 First Ave NE, Cedar Rapids, IA 52402)

Asian free reed instruments generally employ bamboo pipes, and the properties of bamboo are of current interest, especially in relation to pipe wall vibrations. Recent results on measured physical properties of bamboo as used in Asian free reed instruments are presented, including mechanical properties of bamboo reeds as well as pipes. Recent investigations have been made of wall vibrations in the bamboo pipes of free-reed mouth organs for mechanically excited pipes. Modal frequencies and mode shapes of a number of pipes were measured, and measurements of pipe input impedance were made, some of which suggested possible changes occurring as a result of damping the pipe vibrations [Cottingham, J. Acoust. Soc. Am. 114: 2348 (2010)]. The most recent work involves the study of pipe wall vibrations for a mechanically blown reed-pipe combination. This was done for undamped pipes and pipes heavily damped with sand or other damping material. Measurements were made of the internal and external sound fields as well as measurements of the wall vibrations. [Work partially supported by US National Science Foundation REU Grant PHY-1004860.]

Session 2aNSa

Noise, Animal Bioacoustics, and ASA Committee on Standards: Ground Transportation Noise III

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Contributed Papers

9:20

2aNSa1. Comparison of the insertion loss of diffusive noise barriers using scale model experiments. Chan Hoon Haan and Jung youn Lee (Chungbuk National University, chhaan@chungbuk.ac.kr)

A new design of an environmental friendly noise barrier was suggested which can be assembled by unit blocks. It was anticipated to decrease of noise levels because the shape of the blocks is so diffusive that it may reflect and diffuse the sound. The unit blocks also contain the soil and vegetation for landscape which can contribute the noise control as well. Four different designs of noise barriers were suggested considering various practical conditions of construction. In order to investigate the acoustical performances of the various diffusive noise barriers, different size of unit block including 30, 25, 20 and 15cm were introduced. 1/10 scaled models were made and the insertion loss of each model was measured in an anechoic chamber. Also, the difference of diffusive and flat surfaces of noise barriers was analyzed. As a result, it was found that there is no difference of insertion losses depend on the distance from the noise barriers. It was also revealed that the diffusive noise barriers have larger insertion loss than flat noise barriers. And the bigger the unit block is, the larger noise insertion losses were acquired.

9:40

2aNSa2. Noise from railway expansion under close watch. Johnny C. Y. Wong, Geli K. T. Ma, K. H. Lam, and C. L. Wong (Environmental Protection Department, Government of the Hong Kong Special Administrative Region, johnnywong@epd.gov.hk)

New railway projects are underway in Hong Kong, not only to strengthen transport connections across different districts, but also to provide a high-speed link to enhance Hong Kong's role as the southern gateway to Mainland China. While the new railway systems are expected to provide the much-needed travelling facilities, their construction would inevitably generate noise. These projects are usually subject to tight time-frames and quiet technologies would have to be used when night-works are critically wanted for meeting the deadlines. That major parts of those new railways are constructed in densely populated areas would also increase the technical difficulty in meeting stringent noise criteria. This paper will describe the legislative framework in controlling the construction noise impact in particular from mega railway projects, including (i) various issues considered at the early planning stage by identifying potential noise impacts and recommending effective noise mitigation measures; (ii) the permit systems that govern the construction phase of the projects; and (iii) a transparent monitoring and audit scheme, in order to safeguard the well being of the people being affected.

10:00

2aNSa3. An experimental and numerical investigation of the sound distribution in street canyons with non-parallel building façades. Kaj Erik Piippo and Shiu-keung Tang (The Hong Kong Polytechnic University, Hunghom, Kowloon, kaj.piippo@connect.polyu.hk)

In this paper the sound distribution in a street canyon is investigated both experimentally and numerically. The sound field inside a 1:4 scaled down model of a street canyon has been investigated. A line source was used as sound source in order to generate steady cylindrical wave propagation. The sound distribution on the façades were mapped in the frequency domain and previously presented as contour plots. In order to validate the measurements, two-dimensional numerical simulations were conducted using COMSOL Multiphysics software, which uses a finite element approximation method. A cross-section at the centre of the scale model was chosen to be compared with the 2D simulation results. The comparison showed reasonable agreement, especially for frequencies starting at 1000Hz and up. The experimental data was expressed in 1/24 octave band frequency, while the simulated data was a single narrowband frequency. In order to get a better agreement for frequencies below 1000Hz more simulated data has been generated and is presented in this paper. Furthermore, initial results of 3D simulations are presented, as well as the impulse response of the street canyon model.

10:20

2aNSa4. The relationship between different measurement methods of testing tire noise. Shi Zuoteng (Institute of ATongji University, No. 1239, Siping Road, Shanghai 200092, China, satoshi.dawn@gmail.com)

With the increasing demands on the noise, the noise of vehicles has become a problem to which the auto -industry attaches great importance. Tire noise measurement constitutes a major part in vehicles noise measurement. To measure tire noise, traditionally there are 3 way to accomplish it: Pass-by noise method, Trailer Method, Laboratory Drum Method. The measure result depends on different contact surface and different tire tread pattern. Different measurement method shows different result. To find out the relationship between different tire measurement method can not only save tire noise measurement expense, but also enhance the data conviction. The paper runs a huge amount of experiments by each tire noise measurement method to compare the difference and adopts analogy of material, force, and acoustic fully analyzing the relationship of different measurement methods.

10:40–11:00 Break

11:00

2aNSa5. Compatibility of traffic noise planning control and land use in intensive city. Weichen Zhang, Wenying Zhu, and Yude Zhou (Shanghai Academy of Environmental Sciences, No. 508, Qinzhou Road, 200233, zhvivil@gmail.com)

The optimal way to solve traffic noise pollution is the planning-control. But it is a problem that how to get a perfect balance between the control distance and the land use, especially in some intensive City, such as Shanghai, Hong Kong etc. This article intends to put forward some exploratory ideas about the distance control with the study on the sound field of urban traffic noise distribution. Different from Euro and USA, The traffic noise is a serious problem in most Asia City. At present, these governments limited the distance along the traffic roads to reduce the noise level, yet it restricted the land-use. We plans to sketch a 3D view spatial pattern different from the past horizontal sound field research. And then; we should propose a constructive restriction on the sensitive buildings along the distinct roads, such as the layout/height of the front and back buildings. This paper proposes a new idea about the planning space control along the traffic roads, and put forward different control requirements for sensitive buildings on heights/distance. It will be applied to solve the traffic noise of intensive cities, and promoted the harmoniously progress between cities' planning and land use.

11:20

2aNSa6. Investigation of building envelop design for effective traffic noise reduction in Hong Kong. Chi Chung Chiu, Wing Kwok Szeto, and Marco Chi Wai Wu (Environmental Protection Department, Hong Kong SAR, 26th Floor Southorn Centre, Hong Kong, ccchiu@epd.gov.hk)

Road traffic noise is a major environmental noise problem in the densely populated city of Hong Kong. The Government of Hong Kong is committed to address the problem and has adopted a series of proactive actions to tackle the problem. Due to the compact cityscape of Hong Kong with major roads running near high-rise residential buildings, besides the more conventional form of measures such as land use planning, roadside barrier and enclosure, innovative form of building envelop design is considered worth exploring. This paper will present investigations for effective traffic noise reduction from building envelop design in Hong Kong. Initial laboratory investigation of insertion loss of various building envelop designs, including different forms of plenum windows and window designs has been conducted. The laboratory investigation is the first step. Further study on building envelop design for practical application would be the next step. The innovative feature would have the advantage of providing considerable amount of noise reduction and a comfortable open-window environment at the same time to the residents.

11:40

2aNSa7. The research of resonance frequency influence by 1/20 scale model test. Xiangdong Zhu, Xiang Yan, Xiaoyan Xue (School of Architecture, Tsinghua University, Beijing, China, zxd@abcd.edu.cn), and Hexiang Jia (Zisen Environmental Protection Company Sichuan China)

Acoustical enclosed workshops are used in high-level noise machine, especially for natural gas compressor. By this method the noise source was closed in workshop. Combined with sound absorption construction can decrease the noise level can be decreased out of workshop. But these kinds of methods have disadvantages. The low frequency noise levels of this type machine are higher than other machine, especially of infrasonic sound. The typical shape of workshop is rectangular and their sizes are about ten by ten meters. So the resonance frequencies are coincidence with the noise frequency of machine. All of these lead to the "sound box effect", which means low frequency and infrasonic sound will be amplified. The test data of practical cases in China indicate that these methods can be increase the pollution of infrasonic sound. In this study, we test the resonance frequency of a 1/20 scale model. In order to find out what the influence of noise control issue.

TUESDAY AFTERNOON, 15 MAY 2012 THEATRE 2, 12:00 NOON TO 12:40 P.M.

Session 2aNSb

Noise: Numerical Methods in Noise I

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Contributed Papers

12:00

2aNSb1. A FEM formulation for sound propagation over porous materials. Hyun Hong, Siu-Kit Lau (University of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, hhong@huskers.unl.edu), and Kai Ming Li (Ray W. Herrick Labs., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2031)

It is well known that the propagation of sound is sensitive to the acoustical properties and thicknesses of the porous materials when the source is placed near them. An efficient yet accurate numerical scheme to compute the sound propagation over an extended reaction surface is needed for the prediction and control of environmental noise. The numerical scheme can also find its application in enclosed spaces with the installation of absorption materials on the reflecting walls. To meet these objectives, a finite element method (FEM) is explored in a pilot study for determining the sound field above a layered porous ground. The sound fields computed by the FEM formulation are compared with those calculated by exact analytical formulas due to a monopole source. Three types of porous materials: locally reacting materials, semi-infinite extended reaction materials, and extended reaction materials with an impedance backed layer, have been considered in the present study. It has been demonstrated that the FEM formulation provides an accurate numerical solutions for predicting the sound field above a flat porous ground.

12:20

2aNSb2. Numerical study on scattering and absorption by periodically arranged acoustical treatment at oblique incidence. Shuk Ching Cheung, Chunqi Wang, and Lixi Huang (Department of Mechanical Engineering, The University of Hong Kong, Pokfulam Road, Hong Kong, cindycheung@hku.hk)

The propagation of sound over an impedance strip has been a topic of interest in sound abatement design. Excess absorption by the periodical arrangement of two or more distinct impedance conditions has been shown by various theoretical and experimental studies. It is believed that the scattering by the impedance discontinuities can enhance the absorption in some designs. This gives motivation to design a more elaborate set of impedance distribution within one periodic module. In this study, the scattering and absorption by periodically-arranged acoustical treatment at oblique incidence is investigated using the spectral method of Chebyshev collocation. The effects on the sound absorption and reflection by the length of the repeating unit, the angle of incidence and scattering characteristics due to the discontinuities of the acoustical impedance are analyzed. Central to the method is the derivation of out-going waves which allows scattered sound of all directions to leave the computational domain without reflection. The full picture of scattering is captured and analyzed using a rather coarse set of grid suitable for further optimization studies.

TUESDAY MORNING, 15 MAY 2012 — HALL C, 9:20 A.M. TO 12:20 P.M.

Session 2aNSc

Noise and Animal Bioacoustics: Future of Acoustics: East and West

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Invited Papers

9:20

2aNSc1. Social networks and networking of scientists: benefits and drawbacks. Betina Hollstein (Hamburg University Chair of Microsociology, School of Business, Economics and Social Sciences Welckerstr. 8, 20354 Hamburg, Germany, betina.hollstein@wiso.uni-hamburg.de)

Topic of the presentation is the contribution of social networks and social network analysis with regard to global change and the future of Acoustics. What are the outcomes of coopera-tion and networking of scientists and how is networking be enhanced? The paper elaborates on different types of social networks (among scientists and among science and other societal actors, like industry, political actors etc.) and its respective outcomes. How do networks matter and what are gains and possible losses of networking? Emphasis is placed on different cultures and contexts of networking. With respect to governance of networks I distinguish between “organic” networks and “organized” networks. Finally, consequences for networking between scientists are discussed.

9:40

2aNSc2. Facts and ideas for the development of an integrated sound and health effects research in a globalized world. Peter Lercher (Division of Social Medicine, Medical University of Innsbruck, Austria, Peter.Lercher@i-med.ac.at)

The environmental health effects research in environmental acoustics often reveals substantial differences in the obtained results which consequently lead to different conclusions and implementations in administration and policy. This paper intends to discuss some of the possible reasons underlying these discrepant results from a socio-cultural and social medicine viewpoint. For this purpose three complementary approaches are outlined and respective examples are presented. First, a sound source related perspective is investigated to explain potential differences in health outcomes. Second, a context related perspective is used to show empirical evidence for the variety of the contextual frameworks possibly responsible for observed differences in outcomes or importance of moderating factors. Eventually, with a health outcome related perspective possible differences in the underlying morbidity structure and health concepts are explored as potential sources for discrepant results.

10:00

2aNSc3. A western perspective on research in underwater acoustics and acoustical oceanography. Michael J. Buckingham (Scripps Institution of Oceanography, University of California, San Diego, 9500 Gilman Drive, La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

Underwater acoustics and acoustical oceanography are both concerned with sound underwater, the distinction between them being that the former deals primarily with forward problems such as acoustic propagation and scattering, whereas the latter involves the inversion of sound fields to obtain information about the oceanographic environment. Over recent years in the USA and Europe, both disciplines have tended to move away from the use of dedicated acoustic sources, over concern about damage that such sources may inflict on marine mammals. Ambient noise has, to some extent, replaced dedicated sources as the sound field of choice, since it offers the prospect of returning a wealth of information about ocean processes but without the threat to the marine mammal population. To extract such information, the properties of the noise field itself must be well understood (underwater acoustics), as must the inversion procedures necessary to recover the information contained in the noise (acoustical oceanography). A brief introduction to the properties of ambient noise fields will be followed by several examples of noise inversions, the latter illustrating not only the utility of ambient noise inversions but also the complexity of typical noise fields in the ocean. (Research supported by the Office of Naval Research).

10:20

2aNSc4. A prospect of the future of automotive sound quality development. Koo Tae Kang (Hyundai Motor Company, kanggood@hyundai.com)

In the development of vehicle sound design, sound quality is getting more important as opposed to the traditional sound level design. To get the proper sound quality with the vehicle image, the sound targets of the vehicle is needed to be defined appropriately in the vehicle level, system levels, etc., which become the core part in the vehicle design procedure. In sound design of the electrified vehicles including fuel cell vehicle, hybrid vehicle, extended range EV, and pure EV, artificial sound design both for vehicle interior and exterior, gain popularities for pedestrian protections as well as the driver satisfactions. As for the sound characteristics of the future vehicles, new noise sources are expected to be more concerns than powertrain noise sources with reduced noise level. Thus, in design of future vehicles, the efforts of the vehicle sound development should shift from the traditional sound level reduction to sound quality design and sound synthesis dealing with new noise sources.

10:40–11:00 Break

11:00

2aNSc5. (R)Evolution in vehicle acoustics—sound design, warning signals and quiet cities. Klaus Genuit (HEAD acoustics GmbH, Ebertstr. 30a, 52134 Herzogenrath, Germany, klaus.genuit@head-acoustics.de)

The increasing electrification of the powertrain offers the unique opportunity for complete new sounds with respect to vehicle interior sound and exterior noises after an era of 125 years of combustion engine. With this expected development noise affected persons in urban areas hope for quiet cities and a better quality of life in general. In particular, the creation and preservation of quiet zones in cities is a special focus in European noise policy exploiting the potential of electric vehicles. However, with the expected engine concepts changes - besides the hopes - several conflicts and problems are seen. These concerns are related to apparent technological problems (infrastructure, costs, range limitations) as well as to noise issues probably leading to an decrease of pedestrian safety. This has caused a general demand for acoustical warning signals for quiet (electric) vehicles to alert blind and visually-impaired persons. Are the concepts recently argued really well-thought-out and sustainable? Recent and potential future sound developments are highlighted from different perspectives. Here, from the constituting frame of reference regarding public opinion, automotive manufacturers' considerations, legislative initiatives and political actions the potential design of electric vehicle sound is discussed.

11:20 a.m.–12:20 p.m. Panel Discussion

Session 2aPA

Physical Acoustics and Biomedical Acoustics: Acoustic Micro- and Nanofluidics I

John S. Allen, Cochair
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James Friend, Cochair
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Invited Papers

9:20

2aPA1. High frequency ultrasonic particle sorting. K. Kirk Shung (Univ of Southern California, Department of Biomedical Engineering, Los Angeles, CA 90089, kkshung@usc.edu), and Changyang Lee (Univ of Southern California, Department of Biomedical Engineering, Los Angeles, CA 90089)

Single particle sorting devices have been used in bioassay compartmentalization and cell sorting. Fluorescence-activated cell sorting (FACS) is a well-known example. This paper presents an acoustically driven particle sorting device integrated with a poly(dimethyl) siloxane (PDMS) microfluidic channel. The device consists of two independent and sequential processes, acoustic sensing and sorting. Hydrodynamically focused lipid microspheres flowing in the channel are non-invasively sensed by a low intensity high frequency ultrasonic beam at 30 MHz for quantitative measurement of backscatter. Following sensing, the particles are sorted via acoustic radiation force or acoustic tweezing by a high intensity beam generated by the same transducer. The device has been successfully used to separate a mixture of 50 μm and 100 μm lipid spheres by size. Its performance and potential applications will be discussed in this paper.

9:40

2aPA2. Particle sorting using an oscillating microbubble. Adrian Neild, Priscilla Rogers, and Lin Xu (Monash University, Clayton Campus, VIC 3900, adrian.neild@monash.edu)

The ability to sort suspended matter within complex fluid samples is a key part of the functionality of Lab-on-a-chip devices. This study investigates the use of microbubbles to achieve this task. Bubbles which vibrate due to acoustic excitation are very effective at concentrating energy which can cause strong acoustic microstreaming. This fluid motion brings particles very close to the bubbles' surface. When in close proximity to the bubble the Bjerknes force arising between the particle and bubble can be large enough to pull the particle out of the swirling flow which characterizes acoustic streaming. If this occurs the particle is captured on the bubble surface, otherwise the force balance, which is size and density dependent, is such that the particle will remain in the streaming flow, apparently rejected by the bubble. The bubbles can be excited at their resonance or by the presence of an acoustic standing wave which can be created by exciting the fluid chamber at resonance. In the latter case, the interaction between the standing wave and particles can be used to bring more particles into the vicinity of the bubble.

10:00

2aPA3. Continous separation of microparticles using standing surface acoustic wave in microchannel. Sehyun Shin, Jeonghun Nam, and Hyunjung Lim (Department of Mechanical Engineering, Korea University, Korea, lexerdshin@korea.ac.kr)

Manipulation of microparticles in heterogeneous complex colloids has become important in various research fields that use microfluidic devices, such as biochemical analyses and clinical diagnosis. Although various techniques for microparticle manipulation in microfluidics have been developed, further advancements are still required for highly accurate analyses. Microparticle manipulation techniques, which include microparticle focusing, tweezing, and separation, using surface acoustic waves (SAWs) have emerged and gained attention. Since SAW-based microfluidics has advantages of being non-invasive, being harmless to particles, and consuming low power intensity and so on, it has great potential for further advancements, especially for biochemical research field. Recently, SAW-based techniques which can manipulate a variety of microparticles have been developed in my group. To predict the behavior of microparticles in microchannel flow, advanced analytical model was developed, and validated with experimental results. In the experiment, the heterogeneous sample, which includes blood, engineering particles, encapsulated cells, sperms, etc., were separated successfully into homogeneous sample with the separation efficiency over 99%. In this presentation, developments of SAW-based microparticle manipulation techniques to date and recent results of my group will be reviewed. In addition, some recommendations for future work of novel applications of SAW will be suggested. Acknowledgement This research was supported by Nano. Material Technology Development Program (Green Nano Technology Development Program) through the National Research Foundation of Korea (NRF) funded by the Ministry of Education, Science and Technology (No. 2011-0020090)

10:20

2aPA4. Use of phononic materials in microfluidic & lab-on-a-chip manipulations. J. M. Cooper (School of Engineering, University of Glasgow, Jon.Cooper@glasgow.ac.uk)

The development of microfluidic systems is often constrained both by difficulties associated with the chip interconnection to other instruments, and by mechanisms that can enable fluid movement and processing. Surface acoustic wave (SAW) devices have previously shown promise in allowing samples to be manipulated, although designing complex fluid manipulations involves the generation of mixed signals at multiple electrode transducers. We now demonstrate a new and simple interface between a piezoelectric SAW device and a disposable microfluidic chip, involving the use of phononic structures, to shape the acoustic field. Such phononic structures can be designed in such a way that the interaction of the fluid within the chip structure is dependent upon the acoustic frequency, providing a new method to programme complex fluidic functions into a microchip. We demonstrate applications in biological sensing involving enrichment enrichment of cells, lysis, PCR and sensing, exploring the application of this chip based technology to Developing World Diganostics.

10:40–11:00 Break

11:00

2aPA5. Surface acoustic waves (SAW) accelerated microfluidic mixing for improved microcalorimetry in biochips. Alan Renaudin, Rémy Béland (Université de Sherbrooke, 2500 boul. de l'Université Sherbrooke, QC J1K 2R1, Canada, alan.renaudin@usherbrooke.ca), Jean-Pierre Cloarec (Universite de Lyon, Institut des Nanotechnologies de Lyon, Site école Central de Lyon, France), Yann Chevolot (Universite de Lyon, Institut des Nanotechnologies de Lyon, site École Centrale de Lyon, France), Vincent Aimez, and Paul G. Charette (Université de Sherbrooke, 2500 boul. de l'Université Sherbrooke, QC J1K 2R1, Canada)

By measuring very small local temperature changes, microcalorimetry is used to determine the rates of energy released or absorbed during biochemical reactions. The measurement signal-to-noise ratio can be significantly increased by accelerating the reaction kinetics by active microfluidic mixing. We present a biochip incorporating a self-referencing droplet-based microreactor consisting of a thermopile-based microcalorimeter (50 Ni/Au thermocouples in series) on a glass substrate with a surface acoustic wave (SAW)-based microfluidic mixing system on a $LiNbO_3$ piezoelectric substrate. In our design, the SAW mechanical energy is transmitted from the piezoelectric substrate through the glass substrate to the droplets via a water film which acts as a pseudo mechanical impedance matching layer. The cumulative energy released by a standard calorimetric test reaction (sucrose dilution) is measured with the system. Results show that, by overcoming the diffusion-limited reaction rate, SAW-accelerated mixing in the droplets increases the thermal power released during the experiment by a factor 2, increasing the measurement SNR by the same factor. This enthalpy measurement accuracy improvement makes the system well-suited to sensitive thermodynamic measurements on biochip devices.

11:20

2aPA6. Acoustically-driven microcentrifugation. Leslie Yeo and James Friend (RMIT University, Melbourne, VIC 3001, Australia, leslie.yeo@rmit.edu.au)

A microcentrifugation technique is demonstrated in which symmetry breaking of a planar surface acoustic wave (SAW) propagating along a piezoelectric substrate can generate rotational acoustic streaming in a nanoliter drop. The azimuthal flow is rapid, with linear velocities of several cm/s - such fast azimuthal streaming, shown to be chaotic beyond a threshold power, can be used to generate intense micromixing within the drop. Indeed, we show that the rate and yield of a variety of distinct chemical and biochemical reaction classes far exceed that obtained using ultrasonic or microwave-assisted mixing, and with considerably lower power. Further, samples can be atomised from a cheap paper-based microfluidic system, thus demonstrating the potential for direct interfacing with mass spectrometry following chemical synthesis. The microcentrifugation effect can also be exploited for particle manipulation and sorting. This is demonstrated for bioparticle concentration for rapid and sensitive pathogen detection or the separation of red blood cells from plasma for miniaturized diagnostic applications. In addition, it is also possible to separate two different particle species by size by exploiting the unique size-dependent scaling between the acoustic and drag forces acting on the particle.

Contributed Papers

11:40

2aPA7. Acoustic bubble sorting: contrast enrichment by primary radiation forces. Tim Segers and Michel Versluis (University of Twente, t.j.segers@utwente.nl)

Ultrasound contrast agents consist of a suspension of encapsulated microbubbles with radii ranging from 1 to 10 μm. Medical transducers typically operate at a single frequency, consequently only a small selection of bubbles resonates to the driving ultrasound frequency. Thus, the sensitivity can be improved by narrowing down the size distribution. Here, a simple lab-on-a-chip method is presented to acoustically sort microbubbles on-line by a piezoelectric actuator positioned perpendicular to a microfluidic channel in a PDMS chip. Bubbles are produced in a flow focusing geometry at a rate of 500 bubbles per second. The bubbles are characterized in the unbounded fluid to provide physical input parameters for a force balance model. Good agreement is found with the experimentally observed displacement as a function of the bubble radius in free space as well as in the confinement of the sorting chip. This novel sorting strategy may lead to an overall improvement of the sensitivity of contrast echo by at least an order of magnitude.

12:00

2aPA8. High-frequency acoustic atomisation: do Faraday's results still apply?. James Friend and Leslie Yeo (MicroNanophysics Research Laboratory, SECE, RMIT University, City Campus, 10.10.01 Swanston Street, Melbourne VIC 3001, Australia, james.friend@rmit.edu.au)

Atomisation using high-frequency surface acoustic waves offers a revolutionary means to form monodisperse aerosols for nanoparticle fabrication and pulmonary drug and stem cell delivery. The underlying mechanism of atomisation is more complex than presented in the literature: rather than a simple parametric or subharmonic Faraday capillary wave mechanism which is physically impossible in this system, capillary waves form from a previously unknown mechanism that transforms the 10 to 1000 MHz excitation into broadband capillary waves around 10 kHz. Capillary waves of a

particular *most dangerous* wavelength are driven to breakup and droplet formation. The new theory describes the droplet size with excellent accuracy, and fits experimentally peculiar behaviour obtained upon changing the viscosity and surface tension of the fluid to be atomised and the amplitude and frequency of the acoustic wave. In the presentation, genuine applications for the technology will be shown from our laboratory, *in vitro*, and animal *in vivo* studies followed by a thorough explanation of how the atomisation takes place including high-speed video of the phenomena of atomisation and details of our experiments and analyses.

12:20

2aPA9. Enhanced surface acoustic wave atomization via amplitude modulation. Aisha Qi (RMIT, qiaisha@gmail.com), Anushi Rajapaksa (Monash University), James Friend, Leslie Yeo (RMIT), and Peggy Chan (Monash University)

Recent developments in the miniature chip-based microfludic nebulization platform utilizing surface acoustic wave (SAW) atomizer for inhalation therapy offers distinct advantages over other conventional nebulizers. Such efficient hand-held nebulizer system also requires the optimization of the usage of available power systems in the simplest manner. Here, amplitude modulation (AM) is presented as a simple yet effective means of optimizing the power requirement. SAWs are nano meter order amplitude acoustic waves that originate as a result of the application of an alternating current onto a pair of single-phase unidirectional transducers patterned on a piezoelectric substrate. The effect of the AM at modulation frequencies of 500 Hz, 1 kHz, 5 kHz, 10 kHz, 20kHz and 40 kHz sinusoidal signals, on shear-sensitive biomolecules such as plasmid DNA and antibody molecules are shown to be minimal. Energy savings of around 40% can be obtained with more efficient atomization achieved with AM less than 10 kHz. Together with these advantages, AM having little effect of the mean aerosol diameter; particularly important when therapies are to be targeted for the deep lung regions, holds great promise for its use in the SAW nebulizers for non-invasive inhalation therapy. Acknowledgements are dedicated to Asthma Foundation Victoria (Australia).

TUESDAY MORNING, 15 MAY 2012 S423, 9:20 A.M. TO 12:40 P.M.

Session 2aPP

Psychological and Physiological Acoustics: Binaural Hearing and Cochlear Mechanics

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Contributed Papers

9:20

2aPP1. Fast measurement system and super high directional resolution head-related transfer function database. Guangzheng Yu, Yu Liu, Bosun Xie (South China University of Technology, No. 381, Wushan Rd., Guangzhou, P.R. China, 510641, scgzyu@scut.edu.cn), and Qizhu Zhong (China Mobile Limited)

Head-related transfer functions (HRTFs) describe the acoustical transmission process from a point sound source to two ears in the free field. They are vital to the researches of binaural hearing and virtual auditory display. Measurement is the most important approach to obtain HRTFs, but it is time-consuming. To accurate HRTF measurement, a computer-controlling measurement system is designed. The system consists of multiple sound sources at different elevations with a finest interval of 5° and a horizontal turntable with an azimuthal resolution of 0.1°. It is able to work in non-anechoic environment and suitable for both mannequin and human subject measurement. A practical example indicates that the system allows for measuring far-field HRTFs at 493 source directions within half an hour. By using the system, a super high directional resolution HRTF database for KEMAR mannequin is established. The database includes 3889 pairs of head-related impulse responses (HRIRs, the time domain counterpart of HRTFs) at distance of 1.0 m, elevation interval of 5° (from -45° to 90°), and azimuthal interval of 2.5°. Each HRIR is 1024-point length at 96 kHz sampling frequency and 24-bit quantization. The database is applicable to the research on binaural hearing and virtual auditory display.

9:40

2aPP2. Comparison of transfer functions between the actual pinna and the simple pinna model which is composed of a rectangular plate and three rectangular cavities. Yohji Ishii (Graduate School of Engineering, Chiba Institute of Technology, 2-17-1 Tsudanuma, Narashino, Chiba 275-0016, Japan, s0972004QT@it-chiba.ac.jp), Hironori Takemoto (National Institute of Information and Communications Technology, 2-2-2, Hikaridai, Seika-cho, Soraku-gun, Kyoto 619-0288, Japan), and Kazuhiro Iida (Chiba Institute of Technology, 2-17-1 Tsudanuma, Narashino, Chiba 275-0016, Japan)

A simple pinna model which is composed of a rectangular plate with three rectangular cavities (three-step model) has been proposed (Takemoto et al., 2010). The results of numerical simulations using FDTD method showed that the three-step model generates typical peak-notch pattern of HRTFs. However, it is not clear whether transfer functions similar to the subject's HRTFs are generated by the three-step model when the size of each part of the model is adjusted to that of the subject's pinna. Since the 1st and 2nd notches (N1, N2) and the 1st peak (P1) in HRTFs are known as the spectral cues for the front-back and the vertical localization (Iida et al., 2007), it is important that the frequencies of these notches of the pinna model agree to those of the actual HRTFs. In the present study, various three-step models whose sizes are adjusted to those of the subject's pinna are created, and the transfer functions are measured. The results show that the frequencies of N1, N2, and P1 for several models are almost same as

those of the subject's HRTFs. A part of this work is supported in part by Grant-in-Aid for Scientific Research (A) 22241040.

10:00

2aPP3. Analysis on the audibility in directional differences of head-related transfer function magnitudes. Yu Liu, Bosun Xie, and Guangzheng Yu (South China University of Technology, No. 381, Wushan Rd., Guangzhou 510641, P.R. China, janworc@gmail.com)

A head-related transfer function (HRTF) varies as a function of sound source directions. In a dynamic virtual auditory display (VAD) based on HRTF filtering, dynamic binaural synthesis requires HRTFs to be updated constantly to accommodate the transient virtual source direction relative to listener. A HRTFs set with appropriate directional resolution can generate smooth transition when switching between HRTFs, and at the same time, simplifies dynamic binaural synthesis. Based on the measured HRTFs from KEMAR mannequin and human subjects with supper high directional resolution, the directional difference in HRTF magnitude spctra, binaural loudness level spectra and interaural localization cues are analyzed in present work. Combined with the statistical results of psychoacoustic experiments, psychometric functions for directional discrimination of HRTF magnitudes are derived. Finally, in terms of resultant audibility in directional differences of HRTF magnitudes, the directional resolutions of HRTFs for dynamically synthesizing virtual source at various directions are suggested.

10:20

2aPP4. The distance-dependence of interaural level difference cues to sound location and their encoding by neurons the inferior colliculus – implications for the Duplex theory. Heath G. Jones, Jennifer L. Thornton, Kanthaiah Koka, and Daniel J. Tollin (Department of Physiology and Biophysics, University of Colorado Medical School, Aurora, CO 80045, hgjones@wisc.edu)

The Duplex theory posits that low- and high-frequency sounds are localized using two different acoustical cues, interaural time (ITDs) and level (ILDs) differences, respectively. Anatomically, ITDs and ILDs are separately encoded in two parallel pathways consistent with ecological and efficiency principles which state that neural systems evolved strategies to represent the full spectrum of sensory signals as experienced by an organism in its natural habitat. ILDs are location and frequency dependent such that lower and higher frequencies exhibit smaller and larger ILDs, respectively. Neurons throughout the auditory neuraxis encode ILDs for high-frequency sounds. However, although low-frequency ILDs are negligible, humans are quite sensitive to them and physiological studies report low-frequency ILD sensitive neurons. The presence of such neurons is at odds with the Duplex theory and ecological and efficiency principles. We suggest these discrepancies arise from inadequate understanding of the ecological acoustical environment. Via measurements in the chinchilla of acoustical ILDs and their encoding by inferior colliculus neurons the hypothesis is explored that low-frequency ILDs become useful when sound source distance is varied. We demonstrate that a population of neurons is sufficient to encode the frequency-dependent range of ILDs that would be experienced as a function of location and distance. (R01-DC01155)

10:40–11:00 Break

11:00

2aPP5. Analysis on the stability of spatial interpolation schemes for head-related transfer function. Yang Liu and Bosun Xie (South China University of Technology, No. 381, Wushan Rd., Guangzhou 510641, P.R. China, shenhua.liuyang@163.com)

Head-related transfer functions (HRTFs) are transfer functions from sound source to two ears. They vary as continual functions of source direction. Usually, measurement yields the HRTFs at discrete directions, and HRTFs at unmeasured directions should be estimated by spatial interpolation. There are several familiar interpolation schemes, such as adjacent linear interpolation, the global interpolation, bilinear interpolation and spherical-triangular interpolation. In practice, potential subject's head movement in HRTF measurement may result in error in measured HRTFs, which may in turn deteriorate the interpolation performance. In present work, the performances for interpolation schemes against the error caused by slight head movement in HRTF measurement are evaluated and compared. The results indicate that overall, global interpolation scheme is superior to others in terms of signal-to-distortion ratio in interpolated HRTFs when undergoing a slight head movement in HRTF measurement. Taking advantage of the analog relationship between HRTF spatial interpolation and signal panning or mixing methods for multi-channel sound, the present analysis can also be extended for evaluating the signal panning methods for multi-channel sound.

11:20

2aPP6. Efferent modulation of physiological and behavioral measures of cochlear mechanics. Sumitrajit Dhar, Wei Zhao, and James Dewey (Roxelyn and Richard Pepper Department of Communication Sciences and Disorders, Northwestern University, Evanston, IL, s-dhar@northwestern.edu)

Efferent control of cochlear mechanics is of interest to scientists and clinicians alike. The functional roles of the descending auditory neural pathway in various species is being actively investigated by multiple groups around the world. Its role in the human is also of interest, with signs of efferent involvement in attention, learning, protection from noise, and signal detection in noise. The final leg of the auditory efferent pathway extends from the superior olivary complex in the brainstem to the cochlea with direct termination of neurons of the medial branch on outer hair cells. Activation of the medial olivocochlear circuit alters outer hair cell gain thereby modulating cochlear mechanics. Otoacoustic emissions provide a convenient tool for probing the efferent pathway. We will present results of recent work by our group examining the modulation of otoacoustic emissions and behavioral hearing thresholds by the medial olivocochlear efferents. Results demonstrate a common mechanism driven by changes in both magnitude and phase in cochlear mechanics manifest in both otoacoustic emissions and behavioral thresholds. These findings have important implications for developing reliable tools for the quantification of efferent modulation of cochlear physiology for possible diagnostic and therapeutic purposes.

11:40

2aPP7. Similarity and cluster analysis on magnitudes of individual head-related transfer functions. Bosun Xie and Xiaoli Zhong (South China University of Technology, No. 381, Wushan Rd., Guangzhou 510641, P.R. China, phbsxie@scut.edu.cn)

Although head-related transfer functions (HRTFs) vary with individuals, non-individualized HRTFs are typically employed in virtual auditory display (VAD) due to the difficulty in measuring individual HRTFs. Similarity among the HRTFs of different individuals allows for customizing a matched set of HRTFs for a user from an existing database and thereby improving performance of VADs. The first step of HRTF customization is evaluation of the similarities among individual HRTFs. In the present work, based on a HRTF database of 52 Chinese subjects, the similarities are evaluated by calculating the directional-mean of normalized cross-correlation coefficient (MCC) of HRTF magnitudes between each pair of subjects and then applying cluster analysis to the resultant MCC. The results indicate that MCC depends on subject pair with values ranging from 0.934 to 0.562 (mean 0.842) and from 0.948 to 0.635 (mean 0.863) for the left and right ear, respectively. HRTF magnitudes for most subjects can be classified into six to eight clusters and represented by the corresponding cluster centers. Some singleton clusters are also observed on a few subjects, which reflect the diversity in HRTFs and should be careful in practice. Psychoacoustic experiment also validates above analyses.

12:00

2aPP8. Physics prospect of cochlear nonlinear signal processing. Chang-Cai Long (Department of Physics, Huazhong University of Science and Technology, Wuhan 430073, China, longzc01@mails.tsinghua.edu.cn), Lin Tian, and Fei Wang (Department of Physics, Huazhong University of Science and Technology, Wuhan 430073, China)

We demonstrate in a model that cochlea nonlinear signal processing characteristics, nonlinear tuning, nonlinear amplification, and two tone suppression, stem from a common physics base, nonlinear active force acting on basilar membrane, which decreases with vibration amplitude. This

unveiled cochlear physics prospect provides theory for the exploring of active force physiological mechanism, the interpreting and the improving of impaired hearing.

12:20

2aPP9. Suppression tuning of distortion product otoacoustic emissions in humans: results from cochlear mechanics simulation. Yi-Wen Liu (National Tsing Hua University, Hsinchu 30013, Taiwan, ywliu@ee.nthu.edu.tw), and Stephen T. Neely (Boys Town National Research Hospital, Omaha, NE 68131)

Is human hearing more sharply tuned than other mammals? This has been a heavily debated subject in the field of cochlear mechanics. The debate continues partially due to lack of non-invasive methods to estimate human cochlear tuning accurately. Recently, Gorga et al. (2011, JASA) derived tuning curves from suppression of distortion product (DP) otoacoustic emissions (OAEs) in normal-hearing human ears. Frequencies of the primary tones were varied from 0.5 to 8 kHz. Sharpness of tuning was analyzed in terms of the Q-value of equivalent rectangular bandwith, which ranged from 4 to 10 at the lowest stimulus level tested. The Q-values were similar to that of psychoacoustic tuning but lower than inferred from latencies of stimulus-frequency OAEs (Shera et al. 2002, PNAS). In the present work, we simulate DPOAE suppression based on a computer model of cochlear mechanics (Liu and Neely, 2010, JASA). The simulated DPOAE suppression tuning curves (STCs) resemble those obtained in experiments but discrepancies remain. At high frequencies, the simulated DPOAE STCs are not as sharply tuned as the magnitude response of traveling waves in the model. Confounding factors and interpretation of results will be discussed. (Supported by Taiwan's NSC and NTHU)

TUESDAY MORNING, 15 MAY 2012 — S222, 9:20 A.M. TO 12:40 P.M.

Session 2aSA

Structural Acoustics and Vibration and Noise: Machinery Noise and Vibration I

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Invited Papers

9:20

2aSA1. The evaluation of pipe corrosion through the use of ultrasonic guided wave and novel matching pursuit. Peter W. Tse and Xiaojuan Wang (SEEM, City University of Hong Kong, Tat Chee Ave Hong Kong, meptse@cityu.edu.hk)

Ultrasonic guided wave is in routine use of the nondestructive testing fields as an advanced technique. However, in guided wave based pipeline inspection, the accurate evaluation of the severity of defect is always a challenging task. That is, although the reflection signal in principle includes substantial defect information related to severity and other features of the defect, it is usually rather difficult to be interpreted because of the complexities of reflection process. To carry out the planned maintenance on defective pipelines accurately and efficiently, the ability of evaluating defect severity for pipeline inspection is very important in practical application of guided waves technique, particularly for the cases in which defects exist in the parts of pipeline where their accesses are difficult. In this paper, we propose a method based on matching pursuit for quantifying the severity of pipeline defect along axial direction. The optimized dictionary through introducing prior-knowledge about reflection components is constructed for interactive process of matching pursuit to efficiently decompose the required edge reflection components from defect reflection signal. The axial extent of defect can be then quantitatively evaluated by using obtained information. The experimental results are used to demonstrate the effectiveness of the proposed method.

9:40

2aSA2. Machinery vibration diagnostics using a statistical analysis on frequency bands of interests. Zhuang Li and Lei Jin (McNeese State University, Lake Charles, LA 70609, zli@mcneese.edu)

Machinery fault diagnostics is vital for safety and reliability of operation in order to avoid serious consequences such as production downtime and even human lives. As the vibration signals carry useful information to reflect the machinery conditions, many theoretical models have been established to describe the relationship between the vibration signals and the existence of damage. However, sophisticated data analyses are needed to provide insight to the machinery vibration. A statistical model was proposed by the authors based on that the vibration spectra at the Fourier frequencies are exponentially distributed. Such a statistical model is an effective screening technique for damage identification and health monitoring in rotating machinery. While the previous statistical model was based on the full frequency-range spectra, in this research the model has been modified to focus on frequency bands of interests only. Such modifications improve the performance of the fault detection using the statistical model. Experimental vibration data were collected to validate the statistical model.

Contributed Paper

10:00

2aSA3. Fault diagnosis of railway roller bearing based on vibration analysis and information fusion. Bin Chen (School of Automation, Beijing University of Posts and Telecommunications, No. 10, West Tu Cheng Road, Haidian District, Beijing 100876, P.R. China; Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, chenbin@mail.ioa.ac.cn), Zhaoli Yan, Xiaobin Cheng, and Wei Liu (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)

Roller bearing is an important mechanical element of railway vehicle. It usually has defects in outer race, inner race or balls due to continuous metal-metal contacts in high-speed operating conditions. For the reason of impurity in lubricant oil, measuring locations and background noise, fault features extracted from vibration signal directly in time-domain or frequency-domain are unstable or uncertain, which may seriously affect the diagnosis accuracy. This paper presents a diagnostic method based on vibration analysis and information fusion. In the method, the signal analysis methods such as wavelet packet are processed to depress background noise of collected vibration signals. Considering that vibration energy at characteristic rotational frequency may increase with defect on a particular bearing element, they are extracted as fault feature vectors, which are used to train support vector data description (SVDD) classifiers. To reduce recognition uncertainty of single fault classifier, each classifier is regarded as independent evidence, and all evidences are aggregated by Dempster's combination rule. Experiment results show that the proposed algorithm can improve the diagnosis accuracy of roller bearing.

Invited Paper

10:20

2aSA4. Design and experiment of a cradle truss type floating raft system. Zhang Feng, Bai Zhenguo, Liu Xiaobin, and Yu Mengsa (No. 222, East Shanshui Road, Wuxi Jiangsu, 214082, China, zhangfeng5304@163.com)

In this paper, a new design concept of an isolated cradle trusslike structure to support vibrating machinery is proposed. The acoustic and vibratory energy transmission in this isolation system is investigated to improve the vibration isolation performance of conventional floating raft system. The finite element method and experimental measurements are used to understand the dynamic behavior of this cradle truss. In conjunction with experiments, a vectorial four pole parameter numerical model is used to identify power flow of the floating raft isolation system. The proposed floating raft design achieved a vibration reduction of 30–40dB in the frequency range of 20–600Hz. It is also shown that the loss factors of the cradle truss configured floating raft are up to 4dB than conventional floating flat raft.

10:40–11:00 Break

Contributed Papers

11:00

2aSA5. Fault diagnosis method using support vector machine and errors-in-variables for rotating machines. Hyungseob Han and Uipil Chong (University of Ulsan, 680 - 749, overhs@naver.com)

As rotating machines play an important role in industrial applications such as aeronautical, naval and automotive industries, many researchers have developed various condition monitoring systems and fault diagnosis systems by applying artificial neural networks. In order to increase performance of a classifier based on a neural network, it is most important to extract significant features of measured signals and to apply suitable features into a diagnosis system according to the types of the signals. Therefore, this paper proposes a neural-network-based fault diagnosis method combining Support Vector Machines (SVM) as a classifier and AR coefficients as feature vectors by Errors-In Variables (EIV) analysis. The system extracted feature vectors from sound, vibration and current signals and evaluated the suitability of feature vectors depending on the classification results and training error rates by changing AR order and adding noise. From the experimental results, it is concluded that classification results using feature vectors by EIV analysis indicate more than 90% stably for less than 10 orders and noise effect comparing to linear predictive coding (LPC).

11:20

2aSA6. Numerical analysis comparison of joint matrix and transfer matrix approach for acoustic transmission of composite elastic plate. J. H. Huang (Graduate Program of Electro-Acoustics and Ph. D. Program in Mechanical and Aeronautical Engineering, Feng Chia University, jhhuang@fcu.edu.tw), and Yu-Ting Tsai (Ph. D. Program in Mechanical and Aeronautical Engineering, Feng Chia University)

This paper investigates the phenomenon of sound transmission loss of the muti-layer composite elastic plate which includes the different thickness and material in each isotropic laminate. Unlike the classic plate concept, this paper considers that the sound waves in a fluid medium are converted into elastic waves in the plate and then converted back into sound to the fluid medium on the other side. According to thick plate theory, the two numerical methods are proposed to solve the muti-layer dynamic stiffness matrix by using transfer matrix approach and joint matrix approach. Two experiments are explored in this paper, one is to change the number of laminate of composite plate to compare the computational efficiency and differences for both numerical analysis; the other is by changing the material properties of each laminate of composite plate to explore the lossless level of sound transmission, sound reflection and sound absorption of different

composite elastic plate. Through the result and discussion, this paper provides the useful acoustic transmission assessment concept of sound absorption sandwich panel for designing the mechanical noise/vibration suppression and architectural acoustics isolation.

11:40

2aSA7. Research on engine exhaust noise control based on the neural network. Ye Wang, Xueguang Liu, Changchun Yin (School of Energy and Power Engineering; Harbin Engineering University, Harbin, Heilongjiang, xiaoyezi3152008@163.com), and Min Zhu (Aviation Industry Corporation of China, Shenyang, Liaoning)

A new type of semi-active control method which called bypass duct silencer is proposed. For the sake of reducing the exhaust noise by 15 dB, the relationship among the engine speed, the exhaust temperature and the structure of the bypass duct silencer is got. Then the bypass duct silencer which can change the internal structure of the silencer is present. The silencer contains several valves, which could control the flowing direction of sound wave aiming to neutralize sound wave in the downstream intersection of the duct. When the BP neural network structure was trained, the engine speed and the exhaust temperature measured were considered as input signals, meanwhile, the controlling situation of the valves was considered as output signal. The network structure is certain when the error of this method lies within a tolerable range the thresholds and weights are determined. The trained neural network structure could be used to reduce the exhaust noise in practical application.

12:00

2aSA8. Research on hybrid isolator technique. Xueguang Liu, Ye Wang (Research Institute of Power Engineering Technology, Harbin Engineering University, Harbin, Xueguang_liu@hotmail.com), and Bin Zhang (Shanghai Space Propulsion technology Research Institute, China)

This paper presents a new hybrid isolator technique that composes of passive vibration device and active actuator. It contains the advantages of both active and passive isolation system. During the optimization design, the passive vibration isolator as cylindrical rubber is designed, then the active actuator is designed as electromagnetic actuator with advantages of compact construction and larger forces. After manufacturing the prototype designed, the vibration active control experiment is carried out, which can verify the hybrid isolator's performance. During the mono-layer active control experiment, vibration of the upper layer mass reduced about 29dB~40dB. In the two-stage experiment, it has gained a good effect in both single and double frequency excitation. It means that the hybrid isolator technique designed has good performance of anti-vibration in practical application.

12:20

2aSA9. Paraseismic vibrations—disadvantages and advantages in application in a new system for spatial orientation of blind people. Jerzy Wiciak (AGH- University of Science and Technology, Faculty of Mechanical Engineering and Robotics, Department of Mechanics and Vibroacoustics, wiciak@agh.edu.pl)

Since 2009 the project System For Determination Of Hazardous Areas For The Blind People Using Wave-Vibration Markers has been in constant development. Its first stage was the public opinion survey. The aim of the survey was to analyze and evaluate problems associated with movement of the blind and partially sighted people through an urban environment. This article presents results of research on possible application of paraseismic vibrations and problems with paraseismic vibrations in the system for spatial orientation of blind and partially-sighted people which is built. Particular attention was paid to methodology of signal acquisition of paraseismic vibrations for the purpose of education of spatial orientation of people with vision dysfunction. Finally the frequency bands of vibrations generated by means of traffic and transport in various places and situations in the city are defined. These frequencies allow better designing of the system that support movement of blind people in urban conditions.

TUESDAY MORNING, 15 MAY 2012 S428, 9:40 A.M. TO 12:20 P.M.

Session 2aSC

Speech Communication: Speech Perception Across Languages, Modalities, and Levels of Ability (Poster Session)

Puisan Wong, Chair
pswresearch@gmail.com

Contributed Papers

All posters will be on display from 9:40 a.m. to 12:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:40 a.m. to 11:10 a.m. and contributors of even-numbered papers will be at their posters from 11:10 a.m. to 12:20 p.m.

2aSC1. Canadian raising and the perception of consonant voicing. Rebekka Puderbaugh and Terrance M. Nearey (Univ of Alberta, 4-32 Assiniboia Hall, Edmonton, AB, Canada T6G 2E7, puderbau@ualberta.ca)

Studies have shown that monophthongal VC sequences with higher F1 and shorter durations result in more [-voice] consonant responses than those with lower F1 and longer durations [Moreton, E. 2004 J. Phon. 32(1), 1, Nearey, T. M. 1997, J. Acoust. Soc. Amer., 101(6), 3241]. However, certain diphthongs in varieties of English with Canadian Raising show the opposite pattern, namely that before C[+voice], /aɪ/ diphthongs will have higher F1 in the nucleus than before C[-voice]. This study will investigate the interaction of interpretations of vowel tokens as monophthongs or diphthongs on the perception of the voicing of the following consonant. Measurements will be made from the speech of 10-20 speakers of Canadian English from the Edmonton area to be used in the resynthesis of hVd and hVt tokens. This will involve construction of a multi-dimensional continuum varying F1 values of the vowel nucleus as well as vowel duration and trajectory from nucleus to offglide. The resulting continuum will span both the four phonetic vowel categories [^, a, ^ɪ, aɪ] (where [^ɪ] and [aɪ] are viewed as allophones of a single phoneme /aɪ), and also the final consonants /t/ and /d/.

2aSC2. Preceding non-linguistic stimuli affect categorisation of Swedish plosives. Johannes Bjerva, Ellen Marklund, Johan Engdahl, Lisa Tengstrand, and Francisco Lacerda (Department of linguistics, Stockholm University, SE – 106 91 Stockholm, bjerva@ling.su.se)

Speech perception is highly context-dependent. Sounds preceding speech stimuli affect how listeners categorise the stimuli, regardless of whether the context consists of speech or non-speech. This effect is acoustically contrastive; a preceding context with high-frequency acoustic energy tends to skew categorisation towards speech sounds possessing lower-frequency acoustic energy and vice versa (Mann, 1980; Holt, Lotto, Kluender, 2000; Holt, 2005). Partially replicating Holt's study from 2005, the present study investigates the effect of non-linguistic contexts in different frequency bands on speech categorisation. Adult participants (n=15) were exposed to Swedish syllables from a speech continuum ranging from /da/ to /ga/ varying in the onset frequencies of the second and third formants in equal steps. Contexts preceding the speech stimuli consisted of sequences of sine tones distributed in different frequency bands: high, mid and low. Participants were asked to categorise the syllables as /da/ or /ga/. As hypothesised, high frequency contexts shift the category boundary towards /da/, while lower frequency contexts shift the boundary towards /ga/, compared to the mid frequency context.

2aSC3. Does a stop bias exist in infant consonant manner-of-articulation perception? Young-Ja Nam (McGill University # 205 School of Communication Sciences and Disorders Beatty Hall 1266 Pine Avenue West Montreal, QC H3G 1A8, Canada, young.nam@mal.mcgill.ca)

Phoneme inventories are biased favoring stop over fricative consonants. A similar bias is evident in acquisition. For example, an asymmetrical pattern was observed when infant word learning was assessed using the switch task with stop-initial and fricative-initial minimal pair CVC nonsense syllables (Altvater-Mackensen & Fikkert, 2010). In this task, Dutch-learning fourteen-month-olds noticed a fricative to stop change but failed to detect a stop to fricative change. These findings were interpreted in terms of phonological representations emerging in early lexical development. In this study, we tested English and French infants aged 4-5 months to determine whether they show a perceptual bias favoring stop manner. We presented CVC nonsense syllables - /bas/ and /vas/- in a preference task using the look-to-listen procedure. The /b-v/ contrast is phonemic in English and French. Infants listened significantly longer to /bas/ than to /vas/ trials (p = .004). This perceptual preference cannot be explained in terms of phonological representations in young infants who are not yet producing stops or fricatives and have almost no receptive vocabulary. We will discuss this phonetic bias in light of adult data showing similar perceptual asymmetries and consider the implications for the development of infant speech processing and early word learning.

2aSC4. Voice onset time as a cue for perceiving place of articulation in stop consonants. Lan Shuai (Department of Chinese, Translation and Linguistics; City University of Hong Kong, susan.shuai@gmail.com), and Tao Gong (Department of Linguistics; University of Hong Kong)

It is well-established that the formant transitions are the cues to differentiate the perceived place of articulation (POA) of a stop consonant, regardless of voice onset time (VOT). However, as shown in the acoustic analysis of utterances, it is also documented that stop consonants with different POAs have distinguished VOTs, in English and other languages. Moreover, various models have been proposed to explain the covariation between POA and VOT. Given that there is a correlation between these two, it is possible that VOT also serves as a cue for perceiving POA in addition to formant transitions, and that POA affects the judgment of VOT in addition to temporal cues. Previous research addressed the role of POA in distinguishing voicing contrasts, but the effect of VOT on perceiving POA has not been reported in previous literature. By varying the VOTs of bilabial, alveolar, and velar stops, the current study shows that VOT is also an important cue in distinguishing POA. The results are not consistent with a one-to-one mapping between specific phonetic cues and articulatory gestures, but support a statistical learning of multiple phonetic features in phonemes.

2aSC5. A cross-linguistic study of the effect of perceived gender on the categorization of children's vowels. Benjamin Munson (University of Minnesota, 115 Shevlin Hall, Minneapolis, MN 55455, munso005@umn.edu)

Previous research has shown that listeners classify acoustically gender-ambiguous vowels differently depending on whether they believe the speaker to be a man or a woman (Johnson, Strand, & D'Imperio, 1999). In real-world listening situations, this tendency could be especially pronounced in the perception of vowels produced by children, as children's voices are inherently gender-ambiguous (Perry, Ohde, & Ashmead, 2001). Given Johnson et al.'s findings, we would predict that adults would categorize children's vowels differently depending on whether they believe the speaker to bes male or female. To examine this hypothesis, a set of experiments was conducted in which native speakers of Cantonese, Japanese, and English categorized a series of synthetic vowels (Menard et al., 2006). These were generated using an articulatory synthesizer, and were meant to represent the vocal tracts of newborn children, 2-, 4-, 5-, 10-, 16-, and 21-year olds. Adults categorized these vowels and provided judgments of the perceived gender and age of the speakers. Preliminary results suggest that speakers of English categorize vowels differently depending on whether they judge the child to be male or female. This tendency is especially marked for the most gender-ambiguous stimulus set, those based on the 10-year-old vocal tract.

2aSC6. Neural representations of vowels and vowel-like sounds. Laurel H. Carney (University of Rochester, Rochester, NY, laurel.carney@rochester.edu), and Joyce M. McDonough (University of Rochester, Rochester, NY)

Neural representations of speech at every level of the neuraxis have nonlinear features that are not described by spectrograms or linear filter-banks. In this study, recent computational models for populations of cells in the auditory periphery, brainstem, and midbrain were used to explore the implications of vowel features for neural responses. Peripheral neural responses are characterized by strong periodicities, dominant frequencies, that depend upon the distribution of energy across the harmonics and vary across vowels [Fant, 1970]. Strong periodicities related to the fundamental and low-frequency harmonics are observed for peripheral neurons tuned to a wide range of frequencies [Delgutte & Kiang, 1984]. The computational model captures this feature of the physiological responses. These periodicities are interesting because many midbrain neurons are tuned to fluctuations in this frequency range. Single-formant vowels allowed systematic manipulations of the relationship between formant and harmonic frequencies. Larger envelope fluctuations occur when harmonic and formant frequencies are mismatched than when they are aligned. The auditory models suggest that these differences in fluctuation amplitude are significant for responses of higher–order auditory neurons that are tuned to fluctuation rate. The long-term goal is to understand the interrelationship of vowel space and neural responses.
Support: NIH-NIDCD-R01-001641(LHC) & NSF-0853929(JMM)

2aSC7. The influence of amplitude modulation depth on perceived roughness of vowels. Rahul Shrivastav (115 Oyer Hall, Michigan State University, East Lansing, MI, 48824, rahul@msu.edu), Lisa Kopf (Dauer Hall, University of Florida, Gainesville, FL 32611), and David Eddins (4202 E. Fowler Ave., University of South Florida, Tampa, FL 33620)

Previous research has shown that roughness perception for vowels is influenced by the waveform amplitude modulation. It was observed that sinusoidal modulation (constant modulation depth of 100%) between 20 Hz and 70 Hz has the greatest impact on perceived roughness of vowels [Shrivastav & Eddins, 2011; JASA, 129, 2661]. The present experiment examined the effects of modulation depth on the perception of roughness for two of 10 vowels used by Shrivastav & Eddins (2011). Synthetic copies of two vowel stimuli selected from the Sataloff/Heman-Ackah database were generated using a Klatt synthesizer. Sinusoidal amplitude modulation was superimposed on each vowel using three modulation frequencies (20 Hz, 30 Hz, and 45 Hz) and five modulation depths (0, -5, -10, -15, and -20 dB). Listeners judged the perceived roughness of each of the 30 speech stimuli (2 talkers x 3 modulation frequencies X 5 modulation depths) by matching it to

a comparison sound. The comparison sound consisted of a sawtooth wave, mixed with speech-shaped noise, and amplitude modulated with a raised (power of 4) cosine wave. Results will show how depth of amplitude modulation affects perceived roughness and will help to develop predictive models to quantify roughness. Research funded by NIH.

2aSC8. Stimulus-dependent modulation of vocalization-induced cortical activation. Zhaocong Chen, Peng Liu, Weifeng Li, Qiang Lin, and Hanjun Liu (Department of Rehabilitation Medicine, The First Affiliated Hospital, Sun Yat-sen University, Guangzhou 510080, lawrence.sums@gmail.com)

It has been well documented that sensory responses to self-produced speech/vocalization are suppressed when compared to the sounds that are produced externally. Some recent studies, however, reported enhancement effect for active vocalization relative to passive listening. The present study was to address whether speaking-induced cortical activation can be modulated by the physical features of the stimulus. Subjects sustained a vowel phonation and heard either their pitch-shifted voice (100 cents, or 500 cents) or a sum of their vocalization and pure tone or white noise during mid-utterance. During passive listening, subjects remained silent and listened to the playback of what they heard during active vocalization. Compared with passive listening, the results showed enhanced P2 responses for 100 cents condition whereas suppression effect for tone or noise condition. 500 cents condition elicited nothing but suppressed effect for N1 response. These findings suggest a stimulus-dependent modulation of vocalization-induced cortical activation, leading to enhancement or suppression effect relative to the play back of the vocalization. The results are discussed in relation to differential mechanisms underlying online mornitoring of auditory feedback at utterance onset and during mid-utternance.

2aSC9. Relative role of pitch vs. phonation cues in White Hmong tone identification. Marc Garellek (Dept. Ling., UCLA, Los Angeles CA 90095-1543, marcgarellek@ucla.edu), Christina Esposito (Dept. Ling., Macalester College), Patricia Keating (Dept. Ling., UCLA), and Jody Kreiman (Dept. Head & Neck Surg., UCLA)

This study investigates the relative importance of phonation and pitch cues in (White) Hmong tone identification. Hmong has seven productive tones, two of which involve non-modal phonation. The breathy tone is usually produced with a mid- or high-falling pitch contour similar to the high-falling modal tone. Similarly, aside from some pitch differences between the low modal tone and the low-falling creaky or checked tone, production studies have shown that the phonation differences between the two tones are large. Fifteen native listeners participated in two perception tasks, in which they were asked to indentify the word they heard. In the first task, participants heard natural stimuli with manipulated F0 and duration (phonation unchanged). Results indicate that the phonation of the stimulus is important in identifying the breathy tone, but not the creaky one. Duration and F0 were more closely tied to creaky tonal identification than phonation. In the second task, source spectrum components were manipulated to create stimuli ranging from modal to breathy sounding, with the F0 held constant. The results of this task indicate that changes in H1-H2 and H2-H4 are independently important for distinguishing breathy from modal phonation when F0 is held constant. [Work supported by NSF and NIH]

2aSC10. Cue-trading of tonal perception in Hai'an Mandarin. Sunjing Ji (University of Arizona, Linguistics Department, Douglass Building Rm 200E, Tucson 85719, sunjing@email.arizona.edu)

It has been known that the perception of a phonological category is sensitive to cue trading among multiple phonetic parameters (Liberman, 1996; Francis et. al., 2008; Holt and Lotto, 2006). Following Abramson and Erickson (1992), this paper further explores the question of whether the perception of tonal categories in a Chinese dialect, Hai'an Mandarin, is sensitive to the trade-off between the VOT (voice onset time) of the onset of a monosyllabic word and fundamental frequencies (F0) of the rhyme of the word. The interactions between VOT and F0 for the bootstrapping of the tonal categories are discussed in the context of several perceptual experiments. It is suggested that such psychological basis of tone perception is in tune with one tonogenesis theory, which says that tonal contrasts are generated as compensation for the loss of voicing distinction of the onset consonants (Hombert et al., 1979).

2aSC11. Statistical analysis on the possibility of distinguishing Chinese homonyms by the contrast of their tones. Song Liu, Qi Sun, Kazuko Sunaoka, and Shizuo Hiki (Language and Speech Science Laboratory, Waseda University, 1-104, Totsuka-machi, Shinjuku-ku, Tokyo 169-8050, Japan, liusong@aoni.waseda.jp)

The present authors have been analyzing statistically the nature of the Chinese tones (Hiki et al., Proc. Int'l Symposium on Tonal Aspects of Languages, 2004, Beijing, 73-74). The letter database used is *the Grammatical Knowledge-base of Contemporary Chinese*, S-W. Yu, editor, Tsinghua University Press, China, 1998. In this report, the use of contrast of tones for distinguishing words within a group of homonyms is assessed, from the viewpoint of the amount of information transmitted. Among the bisyllabic words having different simplified Chinese character notation, more than 10% of the words belongs to the homonymous groups having the same Pinyin notation. However, the number of words having the same Pinyin notation and combination of tones is not so many, that about 80% of them can be distinguished by the contrast of their tones. Detailed analysis showed some characteristic distribution regarding the type of tones in the contrast, depending on whether the number of words in a homonymous group is two or more, or whether the position of syllable of the contrastive tones in bisyllabic words is first, second or both. This result is useful for guiding the beginners' Chinese tone learning.

2aSC12. Speech intelligibility in military noise for normal-hearing and hearing-impaired listeners using level-dependent tactical hearing protectors. Christian Giguère, Chantal Laroche, and Véronique Vaillancourt (Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H 8M5, Canada, cgiguere@uottawa.ca)

The effects of two tactical headsets with level-dependent hearing protection capabilities on face-to-face speech intelligibility in military noise were investigated. Devices were the Nacre QuietPro and the Peltor Powercom Plus. Noises recorded from light-artillery vehicles in the Canadian Forces were reproduced in a simulation room at 80-95 dBA. Over 45 subjects covering a wide range of hearing profiles were tested using sentences from the Hearing-In-Noise Test. When used as passive devices with the electronics powered off, the devices performed as expected from conventional protectors having the same amount of attenuation. In this mode, there were large performance differences among subject groups in terms of the effects of wearing the devices compared to unprotected listening. When used in active talk-through (or surround) mode, both devices showed large intelligibility benefits over the passive mode and demonstrated a level of performance often exceeding that in unprotected listening. The subject group with the most impaired hearing benefitted the most from the active mode. The findings indicate that the current technology of high-end tactical headsets could provide substantial benefits in situational awareness during noisy military operations for a wide range of hearing profiles. [Work based on Defence R&D Canada Contractor Report DRDC Toronto CR-2011-101.]

2aSC13. The effect of training structure on perceptual learning of accented speech. Christina Y. Tzeng and Lynne C. Nygaard (Emory University, christina.tzeng@gmail.com)

Although previous research suggests that high variability training facilitates perceptual learning of systematic variation in speech, the extent to which the organization of training materials affects this learning remains relatively unexplored. The present study examined the role of training structure on the perceptual learning of speaker-independent properties of Spanish-accented speech. During training, native adult speakers of American English transcribed sentences spoken in English by four native Spanish-speaking adults. Training stimuli were presented to listeners either grouped

by speaker, sentence, or randomized with respect to sentence and speaker. At test, listeners transcribed novel sentences produced by four unfamiliar Spanish-accented speakers. Transcription performance at test was found to vary as a function of the organization of training materials. Listeners transcribed test sentences more accurately when training sentences were randomized relative to when training sentences were grouped by speaker or when listeners received no training. Test performance when training sentences were grouped by sentence was intermediate. These findings suggest that variable training structure may direct listeners' attention to accent-general properties of speech, allowing for comparison across speakers' voices and linguistic content, and generalization of learning to unfamiliar accented speakers.

2aSC14. The usefulness of the modified nonsense syllable test as a measure of speech identification. Mini Shrivastav (University of Florida, 336 Dauer Hall, Dept. of Speech, Language, and Hearing Sciences, Gainesville, FL 32611, mnshriv@ufl.edu), and David Eddins (University of South Florida, 4202 E. Fowler Ave, PCD 1017, Tampa, Florida 33620)

The NST is a standardized speech-identification test involving closed-set identification of nonsense syllables. Its organization permits speech-identification testing with a variety of consonants and vowel contexts, but its usefulness is limited by the long time it takes to administer and the limited confusion matrices that can be generated. A modified version of the NST (MNST) was developed by Gelfand et al. (1992). They reported data on young normal-hearing listeners at presentation levels ranging from 20 dB to 52 dB SPL in quiet and in noise. A previous study extended the work of Gelfand et al. (1992) to higher presentation levels more suited for older and/or hearing impaired listeners. This work indicated that the MNST is a potentially useful measure of nonsense syllable identification. A significant advantage of the MNST is that it allows detailed analyses of consonant confusions among all the test consonants while reducing the test time greatly. The present work describes the confusion matrices of the two subsets of the MNST for a group of young normal-hearing listeners. This work further validates the MNST and its usefulness as a tool for measuring speech-identification scores and generating detailed consonant confusion matrices in a relatively short amount of time. The above work was supported in part by NIH NIDCD 5R03DC10266-2 awarded to the first author.

2aSC15. The communicative influence of gesture and action during speech comprehension: gestures have the upper hand. Spencer Kelly (Colgate University, skelly@colgate.edu), Meghan Healey (National Institutes of Health), Asli Ozyurek (Max Planck Institute for Psycholinguistics), and Judith Holler (Max Planck Institute for Psycholinguistics, University of Manchester)

Hand gestures combine with speech to form a single integrated system of meaning during language comprehension (Kelly et al., 2010). However, it is unknown whether gesture is uniquely integrated with speech or is processed like any other manual action. Thirty-one participants watched videos presenting speech with gestures or manual actions on objects. The relationship between the speech and gesture/action was either complementary (e.g., "He found the answer," while producing a calculating gesture vs. actually using a calculator) or incongruent (e.g., the same sentence paired with the incongruent gesture/action of stirring with a spoon). Participants watched the video (prime) and then responded to a written word (target) that was or was not spoken in the video prime (e.g., "found" or "cut"). ERPs were taken to the primes (time-locked to the spoken verb, e.g., "found") and the written targets. For primes, there was a larger frontal N400 (semantic processing) to incongruent vs. congruent items for the gesture, but not action, condition. For targets, the P2 (phonemic processing) was smaller for target words following congruent vs. incongruent gesture, but not action, primes. These findings suggest that hand gestures are integrated with speech in a privileged fashion compared to manual actions on objects.

2aSC16. The role of context in the perception of environmental sounds. Robert Risley, Valeriy Shafiro, Stanley Sheft (Rush University Medical Center, 600 South Paulina Chicago, IL 60612, risleyrobert@gmail.com), Adam Balser (Columbia College Chicago, 600 South Michigan Ave Chicago, IL 60605), Derek Stiles (Rush University Medical Center, 600 South Paulina Chicago, IL 60612), and Brian Gygi (Veterans Affairs Northern California Health Care System, 150 Muir Road Martinez, CA 94553)

Past work has shown involvement of context in the perception of sequences of meaningful speech sounds. The present study extended investigation to meaningful environmental sounds, evaluating the accuracy in identifying individual sources in sequences of five environmental sounds that were either likely or not to have occurred together in place and time. Rating of sequence coherence confirmed the subjective distinction between sequence types. In the main task, young normal-hearing listeners were instructed to identify and order each sound they heard by selecting from among 20 randomly selected labels. Listeners were significantly more accurate in reporting both the names and order of the sounds of the coherent sequences in which sounds were more likely to have occurred together. If not scored in terms of source order, a smaller but still significant difference was obtained between coherent and incoherent sequences. For both sequence types, there was a trend for best performance for the initial and final sources of the sequences. Overall, results are consistent with speech findings and demonstrate a positive effect of context in the perceptual processing of environmental sounds. [Work supported by NIH/NIDCD.]

2aSC17. Understanding concurrent speech is not impaired by removal of spectro-temporal overlap. Piotr Kleczkowski and Marek Pluta (Department of Mechanics and Vibroacoustics, AGH University of Science and Technology, Cracow, Poland, kleczkow@agh.edu.pl)

Removal of spectro-temporal overlap is the operation performed on acoustic signals in the time-frequency domain. It can be seen as extreme energetic masking, where the masking threshold is equal to the level of the masker. In Experiment 1 the triplets of words were presented concurrently with reversed speech of four speakers. The recognition of words with and without spectro-temporal overlap was compared. In Experiment 2, understanding of five words spoken simultaneously, with and without spectro-temporal overlap was compared. Over 50 subjects took part in both experiments. The joint results showed no significant difference between unprocessed presentation and the processed one, where spectro-temporal overlap was removed. This is rather surprising, as the amount of information provided to the ear was considerably reduced in the processed case.

Session 2aSP

Signal Processing in Acoustics, Acoustical Oceanography, and Underwater Acoustics: Model-Based Processing and Analysis I

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Chair's Introduction—9:15

Invited Papers

9:20

2aSP1. Vector sound intensity measurements with a tetrahedral arrangement of microphones in a spherical probe. Thomas Søndergaard

The three dimensional sound intensity vector may be measured using a number of microphone configurations, an example of which is an orthogonal arrangement of three matched pairs of microphones. This setup is prone to unwanted reflections, however, and uses more microphones than theoretically required. Results are therefore presented on the calculation of the sound intensity vector using atetrahedral arrangement of microphones housed within a solid sphere. The acoustic presence of the sphere is quantified; specifically, frequency dependent phase and pressure measurements due to diffraction are obtained for various incidence angles and compared with theoretical values. The proposed intensity probe is further evaluated against traditional arrangements for the calculation of the acoustic intensity vector and advantages of adopting the spherically mounted, tetrahedral arrangement are highlighted.

9:40

2aSP2. A model-based acoustical signal processing method for the passive ranging of sniper gunfire. Brian G. Ferguson and Kam W. Lo (Maritime Operations Division, Defence Science and Technology Organisation, Australian Technology Park, Eveleigh 2015 NSW, Australia, Brian.Ferguson@dsto.defence.gov.au)

Acoustically, sniper gunfire is characterized by a muzzle blast caused by the discharge of a bullet from a rifle and a ballistic shock wave emitted along the trajectory of a supersonic bullet. The location of the firing point can be estimated by using a small microphone array to measure the differences in the angles- and times-of-arrival of the muzzle blast and shock wave. Traditionally, the bullet is assumed to travel at constant speed along its trajectory which, in practice, can lead to significantly biased range estimates. This problem is solved by invoking a physics-based model to describe the deceleration of the bullet along its trajectory. The ballistic model parameters are the initial (or muzzle) velocity and ballistic constant of the bullet, which are assumed to be known a priori. The performances of the traditional and model-based methods for ranging the point of fire are evaluated using 2,500 rounds of 5.56 and 7.62 mm ammunition fired at ranges of 75, 175, 275, 375 and 475 m under two (low and high) wind speed regimes. It is found that the localization errors of the model-based method are smaller by an order of magnitude when compared with those of the traditional method.

10:00

2aSP3. Correction errors in sound barrier measurement caused by temperature and wind variance. Xun Wang and Michael Vorlaender (Institute of Technical Acoustics, RWTH Aachen University, Neustrasse 50, 52066, Aachen, Germany, xun.wang@akustik.rwth-aachen.de)

The impulse response and its associated transfer function are the most important properties of linear and time invariant acoustic systems. Under low signal-to-noise conditions, e.g. measuring the sound barrier outdoors, the time-frequency windowing can be performed to reduce the noise at the time-frequency blocks where the noise does not overlap with the excitation signal, and at the blocks where the noise overlap with the excitation signal, the averaging methods have to be implemented. However, the averaging methods have to be performed in time-invariant systems, and in time-variant systems, averaging methods may lead to unpredictable errors. In sound barrier measurement, the time variance usually results from the temperature shift and the wind action. As for the temperature shift, the impulse response varies with a time-stretching process. And the wind action causes the phase and magnitude shift of the impulse response. Both the time-stretching factor and phase-shifting factor can be estimated by maximizing the cross correlation function between the measured impulse responses. Finally, the averaging can be performed after modifying the temperature- and wind-dependent impulse responses to constant-temperature and non-wind impulse responses. In this presentation, the stability of this time-stretching and phase-shifting average method will also be discussed.

10:20

2aSP4. Two kinds of timbre representations and its application into acoustic target classification. Kean Chen, Yong Liang, Liangfen Du, Jue Chen, Ying Wu, and Huanrong Wang (Organisation: Department of Environmental Engineering, School of Marine Engineering, Northwestern Polytechnical University, China; Postal Address: No. 58 mailbox in Northwestern Polytechnical University, Xi'an (710072), Shaanxi, China, kachen@nwpu.edu.cn)

Being an important auditory attribute of sound, timbre exhibits great potential for classifying sound source and its suitable representation and parameterization are crucial for feature extraction. In this study, we express environmental sound's timbre in terms of verbal description and its projection in independent space, which are respectively referred to as Natural Timbre (NT) and Essential Timbre (ET). In this study, such two kinds of timbre expressions are applied to acoustic target recognition using synthesized steady-state underwater noise with two subjective rating experiments. First a semantic differential test is conducted and the NT-based target identification rates are achieved by forced clustering; then a paired comparison experiment is carried out to get the ET-based identification rates. Finally, the recognition performances for two kinds of timbre representations are compared and its advantages are discussed in association with feature extraction and acoustic target recognition.

10:40–11:00 Break

Contributed Papers

11:00

2aSP5. Passive sonar target localization and tracking using sequential bayesian filter in uncertain sea environment. Hangfang Zhao, Xianyi Gong, and Zibin Yu (Hangzhou Applied Acoustics Research Institute, No. 96, Huaxing Road, Xihu District, Hangzhou City 310012, Zhejiang Province, China, sklzhaohf@gmail.com)

A problem of localizing and tracking an acoustic source is researched when ocean environment including water column sound speed profile, ocean depth and seabed property is uncertain. In a Bayesian framework, the source and environmental parameters are regard as random variables with known prior knowledge, then the prior knowledge of parameters and acoustic model are combined with a likelihood function of data to provide posterior probability density functions (PDF) of both south and environmental parameters, target location parameters are estimated by marginalization integrates over the environmental parameters finally. In other hand, the environmental parameters and target location parameters evolve in time or space, which can be described by state-space model. Information on these parameters evolution and uncertainty at preceding steps can be incorporated to determine future probability of parameters with acoustic data being available at current step. A framework of a sequential Bayesian filter is derived naturally based on the model. A Kalman filter or particle filter could be used to implement the sequential Bayesian filter depend on the linear or nonlinear of the measurement equation or/and the state equation. The sequential Bayesian filter is demonstrated to be able to localize and track a source broadcasting a broadband signal in shallow water using both simulated and real data acquired by a towed array.

11:20

2aSP6. Research on underwater acoustic moving target 3-D imaging using spatial scattering model. Dezhu Liu, Jie Feng, Zhaoli Li, and Feng Zeng (The third research institute of china electronics technology group corporation, Beijing 100015, China, liudezhu_2008@163.com)

A spatial scattering model was proposed to analyze underwater acoustic moving target signals received by 3-D imaging sonar, and moving target signals for 3-D imaging sonar was also simulated. Analysis showed the validity of the spatial scattering model. Based on the spatial scattering model and simulated signals, FFT beam forming method was applied and performance of 3-D imaging sonar was analyzed. Result showed that the 3-D imaging performance effected by underwater acoustic moving target's size, shape and distance etc, emphasized the influence of target moving especially. So the spatial scattering model is both a useful theoretical tool and a simulation method aiding further research and analysis of underwater acoustic moving target 3-D imaging.

11:40

2aSP7. High-resolution Sonar based on carrier-free narrow pulse: I. transmission characteristic. Yunlu Ni and Hang Chen (School of Marine Technology, Northwestern Polytechnical University, Xi'an 710072, China, niyunlu@mail.nwpu.edu.cn)

In comparison with a sinusoidal carrier in the conventional sonar system, currently realizable carrier-free millisecond or microsecond pulse for novel sonar system has advantages: attaining more target information, restraining fluctuation of reverberation envelop efficiently in short-range detection and achieving accurate estimation. Since the attenuation increases rapidly with frequency in variable ocean, waveform distortion of such narrow pulse with transient wide band has occurred and can not be negligible. In this paper the focus will be on the transmission characteristic of carrier-free narrow pulse in viscid ocean by Finite Element Method. Employing received pulse at certain distance in numerical modeling and physical measurement, a special filter as ocean modeling is exactly designed for carrier-free narrow pulse, which is convenience to collect the pulse wave at arbitrary distance for posterior proceeding.

12:00

2aSP8. The harmonic distortion level measurement system of the loudspeaker using two adaptive filters. Toyota Fujioka, Yoshifumi Nagata, and Masato Abe (Iwate University, toy@cis.iwate-u.ac.jp)

The harmonic distortion level is an important criterion to evaluate loudspeaker performance. The harmonic distortion level is generally obtained using the power spectrum of the audio signal from a loudspeaker, and a microphone is used to receive the audio signal. It is important to measure the accurate harmonic distortion level for evaluating performance of the loudspeaker. Generally, The harmonic distortion level is measured by the spectrum analysis. However, the spectrum analysis requires large computational complexity. Therefore, we proposed the new technique to measure the harmonic distortion level of the loudspeaker by using the adaptive filter. And we evaluated the performance of the proposed technique by experiment. The proposed technique can measure the accurate level the same as the spectrum analysis, but requires long measurement time to converge the filter coefficient. This paper presents a description of the new technique to measure the harmonic distortion of the loudspeaker by using two adaptive filters. The propose technique improves the accuracy and the measurement time of the proposed measurement technique by using the adaptive filter. We show some results of real-time experiments. The results of the real-time experiments verified that the newly proposed technique improves measurement time and accuracy of measured level.

12:20

2aSP9. Quasi real time monitoring system for the tomographic reconstruction of the vortex wind field. Haiyue Li, Takuya Hirasawa, and Akira Yamada (Tokyo Univ. of A&T, Koganei, Tokyo 184-8588, Japan, hyli@cc.tuat.ac.jp)

Quasi real time vortex wind field monitoring system for the tomographic reconstruction of the vortex wind velocity field was developed. The system was implemented by installing the multichannel sound transmitter and receiver pairs on both sides of the monitoring area. Sound wave travel time data were measured between the arbitrary combination of the facing transmitters and receivers. To speed up the multichannel data collection time, transmission and reception was made in parallel by sending the coded modulation signals. The demodulation correlation calculation, for the extraction of the desired data from the multiple transmission signals, was made in high speed by using GPU (Graphics Processing Unit). Test examination, using the four channel indoor system, showed the promise to complete the vortex wind velocity image production about every 1 s.

Session 2aUW

Underwater Acoustics and Acoustical Oceanography: Underwater Acoustics in Asian Marginal Seas: Field Experiments and Modeling I

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Contributed Papers

9:20

2aUW1. Inversion of temperature vertical structure by ocean acoustic tomography data in the Luzon Strait. Ju Lin (a. College of Information Science and Technology, Ocean University of China, 238 Song-Ling Rd., Laoshan Dist., Qingdao 266100, China, b. Graduate School of Engineering, Hiroshima University, 1-4-1 Kagamiyama, Higashi-Hiroshima 739-8527, Japan, julin97@gmail.com), Araka Kaneko, Naokazu Taniguchi (Graduate School of Engineering, Hiroshima University, 1-4-1 Kagamiyama, Higashi-Hiroshima 739-8527, Japan), Huan Wang (College of Information Science and Technology, Ocean University of China, 238 Song-Ling Rd., Laoshan Dist., Qingdao 266100, China), and Noriaki Gohda (Graduate School of Engineering, Hiroshima University, 1-4-1 Kagamiyama, Higashi-Hiroshima 739-8527, Japan)

Luzon Strait is the key channel which connects the western Pacific and the South China Sea(SCS). It is very important to monitor the exchanges of materials and energy between the western Pacific and the SCS, which are caused by the Kuroshio and eddy's variability. The Kuroshio can intrude into the SCS through the Luzon Strait in various manners and significantly affects the oceanic variability of the SCS. A deep-sea acoustic tomography experiment was conducted at the northern part of the Luzon Strait during April 2008 to October 2008. Three 800Hz acoustic systems were deployed at depth about 800m on the subsurface mooring line, and the distance between the systems was about 40km. The four-month reciprocal acoustic transmission data between a pair systems were successfully obtained. The three groups of acoustic arrival peak were distinctly separated. The temperature vertical structure along the transmission line are inversed by using the Matched-Mode Processing method. The inversion results are in good agreement with the temperature field observations and HYCOM results. The 17-day and 23-day period signals are found in the inversion result, which correspond to the frequency of occurrences of cold eddy in the Luzon Strait. (Work supported by NBRPC 2007CB411803, NSFC 41176033 and ONR)

9:40

2aUW2. High frequency ocean acoustic tomography observation at coastal estuary areas. Yu Zhang, Zongxi Zhao, Dongsheng Chen, and Wuyi Yang (Key Laboratory of Underwater Acoustic Communication and Marine Information Technology of the Ministry of Education, Xiamen University, Xiamen, China, yuzhang@xmu.edu.cn)

Ocean acoustic tomography (OAT) technique can obtain oceanographic information and has been received a lot of interest. High frequency OAT (in few kHz range) can be used for small and confined areas such as estuaries and bays with complicated hydrological conditions. In this study, we investigate the application of the high-frequency reciprocal transmission OAT to assess the temperature and current in Xiamen sea area using computer simulations and sea experiments. Based on the temperature data obtained from remote sensing, high frequency OAT is employed to reconstruct two-dimensional temperature, sound speed, and current field of the 1.2 km $\times$ 1.2 km region. The results show that increasing the number of acoustic stations decreases the travel-time errors of high frequency OAT; however, excessively increasing stations can not significantly improve the inversion accuracy. Furthermore, this method has also been examined by a sea experiment on monitoring the current and water temperature of Wuyuan Bay. High frequency OAT might provide an effective method on temperature and current observation at coastal estuary areas. (This work was financially supported by the National Science Foundation of China (Grant No. 11174240) and the Fundamental Research Funds for the Central Universities (2011121010))

10:00

2aUW3. Geoacoustic inversion using low and mid frequency bottom reflected signals in shallow water off the east coast of Korea. Jee Woong Choi, Changil Lee (Department of Environmental Marine Sciences, Hanyang University, Ansan, Korea, choijw@hanyang.ac.kr), Sungho Cho, Donhyug Kang (Korea Ocean Research and Development Institute, Ansan, Korea), and Joung-Soo Park (Agency for Defense Development, Changwon, Korea)

Two types of short-range propagation experiments were conducted in shallow water (nominal water depth of 150 m) off the east coast of Korea, using 6 to 10 kHz CW signals and low-frequency broadband bulb implosion as acoustic sources. The received signals were recorded on the vertical line arrays at ranges shorter than 500 m. A marine geological observation conducted at the experimental site showed that there was a thin surficial sediment layer with thickness of less than 1 m overlaying the thicker and higher speed sediment layer and the basement was 15-20 m under the water-seabed interface. Bottom reflection loss as a function of grazing angle and frequency were estimated from the single bottom-interacting path of CW signals, which were used for the inversion of geoacoustic parameters for the surficial sediment structure. The geoacoustic inversion for parameters corresponding to the lower interface was performed using the bulb implosion data. The arrival time difference and the amplitude ratio between the single bottom-reflected and sub-bottom-reflected signals were used to estimate the sound speed and attenuation coefficient, respectively, within the second layer. [Supported by ADD (Agency for Defense Development, Korea), and KORDI (Korea Ocean Research and Development Institute)]

10:20

2aUW4. Study of ocean ambient noise characteristics based on vector signal processing of acoustic energy flow. Jialiang Li (Chinese Academy of Sciences, No. 8, Shangqing Road, Shibei District, Qingdao 266023, Shandong Province, China, lijiaqingdao001@163.com), Jianheng Lin, and Xuejuan Yi

With the development of technology, vector sensors are more and more applied in underwater acoustics measurements. As one of the signal processing methods, vector signal processing method based on acoustic energy flow is gradually developed. Methods based on acoustic energy flow overcome

some inherent shortcomings of traditional pressure signal processing methods and increase processing gain and DOA estimation accuracy. Methods based on acoustic energy flow have been verified to be practically useful by trials. The existing vector sensor frequency measurement model is modified in this paper. The anisotropy and other characteristics of ocean ambient noise are studied in the paper based on vector signal processing of acoustic energy flow, and the simulation results are reasonable.

10:40–11:00 Break

11:00

2aUW5. Uncertainty quantification and sensitivity analysis of transmission loss in the sea area northeast of Taiwan. Yuan-Ying Chang, Chi-Fang Chen (National Taiwan University, No. 1, Sec. 4, Roosevelt Road, Taipei 10617, Taiwan, d94525011@ntu.edu.tw), Yung-Sheng Chiu, and Ruey-Chang Wei (National Sun Yat-sen University, No. 70, Lienhai Rd., Kaohsiung 80424, Taiwan)

Uncertainty of transmission loss, resulted from the uncertainty of geophysical and physical oceanographic parameters (features), could consequently contribute to the uncertainty of sonar performance prediction. This research adopts coupled ocean-acoustic modeling and compares with data of field observations over the continental shelf and slope close to North Men-Hua Canyon offshore northeastern Taiwan in order to study the uncertainty of transmission loss. This area contains range-varying sediment type, rapid-changing bathymetry, and complex water column activities brought by the Kuroshio intrusion, which jointly induce a highly uncertain environment for sound propagation. Mid- frequency sound propagation in this area is observed and modeled to interpret the propagation effect of the water column and seabed. Finally, the uncertainty of ocean model output and transmission loss in this area is quantified and the sensitivity is analyzed. The spatial and temporal effects on the ocean and acoustic field are also quantified in this study. [This research is sponsored by national science council. Project number: NSC 98-2623-E-002-013-D]

11:20

2aUW6. A numerical study of temporal variation of low-frequency sound observed in the obs measurement off the east coast of Taiwan. Chen-Fen Huang and Shih-Chieh Lin (Institute of Oceanography, National Taiwan University, Taipei 10617, Taiwan, chenfen@ntu.edu.tw)

The ambient noise recorded by the OBS system deployed off the east coast of Taiwan in the TAIGER (TAiwan Integrated GEodynamics Research) project was employed to obtain the Noise Cross-correlation Functions (NCF). The data were recorded between the stations located at Yaeyama Ridge (water depth about 4000 m), starting from May of 2008 for a period of more one year. The results of NCF have shown that there exists strong microseism energy in the frequency band between 0.16 Hz and 0.35 Hz all year around, and has also demonstrated a large temporal variations as much as 5 seconds. A series of numerical simulations using OASES was conducted to investigate the effect of various waveguide propagation conditions on the temporal variation of NCF. The analysis may potentially pave the way of applying passive acoustic tomography for the monitoring of ocean climate.

11:40

2aUW7. Underwater acoustic measurements and simulations from air-gun array of R/V Marcus G. Langseth in TAIGER experiment in the west coast of Taiwan. Jeff C. H. Wu (Department of Engineering Science and Ocean Engineering, National Taiwan University, Taipei 106, Taiwan, R.O.C, d98525001@ntu.edu.tw), Linus Y.S. Chiu (Institute of Applied Marine Physics and Undersea Technology, National Sun Yat-Sen University, Kaohsiung 804, Taiwan, R.O.C), Chi-Fang Chen (Department of Engineering Science and Ocean Engineering, National Taiwan University, Taipei 106, Taiwan, R.O.C), and Ruey-Chang Wei (Institute of Applied Marine Physics and Undersea Technology, National Sun Yat-Sen University, Kaohsiung 804, Taiwan, R.O.C)

The high-level noise produced by air-gun array of R/V Marcus G. Langseth was highly possible to impact on marine mammals. However, the west coast of Taiwan is the habitat of many marine mammals, so the environmental impact of the air-gun array noise is extensively concerned. The source level and beam-pattern of the air-gun array have been estimated by the previous study in the east coast of Taiwan. According to the known source level and beam-pattern, sound pressure level can be calculated and compared with the measured data from the air-gun array in the west coast of Taiwan. This study may be taken as a reference to understand the sound propagation and impacts on marine mammals in the west coast of Taiwan. (Sponsored by National Science Council of Republic of China under project "Noise Monitor of TAiwan Integrated GEodynamics Research (TAIGER) Experiment – Middle Section of Taiwan West Coast" No. NSC98-2119-M-110-002)

12:00

2aUW8. Estimation and analysis of the underwater construction noise of the offshore wind farm in the west coast of Taiwan. Henry H. J. Tsai (Department of Engineering Science and Ocean Engineering, National Taiwan University, hungjutsai@gmail.com), Sheng Fong Lin (Industrial Technology Research Institute), Chi-Fang Chen, and Jeff C. H. Wu (Department of Engineering Science and Ocean Engineering, National Taiwan University)

Wind-generated electricity has been one of the green energy in the world. In Taiwan, there is enormous potential for wind energy, especially in the west coast. However, there are many marine mammals in this area, so we cannot neglect the environmental problem due to the construction noise of the offshore wind farm. This research is to estimate and analyze the underwater construction noise of the wind farm. According to the data of hydrographic, bathymetric and sediment, the spreading and impact range of the construction noise arising from the offshore wind farm can be estimated and simulated. The result of this study may be taken as a reference of the construction of the offshore wind farm.

12:20

2aUW9. Acoustic mode coupling resulting from an internal solitary wave approaching a shelf break in South China Sea. Linus Y. S. Chiu (National Sun Yat-sen University, No. 70, Lienhai Rd., Kaohsiung 80424, Taiwan, linus@mail.nsysu.edu.tw), Yuan-Ying Chang, Chi-Fang Chen (National Taiwan University, No. 1, Sec. 4, Roosevelt Road, Taipei 10617, Taiwan), D. Benjamin Reeder, Ching-Sang Chiu (Naval Postgraduate School, 833 Dyer Rd, Monterey, CA), Ying-Tsong Lin, and James F. Lynch (Woods Hole Oceanographic Institution, 266 Woods Hole Road, Woods Hole, MA 02543)

An internal solitary wave (ISW) encountering the shelf break, making the waveguide is compressed, can cause different joint coupling effect for acoustic modes. In this paper, the extended criterion for adiabatic invariance is developed by parameterizing the joint mode coupling effect of an ISW encountering the shelf break and is used for sensitivity studies considering various internal wave amplitudes and slope angles of shelf break, to examine the acoustic coupling effect resulting from both bathymetry and ISW. The modeling results are accurately predicted by the extended criterion of adiabatic invariance and compared to experimental observations from ASIAEX and NLIWI of the effect of acoustic waveguide being compressed by the shelf break and ISW. Results demonstrate that the coupling of acoustic energy to higher modes as the waveguide is compressed when the ISW encounters the shelf break. And as amplitude of the ISW and the incline of the sloping bottom increase, coupling strength for both adjacent- and non-adjacent modes is enhanced. [This research is sponsored by national science council.]