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 △ 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 △ 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 $\pm$ 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 noninvasive 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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Fenghua Li, Cochair
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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.]

Session 2pAAa

Architectural Acoustics and Psychological and Physiological Acoustics: Psychoacoustics in Rooms I

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Bernhard Seeber, Cochair
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Chair's Introduction—1:55

Invited Papers

2:00

2pAAa1. Estimation of speech privacy performance from acoustic parmeters in two adjacent rooms. Hayato Sato, Masayuki Morimoto (Environmental Acoustics Laboratory, Department of Architecture, Graduate School of Engineering, Kobe University, Rokko, Nada, Kobe 657-8501, Japan, hayato@kobe-u.ac.jp), Yasushi Hoshino (Environmental Acoustics Laboratory, Department of Architecture, Graduate School of Engineering, Kobe University, Rokko, Nada, Kobe 657-8501, Japan/Nippon Sheet Glass Environment Amenity Co., Ltd., Takanawa, Minato, Tokyo 108-0074, Japan), and Yasuhiko Odagawa (Environmental Acoustics Laboratory, Department of Architecture, Graduate School of Engineering, Kobe University, Rokko, Nada, Kobe 657-8501, Japan)

Sato et al. suggested the equal-intelligibility contours that enable us to predict sound insulation performance and background noise level required to achieve a certain level of speech privacy/security on the basis of the word intelligibility test [Sato et al., Appl. Acoust. 73, 43-49 (2012)]. However, temporal aspects of sound fields, which would affect intelligibility scores, were not considered in the intelligibility test. In the present study, intelligibility tests were performed to investigated the effects of temporal aspects of sound fields on the scores. The possibility of using reverberation time as one of the predictors of speech privacy performance will be discussed on the basis of intelligibility tests.

2:20

2pAAa2. Making speech announcements intelligible in public spaces from a speech production view. Nao Hodoshima (Department of Information Media Technology, Tokai University, 2-3-23 Takanawa Minato-ku, Tokyo 108-8619, Japan, hodoshima@tokai-u.jp), and Takayuki Arai (Department of Information and Communication Sciences, Sophia University, 7-1 Kioi-cho, Chiyoda-ku, Tokyo, 102-8554, Japan)

Speech announcements in public spaces are often hard to hear due to noise and reverberation, and this is especially true for elderly people compared to young people. This study aims to improve speech intelligibility in noisy/reverberant environments based on the way we change how we speak depending on an acoustic environment. Speech uttered in a noisy environment (noise-induced speech) is generally more intelligible for young people than speech produced in a quiet environment when both types of speech sounds are heard in noise (i.e. Lombard effect). This paper examines whether reverberation- as well as noise-induced speech are more intelligible to young and elderly people. The results of our listening tests showed that elderly listeners had significantly higher word identification scores for noise/reverberation-induced speech than speech spoken in a quiet environment. The results also showed that noise/reverberation-induced speech was more intelligible to the listeners than speech in quiet with background noise/reverberation conditions that were not only identical but also different to those used during the recording of speech. The results suggest that using noise/reverberation-induced speech for public address systems makes speech announcements more intelligible for elderly people in public spaces. [Work supported by KAKENHI (21700203) and Sophia University Open Research Center.]

2:40

2pAAa3. Modeling binaural speech intelligibility in spatial listening conditions. Thomas Brand (Medical Physics, University of Oldenburg, Oldenburg, Germany, thomas.brand@uni-oldenburg.de), Jan Rennies (Fraunhofer IDMT Hearing, Speech and Audio Technology, Oldenburg, Germany), Rainer Beutelmann (Animal Physiology & Behaviour, University of Oldenburg, Oldenburg, Germany), Anna Warzybok, and Birger Kollmeier (Medical Physics, University of Oldenburg, Oldenburg, Germany)

Speech intelligibility is substantially improved when speech and interfering noise are spatially separated. This spatial unmasking is mostly caused by a combination of head shadow and binaural auditory processing. Binaural speech reception thresholds (SRTs) in such spatial conditions can be predicted very accurately using a combination of an Equalization-and-Cancellation (EC) model and the Speech-Intelligibility-Index (SII). This binaural speech intelligibility model predicts effects including levels, frequency spectra, and directions of the speech and noise signals as well as listeners' hearing loss, early reflections and reverberant parts of the noise signals. Earlier versions of the model were only able to predict the intelligibility of near-field speech. Recent extensions can also predict the intelligibility of far-field speech by taking early reflections and reverberant parts of the speech signal into account. However, some

interactions between the direction of the noise source and early speech reflections cannot be predicted yet. The overall high prediction accuracy of the model (more than 90% of the data's variance can be explained) indicates that the model is applicable in real rooms and may serve as a tool in room acoustical design. This work was supported by the Deutsche Forschungsgemeinschaft (SFB TRR 31).

3:00

2pAAa4. A binaural model for predicting speech intelligibility in rooms using noise and reverberation suppression processes. Vanessa Li, Ning Xiang, and Jonas Braasch (Graduate Program in Architectural Acoustics, School of Architecture, Rensselaer Polytechnic Institute, Troy, New York, vanessa.li@gmail.com)

Conventional metrics for predicting speech intelligibility are commonly described using the speech intelligibility index (SII) and speech transmission index (STI)–both of which are calculated using a monophonic signal. Under binaural conditions, these predictors often underestimate intelligibility due to the fact that beneficial binaural cues for unmasking detrimental effects are not accounted for. Beutelmann et al. [J. Acoust. Soc. Am. 127, 2479-2497 (2010)] proposed a binaural speech intelligibility model that determines the SII after suppressing spatially-separate noise using the equalization-cancellation (EC) process. This research expands on previous work by extending the model to analyze auditory scenes with more complex room acoustics, namely reverberation. In order to account for speech intelligibility degradation in the presence of room effects, the SII calculation is replaced by the STI. The current work also incorporates binaural release from reverberation by introducing an additional reverberation suppression mechanism into the model based on interaural coherence. The use of speech intelligibility metrics as an objective form of measurement may be used as a means to further understand binaural suppression processes within various room acoustic configurations through subjective listening tests.

3:20

2pAAa5. Predictions of spatial release from masking from architectural plans. Sam Jelfs (Philips Research Europe, High Tech Campus 36 (WO-p.076), 5656 AE Eindhoven, The Netherlands, sam.jelfs@philips.com), and John Culling (School of Psychology, Cardiff University, Tower Building, Park Place, Cardiff, CF10 3AT, U.K.)

2p TUE. PM

Spatial separation of target speech and interfering noise produces spatial release from masking (SRM). A model of SRM for speech in noise gives accurate predictions for multiple noise sources and in reverberation. The model is based on the frequency-weighted combination of better ear listening and binaural unmasking. Here we use this model to explore the effects of room acoustics, seating choice, listener head orientation and table layout in a virtual restaurant simulation. The modeled restaurant contains nine tables for two in three rows, and each diner listens to their partner across the table. Substantial SRM was predicted in all seating locations, but was lowest at the centre table. Orienting the head away from the target voice by 20-30deg improved SRM for most seating positions, but this benefit was small for those seated at the edges of the restaurant and facing in. Reorienting these tables so that they have a wall at one side improved average SRM. Acoustic treatments applied to the walls produced larger benefits in SRM than treatment of the ceiling that achieved equivalent reverberation time. Reverberation reduced both SRM and its variability across seating locations and head orientations.

3:40

2pAAa6. Sub-cortical envelope and fine structure cues: the interaction of age and individual differences for normal-hearing adults in complex environments. Dorea Ruggles (Boston University & University of Minnesota, druggles@umn.edu), Hari Bharadwaj, and Barbara Shinn-Cunningham (Boston University)

Attending to a stream of speech amid competing speech in the presence of reverberation is an everyday task that many adults with normal hearing thresholds take for granted. Recently, though, we've shown that this ability varies widely for individual young and middle-age listeners and that the variability is not related to listener age. In this follow-up study, we recruited additional middle-aged listeners whose data reveal important differences between young and middle-age normal hearing adults and their use of fine structure and envelope cues in directed spatial attention. Twenty-two listeners ranging in age from 20 to 55 years completed spatial selective attention and frequency modulation (FM) detection tasks and had passive frequency following responses (FFRs) to a monotonized /dah/ syllable recorded with scalp electrodes. Although spatial selective attention ability was unrelated to age, older listeners were more impaired by reverberation than younger listeners. The FFR data was analyzed with a method that separates the contributions of fine structure and envelope phase locking, and results depict an age-related transition in envelope and fine structure relationships with complex listening.

4:00–4:20 Break

4:20

2pAAa7. Performance of binaural technology for auditory selective attention. Janina Fels, Bruno Masiero, Josefa Oberem (RWTH Aachen University, Institute of Technical Acoustics, D-52056 Aachen, Germany, Janina.Fels@akustik.rwth-aachen.de), Vera Lawo, and Iring Koch (RWTH Aachen University, Institute of Psychology, D-52056 Aachen, Germany)

A room-acoustic situation with many sources is one of the best examples for auditory selective attention – relevant information should be selectively observed and irrelevant information should be ignored. In a joint research project between Acoustics and Psychology at RWTH Aachen University the intentional switching of auditory selective attention is examined using dichotic and binaural presentation of the stimuli. The goal is to provide artificially generated acoustic scenes (e.g. typical classroom situation, open plan offices etc.), as in psychoacoustic experiments on auditory selective attention no differences between a real situation and an artificially generated situation occur. Therefore at first we investigate various binaural reproduction and equalization methods using experiments in auditory selective attention. Headphones must always be adequately equalized if they are to deliver high perceptual plausibility. However, the transfer function between headphones and ear drums varies between different persons. Because of this, individual equalizations with different microphone positions in the ear canal are measured. In listening tests the overall quality of the equalization methods is to be rated regarding localization and realism, envelopment as well as immersion. First results concerning the psychoacoustic experiments, the scene generation as well as headphone equalization will be presented.

4:40

2pAAa8. Perceptual metrics in elementary classrooms and their correlations to student achievement. Lily M. Wang and Lauren M. Ronsse (Durham School of Arch. Engr. and Constr., Univ. of Nebraska–Lincoln, Peter Kiewit Institute, 1110 S. 67th St., Omaha, NE 68182-0816, LWang4@UNL.edu)

Binaural impulse response measurements have been made at multiple locations within 20 unoccupied elementary school classrooms in a public school district in Nebraska, USA. Assorted objective metrics have been calculated from these binaural impulse responses, including speech transmission index, distortion of frequency-smoothed magnitude, interaural cross-correlations, and interaural level differences. This presentation highlights the results of these measurements within each classroom and between classrooms. These metrics have been correlated to student scores on standardized achievement tests, obtained as averages for each classroom. One interesting finding is that the distortion of frequency-smoothed magnitude was found to be significantly related to student achievement scores in the language subject areas, even though classroom reverberation times were not, due to the limited range of reverberation times across this investigation.

5:00

2pAAa9. Perceptual compensation for effects of reverberation in isolated test-words. Anthony Watkins and Andrew Raimond (Reading University, Reading RG66AL, UK, syswatkn@rdg.ac.uk)

Reverberation from room reflections tends to degrade speech reception, as happens when listeners are required to identify test words from a “sir”-to-“stir” continuum. When the reverberation is substantial it introduces a “tail” from the [s], which tends to fill the gap that cues the [t]. A degradation effect arises as listeners report correspondingly fewer “stir” sounds. However, a recent report indicates that in certain conditions this degradation is entirely absent, despite substantial reverberation. These conditions are where test words are played in isolation, and where the reverberation is kept at the same level in the test-words of every trial that listeners hear. Conditions used here are generally similar to this except that the level of reverberation in test words is varied unpredictably from trial to trial, with the substantial-level trials intermingled with trials where the level of reverberation is much lower. Under these conditions the degradation effect is restored. This suggests a perceptual compensation from information within test words that can build up over sequences of trials, but only when the test word’s reverberation stays the same from trial to trial. Other conditions confirm that reverberation information from the vowel part of isolated test-words affects identification of preceding consonants.

5:20

2pAAa10. Speech intelligibility improves with listening exposure in reverberant rooms. Pavel Zahorik, Eugene Brandewie, and Nirmal Kumar Srinivasan (Department of Psychological and Brain Sciences, University of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu)

Emerging evidence suggests that speech perception in a reverberant room can be altered by recent listening exposure to the room. This result is interesting and important because it suggests that perceptual aspects related to room acoustics are not constant as a function of listening time, and it may help to better understand why hearing-impaired listeners often report difficulty with speech understanding in reverberation. Virtual auditory space techniques have been a key component of the research on this effect, since they allow both realistic simulation of reverberant room listening environments, and a level of stimulus control that would be impossible for real-room listening. Here, recent work demonstrating objective improvements in speech intelligibility with room exposure is summarized, with particular focus on details of the effect including its time course and its sensitivity to different speech materials. [Work supported by the NIH/NIDCD.]

5:40

2pAAa11. Effect of the inter-aural sound level differences on the speech intelligibility. Chan Jae Park and Chan Hoon Haan (Chungbuk National University, 361-763, Cheongju, Korea, cjpark@chungbuk.ac.kr)

It was defined from the results of many previous researches that the most effective criterion affecting sound clarity in rooms is the early reflected sound. C80 and D50 were used to evaluate sound clarity in rooms. These parameters are related with the sound energy in function of time and they are monoral indexes which do not count for the spatial information including sound directions. Thus, C80 and D50 do not consider the changes of sound clarity caused by difference of sound energy between left and right ears. The proposed study investigates the effect of the inter-aural sound level difference (ILD) on the speech intelligibility in classrooms which can be occurred by the absorption of interior surfaces. In order to do this, sound levels were measured with inter-aural differences of D50, C80 (ID-D50, ID-C80) using dummy-head binaural recording systems. As a result, it was clearly found that ILD was much increased beyond JND after sound absorptions were implied in room. Also, the correlation coefficient of ILD with the distance from sidewalls has increased from 0.696 to 0.890 after sound absorptions. However, there was not clear difference in other parameters (ID-D50, ID-C80). It is also denoted that the increase of ILD can affect the subjective speech intelligibility as well.

Session 2pAAb

Architectural Acoustics and Noise: Multifamily Dwellings and Lightweight Structures (Lecture/Poster Session)

Angelo Campanella, Cochair
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Jeffrey Mahn, Cochair
jeffrey.mahn@canterbury.ac.nz

Chair's Introduction—1:55

Invited Papers

2:00

2pAAb1. Revisions to the EN12354 prediction method of calculating the flanking sound reduction index of lightweight building elements. Jeffrey Mahn and John Pearse (University of Canterbury, Christchurch, New Zealand, jeffrey.mahn@canterbury.ac.nz)

There is great interest worldwide in applying the standard, EN12354 to predict the flanking sound reduction index to lightweight building elements. However, there are several problems which must be overcome before the prediction method can be accurately applied to lightweight building elements. One problem is the prediction of the resonant component of the sound reduction index of the elements under investigation. As part of the work of COST Action FP0702, several methods of calculating the resonant component have been proposed and evaluated. The evaluation was conducted by comparing the predicted flanking sound reduction indices which were calculated using the different methods of calculating the resonant sound reduction index to measured values for a series of elements. The elements included single, homogeneous elements and double leaf elements. This paper presents the details of that evaluation. A correction factor based on the radiation efficiencies of the elements which was proposed by CSTB is recommended.

2:20

2pAAb2. Coping with uncertainties in the design and evaluation of acoustical assemblies. John LoVerde (Veneklasen Associates, 1711 16th St, Santa Monica, CA 90404, jloverde@veneklasen.com), and Wayland Dong (Veneklasen Associates, 1711 16th St, Santa Monica, CA 90404)

The statistical uncertainty in acoustical testing has been insufficiently studied in the acoustical community, and the effects of the uncertainties have been largely ignored. Most acousticians do not know what the reproducibility limit is for acoustical test results from accredited laboratories, but tacitly assume that the limits are on the order 1 or 2 rating points. In recent years the authors have demonstrated that the uncertainties in acoustical laboratory testing in the United States are much higher than most people have realized [LoVerde and Dong, J. Acoust. Soc. Am. 125, 2629 (2009), J. Acoust. Soc. Am. 126, 2171 (2009) J. Acoust. Soc. Am. 130, 2355 (2011)]. The typical acoustician's reaction is that it should be possible to measure to higher precision with suitable changes to the test methodology, and improving the precision of laboratory testing is an important goal that deserves much more attention than it has received. However, there is no guarantee that it will be practical to substantially decrease the uncertainties in acoustical testing, and regardless, the uncertainties characterizing the existing body of research cannot be changed. Accepting this fact demands some changes to how acoustical consultants design assemblies, evaluate products, and interact with clients and regulatory agencies.

2:40

2pAAb3. A ten year evaluation of the sound insulation of a volume based lightweight construction system. Rikard Öqvist (Tyréns AB, Västra Norrlandsgatan 10B, 903 27 Umeå, Sweden, rikard.oqvist@tyrens.se)

Since the middle of the '90s, Lindbäcks Bygg, a company in the North of Sweden has developed a lightweight timber construction system with industrially prefabricated room volumes. During these years, a lot of acoustic measurements have been performed. This information constitutes a valuable body of knowledge for future development of the system. However, the measurements have been performed by many different actors and are thus documented in many different ways, which means that it is difficult to get an overview. The aim of this project is to categorise all measurements of impact and airborne sound insulation that have been made on the volume system during the last ten years and to acoustically evaluate the significant constructional changes that have been implemented. A method to document previous and future measurements and link them to constructional parameters will be presented.

3:00

2pAAb4. Effect of floating floor and raised floor on floor impact sound insulation of wooden construction. Atsuo Hiramitsu (Building Research Institute 1, Tachihara, Tsukuba-City, Ibaraki, 305-0802, Japan, hiramitu@kenken.go.jp)

The Act on the Promotion Wood for Public building was enforced in October, 2010 in Japan. According to this act, a public building in a low layer assumes a wooden construction as a rule or positively uses wood. Moreover, it is expected that the building such as apartment houses comes to be made from wooden. However, the floor impact sound insulation performance of the wooden constructions is low compared with that of the concrete constructions. This paper presents the effect of floating floor (dry double system floor) and raised floor (free access floor) on floor impact sound insulation performance in wooden construction. A reference floor constructed with the wood-frame construction was installed in the floor opening of the reverberation chamber, and the floating floors or the raised floor were built on it. Then, the reductions of transmitted floor impact sound level of them were measured. The results showed that the density of the surface material influences the floor impact sound interception performance in the case of the floating floor. In addition, the reductions of transmitted floor impact sound level in wooden construction were compared with them in concrete construction in the case of the raised floor.

3:20

2pAAb5. Rating the impact sound insulation of flooring from its airborne sound reduction index. George Dodd (University of Auckland, Dept. of Architecture, Private Bag 92019, Auckland 1142, New Zealand, g.dodd@auckland.ac.nz)

With the increasing adoption of higher density forms of dwellings in urban environments we need reliable but economic means of verifying that they meet specified sound insulation requirements. This is particularly so now that territorial and building code authorities are realising that performance checking is an indispensible part of the quality assurance process. This presentation concerns a simplified way of confirming the impact noise insulation in buildings. This makes impact sound insulation rating possible without the need to make impact sound pressure level measurements. The technique estimates the impact sound level by using an accelerometer attached to a hammer (similar to those used in the standard tapping machine) to measure the reaction force from the floor and combines this with the floor's sound reduction index. The method simplifies the equipment required for impact sound insulation rating and allows measurements to be made in the presence of high background noise.

Contributed Papers

3:40

2pAAb6. A study on performance of ventilated soundproof windows with fans. Shih Pin Huang (Department of Architecture, Kao Yuan University, No. 1821, Jhongshan Rd., Lujhu Dist, Kaohsiung City 82151, Taiwan, R.O.C., t60055@cc.kyu.edu.tw), and Rong Ping Lai (Department of Architecture, National Cheng Kung University, No. 1, University Road, Tainan City 701, Taiwan, R.O.C.)

The research discusses sound insulation on some designed types of ventilated soundproof windows which combine with fans and the air lead-in boards. Conclusions are as the following, The result shows that the type of left-inlet and right-outlet could reach 30.4 dB(A) of sound insulation and the type of double-sided duct could reach 30.7 dB(A) of sound insulation. Changing glass thickness of window, 5mm and 10mm, the sound insulation of ventilated soundproof windows show that frequency response are almost the same by comparing three different types. It shows that has no improvement of sound insulation by adding the glass thickness. Discussing the influence of indoor environment due to fan noise. For the type of normal ventilation, the noise is 41.2 dB(A) when the fan set at the middle speed with air lead-in boards. For the type of double-sided duct, the noise is 33.7 dB(A) when the fan set at the middle speed with air lead-in boards. For the type of left-inlet and right-outlet, the noise is 36.4 dB(A) when the fan set at the middle speed with air lead-in boards. Acknowledgment of support to National Science Council, NSC 100-2221-E-244 -019

4:00–4:20 Break

4:20

2pAAb7. Noise reduction of a double-skin façade considering opening for natural ventilation. Jean-Philippe Migneron and André Potvin (Groupe de recherche en ambiances physiques, École d'architecture, Université Laval, 1 côte de la Fabrique, Quebec City QC G1R 3V6, Canada, jean-philippe.migneron.1@ulaval.ca)

The growing interest in natural or hybrid ventilation systems brings a challenge for good integration of openings in building façades. With a noisy environment, there is an important limitation for the use of direct openings in common building envelopes. As a part of a research project dedicated to this problem, it is possible to evaluate the impact of several double-skin configurations, modifying openings, space between façades or the choice of construction assemblies. Experimental measurements made in laboratory conditions lead to the estimation of usual noise reduction and sound transmission class. Moreover, the airflow at constant differential pressure was assessed as functions of the aperture and compared to sound insulation. Analyzing those parameters together give useful information for the design of passive ventilation with a significant airflow when acoustical performance is an important issue.

4:40

2pAAb8. Effects of upper surface layers on the vibration characteristics of floating floor systems in concrete slab structures. Jae Ho Kim and Jin Yong Jeon (Hanyang University, Seongdong-gu, Seoul 133-791, Korea, nosaer4@gmail.com)

The effect of materials with supporting conditions on the vibration characteristics of the floating floor systems have been studied in concrete slab structures. Total 10 types of floor systems which have various sizes of panels and supporting beams with different joints were made based on actual conditions. In the measurement of vibration, ISO rubber ball was used as an impact source in order to reproduce human walking. The vibration characteristics were evaluated through calculating the vibration dose value (VDV) and autocorrelation function (ACF) parameters for the vibrations of the floor surface layers. Finally, a human walking experiment was conducted to investigate the subjective responses to the effect of vibration characteristics of the floating floors. As results, the correlation coefficients between physical parameters and subjective responses were derived and a perception model was obtained.

5:00

2pAAb9. Experimental study of the sound transmission loss in normal incidence through autoclaved aerated concrete material. Delas Olivier (Vipac Engineers & Scientists (HK) Ltd, 9A Wah Kit Commercial Centre, 300-302 Des Voeux Road Central, Sheung Wan, Hong Kong SAR, olivierd@vipac.com.au)

Sound transmission loss through autoclaved aerated concrete has been recently studied with the help of a computer model that uses the matrix transfer method and represents the material as a general poroelastic layer

using Biot theory together with the Johnson-Champoux-Allard model for visco-thermal dissipations. It was found that when the autoclaved aerated concrete material is covered on both sides by plaster daubs, the computed sound transmission loss decreases by 4 dB in the low frequency range (50-400 Hz). One possible explanation of this phenomenon might be the material losing most of its porous characteristics when the pores located at its surface are obstructed. In this research work, the effect of the surface pore obstruction is experimentally studied by measuring the sound transmission loss of the light weight concrete in an impedance tube for different thicknesses and finishes of the material: plain, plaster daubed and painted. Sound transmission loss values are measured in an impedance tube by the 4-microphone method. This article presents the normal incidence sound transmission loss measured for the different configurations of the autoclaved aerated concrete material. The experimental results are compared with the corresponding computed values and conclusions on the effect surface pore obstruction on the sound transmission loss of the material are drawn. Recommendations are provided for the future design of high sound transmission loss wall systems integrating autoclaved aerated concrete materials.

5:20

2pAAb10. The effect of receiving room sound field on the impact ball sound pressure level. Jeong Jeong Ho (Fire Insurers Laboratories of Korea, jhjeong@kfpa.or.kr)

Field measurement method of heavy-weight impact sound pressure level using impact ball have been used in Korea, Japan and Canada. Also this field measurement method is discussing in ISO. Impact force and subjective responses of impact ball are very similar with child's jumping and running. In Korea and Japan is considering that using impact ball as standard impact source instead of bang machine. It is reported that heavy-weight impact sound pressure level using bang machine was varied by the sound field condition of receiving room such as sound absorption power and room volume. In this study, it is checked that impact ball sound pressure level also affected by the receiving sound field condition. Impact ball sound pressure level was measured vertically connected reverberation chamber and sound absorption power was changed by polyester sound absorption blanket with air space and glass wool. The reverberation time at 1 kHz band was changed from 10 s to 0.2 s by sound absorption material. Impact ball sound pressure level measured without sound absorption material was 58 dB in Li, Fmax, AW, but the level was 46 dB with sound absorption treatment. From this result, it is confirmed that sound field correction term may be needed in the heavy-weight impact sound pressure level measurement method using impact ball.

The following abstracts will be presented in poster format. The posters will be on display and the authors will be at their posters from 5:40 p.m. to 6:20 p.m.

2pAAb11. Stud effect of plasterboard partition on sound transmission. Sungchan Lee, Jinyun Chung, and Jungbin Im (Daewoo E&C, sungchan.lee@daewooenc.com)

The plasterboard partition is composed of plasterboard, stud, runner and insulation; each component influences sound insulation performance of partitions. In this study, the sound insulation of plasterboard partitions varied in stud thickness and spacing was investigated. The partition was composed of two layers of fire-proof 12.5 mm plasterboard each side, single frame of 50 mm stud and 24 kg/m^3 50 mm glass wool. The stud spacing was varied from 450 to 900 mm and the thickness of stud was 0.5 and 0.8 mm. The results showed that the sound insulation of plasterboard partitions was improved by increasing stud spacing and decreasing stud thickness.

2pAAb12. Reflection and transmission properties of a wall-floor building element: comparison between finite element model and experimental data. Juan Negreira Montero and Delphine Bard (Lund University Box 118, 221 00 Lund, Juan.Negreira_Montero@construction.lth.se)

Changes in the Swedish construction code introduced in 1994 enabled the construction of wooden multi-storey buildings. The main issue in those is disturbing vibrations and noise propagating throughout the construction. Therefore, gaining knowledge about their behavior is of crucial importance for the industry. In this study, a mockup of a wall/floor junction was investigated by comparing both experimental and simulation (FEM) results. The mockup resembles a section of a real wooden building. It is 9.3 m long and 3.6 m wide. The structure was built using wooden beams as load-bearing components and chipboards as the floor surface. Likewise, a gypsum wall was placed in the middle surrounded by a wooden frame. The reflection and transmission properties of the structure were studied when subjected to harmonic excitations. The junction has been studied experimentally using dual-axis accelerometers attached to the T junction, post-processing the data using the scattering matrix formulation, which allows separating the transmitted and reflected wave as the wave propagates towards the junction, as well as the rate of wave conversion. Subsequently, a FE model of the structure was created allowing the comparation between both cases. This project was funded by Interreg IV, Silent Spaces.

Session 2pBA

Biomedical Acoustics and Physical Acoustics: Subharmonic Contrast Imaging

Michel Versluis, Cochair
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Jeffrey Ketterling, Cochair
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Invited Papers

2:00

2pBA1. Subharmonics in bubble oscillations: history, physics, applications. Andrea Prosperetti (Johns Hopkins University, 223 Latrobe Hall, Baltimore, MD 21218, prosperetti@jhu.edu)

The paper will start with a brief review of the history of subharmonic emssions from bubbles driven into oscillation by a sound afield. The peculiar nature of subharmonic oscillations as opposed to other manifestations of non-linearity will then be explained and the effects of damping will be illustrated. A few recent applications of subharmonic oscillations of bubbles will then be reviewed.

2:20

2pBA2. Modeling subharmonic response from contrast microbubbles for imaging and noninvasive pressure estimation. Kausik Sarkar (Mechanical and Aerospace Engineering, George Washington University, 801 22nd St NW, Washington, DC 20052, sarkar@udel.edu), Amit Katiyar (Mechanical Engineering, University of Delaware, 130 Academy Street, Newark, DE 19716), Jeffrey A. Ketterling, and Parag V. Chitnis (Lizzi Center for Biomedical Engineering, Riverside Research, New York, NY 10038)

In order to characterize the behaviors of encapsulated contrast microbubbles, an approach will be described where progressively more sophisticated models have been developed guided by experimental observations. The development and characterization process includes independent estimation and validation components — attenuation is used to estimate the model material parameters, and then the estimated model is validated against independently measured subharmonic response. The subharmonic aided noninvasive pressure estimation that depends on experimentally observed decrease of subharmonic response of many commercial contrast agents with local hydrostatic pressure will also be critically examined. It will be shown that the basic bubble dynamics predicts either a decrease or an increase of subharmonic response with pressure increase depending on the excitation frequency and the bubble size. This finding indicates a lack of proper understanding of the underlying process. Finally, the minimum threshold excitation for subharmonic generation from an encapsulated microbubble will be revisited to show that in contrast to the classical perturbative result, it is not always obtained at twice the resonance frequency; instead it can occur over a range of frequency from resonance to twice the resonance frequency. The quantitative variation of the threshold with different models of encapsulation will be discussed. [support: NSF, NIH, DOD]

2:40

2pBA3. Optical and acoustical characterization of the subharmonic response of single UCA microbubbles. Michel Versluis (University of Twente, P.O. Box 217, 7500 AE Enschede, The Netherlands, m.versluis@utwente.nl)

Coated microbubbles, unlike tissue, are able to scatter sound subharmonically. Therefore, the subharmonic behavior of coated microbubbles can be used to enhance the contrast in ultrasound contrast imaging. Theoretically, a threshold amplitude of the driving pressure can be calculated above which subharmonic oscillations of microbubbles are initiated. Interestingly, earlier experimental studies on coated microbubbles demonstrated that the threshold for these bubbles is much lower than predicted by the traditional linear viscoelastic shell models. Here we present an optical and acoustical study on the subharmonic response of individual microbubbles, e.g. the radial subharmonic response of the microbubbles was recorded with the Brandaris ultra high-speed camera as a function of both the amplitude and the frequency of the driving pulse. Threshold pressures for subharmonic generation as low as 5 kPa were found near a driving frequency equal to twice the resonance frequency of the bubble. An explanation for this low threshold pressure is provided by the shell buckling model proposed by Marmottant et al. It is shown that the change in the elasticity of the bubble shell as a function of bubble radius enhances the subharmonic behavior of the microbubbles.

3:00

2pBA4. Subharmonic scattering of phospholipid-shell microbubbles as a function of hydrostatic pressure. Peter Frinking, Emmanuel Gaud, Gilles Casqueiro, and Marcel Arditi (Bracco Suiss SA, 31 route de la Galaise, CH-1226 Plan-les-Ouates/GE, Switzerland, peter.frinking@bracco.com)

Subharmonic scattering of phospholipid-shell microbubbles excited at very low acoustic pressure amplitudes has been associated with echo responses from compression-only bubbles. These bubbles are near a tension-free buckling state at rest and have initial surface tension values close to zero. In this work, subharmonic scattering of phospholipid microbubbles was investigated as a function of the

initial surface tension, which was controlled by changing the hydrostatic pressure through the application of an ambient overpressure. Echo responses from a dilution of an experimental contrast agent were measured as a function of ambient overpressure ranging between 0 to 140 mmHg. The microbubbles were excited using a 64-cycle Tukey-windowed transmit burst with a center frequency of 4 MHz and peak-negative pressure of 50 kPa. Echo-power spectra were calculated, and the subharmonic response was determined for each overpressure value; the subharmonic amplitude increased by 20 dB after applying 140 mmHg overpressure (for which the initial surface tension was assumed to be near zero). In this study, an increase in subharmonic amplitude, instead of a decrease as reported by others [Shi et al., UMB 1999], was measured as a function of ambient overpressure. This observation may be exploited in a new method for noninvasive pressure measurement.

3:20

2pBA5. Subharmonic imaging for vasa vasorum. Telli Faez and Nico de Jong (Biomedical Engineering Thoraxcenter, Erasmus Medical Center, Rotterdam, The Netherlands, t.faez@erasmusmc.nl)

It is known that vasa vasorum plays an important role in atherosclerotic plaque pathogenesis and stability. Recent advances in contrast-enhanced ultrasound have shown that this technique can be used to characterize the carotid vasa vasorum and intra-plaque angiogenesis. Ultrasound propagating through tissue is nonlinear and contains higher harmonics of the transmitted wave, but it does not contain energy at the subharmonic frequency, which revives a strong interest in subharmonic emissions (backscattered energy at half the transmit frequency) from contrast agents. Subharmonic imaging (SHI) has potentially a larger contrast to tissue ratio compared to other imaging methods and has already been used in clinical experimental studies. In this study, the subharmonic scattering of phospholipid-coated contrast agents in the frequency range preferred for carotid imaging (5-15 MHz) is investigated optically and acoustically and in vitro and in vivo. The results of the measurements indicate that: -The subharmonic scattering of the microbubbles is sufficiently detectable (−10 dB below the fundamental) at 10 MHz at low acoustic pressures of 100 kPa. -The subharmonic response of microbubbles can be dynamically manipulated using a 2.5 kHz pressure wave. In conclusion, SHI has a great potential to be exploited for carotid imaging.

3:40

2pBA6. A mechanistic investigation of subharmonic response from polymer-shelled microbubbles in response to high-frequency ultrasound. Jeffrey A. Ketterling, Parag V. Chitnis, Jonathan Mamou, and Sujeethraj Koppolu (Lizzi Center for Biomedical Engineering, Riverside Research, 156 William St., NY, NY 10038, jketterling@riversideresearch.org)

Polymer-shelled ultrasound contrast agents (UCAs) can undergo a “compression only” behavior leading to shell rupture and nonlinear response of the released gas bubbles when excited below 10 MHz. This study investigated if polymer-shelled UCAs exhibited a similar behavior when excited at frequencies above 10 MHz. Four varieties of polylactide-shelled UCAs, each with a distinct shell-thickness-to-radius ratio (STRR), were employed; the STRRs were 7.5, 40, 65, and 100 nm/μm. Two experiments were performed: one examined the compression-induced rupture of UCA shells by subjecting them to static overpressure, and the other investigated subharmonic components in the backscattered signal produced by individual UCAs sonicated with 20-MHz tone bursts. The four UCAs exhibited distinctly different compression-induced rupture thresholds that were linearly related to their STRR, but were uncorrelated with UCA size. The subharmonic response of the UCAs increased with increasing STRR. Thus, the UCAs with larger STRRs were more resilient to rupture, but they produced significantly greater subharmonic activity. The results of this two-part study indicated that the polymer-shelled UCAs may not adhere to the rupture-based mechanism of subharmonic generation when excited at 20 MHz.

4:00–4:20 Break

4:20

2pBA7. High frequency subharmonic imaging: practical implementations and recent developments. Andrew Needles (VisualSonics Inc., Toronto, Canada, aneedles@visualsonics.com), Verya Daeichin, Hans Bosch, Nico de Jong, AFW van der Steen (Erasmus MC, Rotterdam, The Netherlands), and F. Stuart Foster (Sunnybrook Health Sciences Centre, Toronto, Canada)

High-frequency ultrasound systems have been developed to provide appropriate imaging resolution for small anatomical structures. Typical applications include small animal preclinical research as well as intravascular ultrasound (IVUS) in larger animals and humans. In addition to traditional B-Mode (structural) and Doppler (functional) imaging, contrast detection modes have been implemented on these systems to improve the sensitivity to microbubbles in the microcirculation. This talk will overview some examples of these implementations and their respective advantages and disadvantages. Examples include direct radio frequency (RF) filtering and pulse sequence approaches, on both single element and array based systems. Additionally, subharmonic imaging will be compared to other harmonic detection approaches (namely nonlinear fundamental and second harmonic) and cases where subharmonic only detection is optimal at high frequencies. Recent advances explore the use of the self-demodulation phenomena to enhance to contribution of the subharmonic signal from microbubbles and improve detection. Examples of in vivo data from mice and rats will be shown, illustrating the ability to detect changes in blood perfusion by analyzing contrast uptake over time with curve fitting algorithms. Finally, the detection of microbubbles targeted to endothelial cells, using subharmonic imaging in small animals, will be demonstrated.

4:40

2pBA8. A new composite particle for both contrast-enhanced ultrasound imaging and cell therapy. Qian Cheng (Institute of Acoustics, Tongji University, Shanghai 200092, China, q.cheng@tongji.edu.cn), Qing-Gang Tan (School of materials science and engineering, Tongji University, Shanghai 200092, China), Ying-Bin Liu, Song-Gang Li, Mao-Lan Li (Department of General Surgery, Xinhua Hospital, Medical School of Shanghai Jiaotong University, Shanghai 200092, China), and Meng-Lu Qian (Institute of Acoustics, Tongji University, Shanghai 200092, China)

In this paper, a new composite nano-particle for both contrast-enhanced ultrasound imaging and cell therapy is introduced. The carbon nanotubes are used for carriers due to their unique hollow structure, nano-diameter and good biocompatibility. The targeted protein, the hematoporphyrin and the gold nanoparticles are assembled on the surface of the carbon nanotubes. The targeted protein is used as

tumor localization. As a kind of sonosensitizer, the hematoporphyrin can produce the singlet oxygen while being activated by ultrasound and are studied for its effects of antitumor and apoptosis induction recently. It is worthwhile to note that the singlet oxygen has a very short lifetime and will transfer to triplet oxygen with light emission band at 1268 nm, or 634 nm and 703 nm. Triplet oxygen is the ground state of the oxygen molecule and much of them will form nano oxygen bubbles under body temperature which can be used as contrast-enhanced ultrasound imaging agent. At the same time, the gold nanoparticles which have good biocompatibility can absorb the light emission and produce the acoustic signals of some bandwidth. This work is supported by the National Natural Science Foundation of China (No. 10804085, 11174223 and 50603019), and the Shanghai Nano Special Foundation(No. 1052nm05400)

Contributed Papers

5:00

2pBA9. Size-dependent backscatter coefficient from lipid-coated monodisperse microbubbles. Yanjun Gong (Department of Mechanical Engineering, Boston University, 110 Cummington Street, Boston, MA 02215, ygong@bu.edu), Mario Cabodi (Center for Nanoscience and Nanobiotechnology, Boston University, 8 Saint Mary's Street, Boston, MA 02215), and Tyrone Porter (Department of Mechanical Engineering, Boston University, 110 Cummington Street, Boston, MA 02215)

In this study, the relationship between backscatter coefficient (BSC) and the size of Ultrasound Contrast Agents (UCAs) microbubbles has been investigated in vitro. Monodisperse lipid-coated microbubbles were produced using a flow-focusing microfluidic device. A single-element unfocused transducer with center frequency 2.25 MHz was used to measure the BSC of microbubbles in the frequency range 1-3 MHz by transmitting 1 cycle broad-band acoustic pulse with a peak-to-negative pressure 35 kPa. When compared to polydisperse microbubble with the same lipid shell composition and concentration, monodisperse microbubbles exhibited a distinct peak in the frequency-dependent BCS curve, which corresponded with the resonance frequency of the monodispersion. Furthermore, the BSC for monodisperse microbubbles was higher than that for polydisperse microbubbles around resonance frequency. This result suggests that a monodispersion driven at resonance will provide greater contrast in ultrasound images than a polydispersion, provided the concentrations are equivalent. Finally, the BSC for five monodispersions with mean diameters of 4.5, 5.1, 5.6, 6.4 and 7.4 μm with the same concentration (5000/ml) was measured and compared. The results showed that the resonance frequency was inversely related and the BSC amplitude was directly proportional to mean diameter. The implications of these results on subharmonic contrast imaging will be discussed.

5:20

2pBA10. Dependence of the subharmonic signal from ultrasound contrast microbubbles on ambient pressure. Fei Li (School of Biological Science and Medical Engineering, Beihang University, Beijing 100191, China, lifei@be.buaa.edu.cn), Tao Ling (Paul C. Lauterbur Research Center for Biomedical Imaging, Shenzhen Institutes of Advanced Technology, Chinese Academy of Sciences, Shenzhen 518055, China), Chengrui Liu (School of Biological Science and Medical Engineering, Beihang University, Beijing 100191, China), Qiaofeng Jin, Feiyan Cai (Paul C. Lauterbur Research Center for Biomedical Imaging, Shenzhen Institutes of Advanced Technology, Chinese Academy of Sciences, Shenzhen 518055, China), Deyu Li (School of Biological Science and Medical Engineering, Beihang University, Beijing 100191, China), and Hairong Zheng (Paul C. Lauterbur Research Center for Biomedical Imaging, Shenzhen Institutes of Advanced Technology, Chinese Academy of Sciences, Shenzhen 518055, China)

Ultrasound contrast agents (UCA) micro-bubbles have been well recognized as a potential noninvasive tool for blood pressure estimation. However, previous UCA indices, e.g., the shift of the resonance frequency, echo amplitude and the disappearance time, suffered from problems of low resolution, nonlinearity in the relationship with blood pressure, and only variations of the local pressure but the absolute values etc. In this paper, the effect of ambient pressure on UCA sub-harmonic optimal driving frequency (SODF) was investigated, at which the sub-harmonic scattering signals were the maximum. By applying transmit frequencies between 3MHz and 8MHz and the acoustic pressure of 300kPa, the acoustic attenuation and scattering of micro-bubbles were measured at overpressures of 4mmHg, 50mmHg and 100mmHg, comparable with the healthy human blood pressure. For groups of micro-bubbles with 80% in the diameter range of 1-2 micrometers, the shift of the SODF (SSODF) was 0.6MHz between overpressures of 4mmHg and 100mmHg, which was approximately twice of the corresponding shift of the resonance frequency, thus had an improved sensitivity of pressure estimation. The SSODF of UCA micro-bubbles may be as a novel and sensitive index of the local blood pressure estimation.

Session 2pEA

Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials II

Michael Haberman, Chair
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Invited Papers

2:00

2pEA1. Acoustic metamaterials by bending layered structures. Zixian Liang and Jensen Li (City University of Hong Kong, zixliang@cityu.edu.hk)

In this talk, basic concepts of metamaterials for acoustic imaging and wave manipulation using layered structures will be introduced. Starting from periodically layered structures, we can easily construct metamaterials with large anisotropy which is very useful to obtain sub-wavelength resolution in acoustic imaging. By geometric scaling the layered structure in a fixed direction, an acoustic hyperlens can be obtained so that the image with subwavelength details can be progressively magnified and exit the lens as far-fields. We have now further developed different geometric transformations by bending the layered structures to achieve a variety of wave manipulations including negative refraction in the effective medium regime, acoustic cloaking, and tunneling with a density-near-zero material.

2:20

2pEA2. Metamaterials for transformation acoustics applications. Steven Cummer and Bogdan Popa (Duke University, PO Box 90291, Durham, NC 27708, cummer@ee.duke.edu)

Transformation acoustics is a paradigm for the creation of sound-manipulating materials and devices that are either difficult or impossible to derive through other theoretical approaches. It is based on the idea of a coordinate transformation of an arbitrary initial sound field. If the device you imagine can be defined in terms of a coordinate transformation, by squeezing, stretching, or displacing the sound field in a finite region, then transformation acoustics provides the mathematics for deriving the properties of a material in that same finite region that will have exactly the same effect on the sound field as the coordinate transformation. Transformation acoustics theory has lead to interest in designing acoustic composites, also known as metamaterials, that can to achieve the large range of material parameters needed for transformation acoustics designs. This presentation will describe recent efforts to design and fabricate composite materials with the acoustic properties needed to realize transformation acoustics devices, and also demonstrate their performance in experimental measurements.

2:40

2pEA3. Transformation acoustics: virtual pinholes and collimators. Jun Xu (MIT, xujun@mit.edu), Yun Jing (NCSU), and Nicholas X. Fang (MIT)

In this invited talk, our preliminary study is presented on a virtual hole and a broadband acoustic collimator, by combination of the concept of complimentary media with transformational acoustics. Such effect is exemplified by a segmental defect in the original cloak, which appears as if a dipole scatterer was under the acoustic imager. A set of spatially varying effective parameter was derived from coordinate transformation. These parameters can be readily implemented using non-resonant acoustic elements. The numerical study confirmed the collimation of acoustic beam from a small hydrophone behind the metamaterial device. The potential application of such novel device concept in underwater communication and medical ultrasound will be also discussed.

Contributed Papers

3:00

2pEA4. Gauge invariance approach, a unified theory of negative refraction and cloaking. Woon Siong Gan (Acoustical Technologies Singapore Pte Ltd, 5 Derbyshire Road, #04-05, Singapore 309461, Singapore, wsgan@acousticaltechnologies.com)

This is an alternative theory of negative refraction. Gauge invariance approach is used. With the discovery that negative refraction is a special case of coordinates transformation when the determinant of the transformation matrix equals -1, this is also a unified theory of negative refraction and cloaking.This is a more formal theory than Veselago's dispersion relation and a more generic theory with wider scope of applications. This is the third important application of gauge invariance to acoustic fields. The other two are invariance/symmetry gives rise to only two elastic constants in isotropic solid and time reversal invariance of acoustic equation of motion gives rise to new field of time reversal acoustics. Reflection invariance or right-left symmetry a form of gauge invariance also gives rise to negative refraction which is a reflection of positive refraction. With -1 as the determinant of the transformation matrix also produces naturally the negative values of permeability and permittivity for the em wave and negative values of mass density and bulk modulus for the acoustic wave. Reflection invariance produces negative refractive index, avoiding the uncertainty of choosing the negative sign of the square root when using the dispersion relation.

3:20

2pEA5. New acoustics, based on metamaterial. Woon Siong Gan (Acoustical Technologies Singapore Pte Ltd, 5 Derbyshire Road, Singapore 309461, Singapore, wsgan@acousticaltechnologies.com)

With the capability to fabricate acoustical metamaterial:phononic crystals, double negative material and materials with material parameters based on the predetermined direction of sound propagation, one can control and manipulate sound propagation in solids and fluids. This gives rise to new forms of refraction, diffraction and scattering, the three basic mechanisms of sound propagation in solids and fluids.Besides the capability of designing perfect lens based on negative refraction, the coordinates transformation can yield a lens of going beyond linear refraction to the nonlinear case of bending the sound wave to any direction of our choice. A generalized Snell's law based on curvilinear coordinates is derived and possible application given. Refraction between two media of different parities also produces new phenomena which can be utilised in resonators and waveguides. A new rigorous theory of diffraction is formulated based on material parameters enabling the manipulation of diffraction and defeating the diffraction limit. Nonlinear acoustics based on nonlinear phononic crystals is also considered. The behaviour of solitary wave is studied. A new Christoffel equation based on double negative material is also derived.

3:40

2pEA6. Making an acoustic sensor undetectable with a pair of single-negative materials. Tao Xu (Key Laboratory of Modern Acoustics, MOE, and Institute of Acoustics, Department of Physics, Nanjing University, Nanjing 210093, China, xutao.nju@gmail.com)

We have proposed a two-dimensional cylindrical multi-layers device which can make a cylindrical acoustic sensor undetectable when the thickness of each layer of our device is much smaller than the wavelength of the incident wave. Our device only consists of a pair of complementary isotropic single-negative materials instead of double-negative ones. The parameters of materials in the device are homogenous and independent of those of host material as well as the cloaked object. Full wave simulations by finite element method are performed to verify the feasibility of the device. This proposal would greatly reduce the difficulty in both experimental design and fabrication.

4:00–4:20 Break

4:20

2pEA7. Achieving a broadband plasmonic-type acoustic cloak using multilayered, spherically isotropic elastic shells. Matthew Guild, Michael Haberman (Dept. of Mechanical Engineering and Applied Research Laboratories, The University of Texas at Austin, Austin, Texas 78713-8029, mdguild@utexas.edu), and Andrea Alù (Dept. of Electrical and Computer Engineering, The University of Texas at Austin, Austin, Texas 78712-0240)

Previous work [Guild, Alù and Haberman, *Wave Motion* **48**, 468-482 (2011)] has shown promising results for the cloaking of a sphere using non-resonant scattering cancellation. Originally developed for electromagnetic waves using plasmonic materials, this plasmonic cloaking technique differs from transformation-based cloaking, canceling the scattered field within the surrounding medium while allowing the incident wave to interact with the object. In contrast to a transformation-based cloak, construction of a plasmonic-type acoustic cloak can be achieved using ordinary fluid and isotropic elastic shells. Although such designs have been shown to be extremely effective, the bandwidth is fundamentally limited in this configuration due to the flexural shell resonances occurring within the isotropic elastic layers. To mitigate these effects, spherically isotropic elastic layers can be utilized, which allow for independent control of the transverse (tangential) layer properties. In this paper, full-wave analytic expressions are developed for a multilayered plasmonic-type acoustic cloak, consisting of an arbitrary number of spherically isotropic elastic layers. Using these expressions, examples for materials of practical interest in acoustic applications will be presented and discussed.

4:40

2pEA8. The theoretical and experimental analysis of an elastic hyperlens for far-field subwavelength imaging in a plate. Hyung Jin Lee, Hoe Woong Kim, and Yoon Young Kim (Seoul National Univ., 599 Gwanak-ro, Gwanak-gu, Seoul 151-742, Republic of Korea, hyungjinlee@snu.ac.kr)

The hyperlens is a novel imaging device that is capable to overcome the diffraction limit, a fundamental limit due to exponentially-decaying wave components containing subwavelength information. Because a hyperlens converts evanescent waves to propagating waves and magnify them, it can transfer a subwavelength image to the far field beyond the hyperlens. These interesting capabilities can be achieved through an extreme anisotropy in an effective permittivity tensor in optics while in a density tensor in acoustics. The extreme anisotropy in an elastic regime, however, is determined by elastic stiffness rather than density unlike in acoustics. We show, by using the homogenization method, how to evaluate the effective elasticity tensor of an elastic plate hyperlens consisting of alternating layers of metal and air. To experimentally demonstrate far-field subwavelength imaging by the hyperlens, a specially-configured experimental setup is suggested. The ratio of the input power to the output power is investigated both theoretically and experimentally.

5:00

2pEA9. Experimental study of backscattering enhancement using acoustic double-positive metamaterials. Wenlin Hu (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, huwenlin1984@yahoo.cn), Yuxian Fan, Peifeng Ji, and Jun Yang (ditto)

Backscattering enhancement can be achieved by superscatterers which magnify the scattering cross section of an object. To realize this kind of scatterers, a posible approach on the conception of complementary media requires double-negative materials which may induce sound absorption. In this paper, a method for backscattering enhancement by using double-positive acoustic metamaterials is presented. A shell with discretely layered metamaterials is coated on the rigid cylinder to enhance the strength of backward scattering. Density and bulk modulus of the metamaterials are both positive and limited in a reasonable range. An experiment of backscattering enhancement is investigated. The layered shell is constructed from the metamaterials employing acoustic transmission line network. The scattering sound fields of the object and the layered structure with the same geometric scale are measured in a parallel plate waveguide. Scattering properties on both backward and forward direction are analyzed from the experimental results and the scattering enhancement effect with positive metamaterials is discussed.

5:20

2pEA10. The dispersion effects in acoustic cloak with multilayered homogeneous isotropic materials. Li Cai, Jihong Wen, and Xiaoyun Han (Key Laboratory of Photonic and Phononic Crystals of Ministry of Education and Institute of Mechatronical Engineering National University of Defense Technology, Changsha 410073, China, caizhou2008@yahoo.com.cn)

In the last decade, the coordinate transformation theory that developed to design electromagnetic (EM) cloaks has been further extended to design acoustic cloaks, which promise potential applications such as sound transparency and insulation. The realization of acoustic cloaks depends on the metamaterials with anisotropic density and bulk modulus. It has been shown that alternating layers of homogeneous isotropic materials can be used to approximate the anisotropic metamaterials that acoustic cloak required. As a kind of complex material, the dispersion with frequency is important to the cloaking effect of it. In this work, we specifically examine the frequency response of the multilayer structured acoustic cloaks by the acoustic scattering theory. The acoustic scattering cross sections of the cloak constructed by multilayered metamaterials with different dispersion are presented and discussed. The relation between the cloaking effect and the dispersion are

analyzed. And the results are confirmed by numerical simulations of the finite element method.

5:40

2pEA11. Experiments in phononic crystal plates for negatively-refracted guided shear-horizontal waves by using magnetostrictive patch transducers. Min Kyung Lee, Pyung Sik Ma, Il Kyu Lee, Hoe Woong Kim, and Yoon Young Kim (Seoul National University, rozeus31@snu.ac.kr)

Experiments performed for negatively-refracted bulk shear waves in elastic media have been reported in literature, not for negatively-refracted guided shear waves in a plate. In this work, we present the negatively-refracted guided shear-horizontal wave experiment results in a thin Phononic Crystal (PC) plate. An interesting feature with negative refraction experiments in a plate, compared with those in a bulk medium, is that one can measure full wave field in a nominal base plate as soon as the negatively-refracted wave exists from a PC plate. Therefore, the actual wave path in the nominal plate can be visualized by the present experiment. For successful experiments, among others, the use of proper transducers is important. In particular, pure shear-horizontal wave generating transducers with narrow beam width are preferred. Here, we employed PSA-OPMT's (Planar Solenoid Array Orientation-adjustable Patch-type Magnetostrcitve Transducers) for wave generation and measurement. The negative refraction angle estimated from the experiments is shown to be in good agreement with the theoretically calculated values through Snell's law. For instance, the experimentally-determined refraction angle is -11.35 degree at the frequency of 220 kHz and it differs within 10% from the theoretical value, -12.67 degree.

6:00

2pEA12. Acoustic band gaps in one-dimensional Helmholtz resonator metamaterials with disperiodical defects. Dongbao Gao and Xinwu Zeng (College of Optoelectric Science and Engineering, National University of Defense Technology, gaodongbao02003@163.com)

The metamaterials containing Helmholtz resonators (HR) can exhibit two kinds of forbidden gaps. The dynamic density and effective bulk modulus are simultaneously negative in the gap of local-resonant-type. It is considered to be one of the possible materials to implement a lot of applications based on transformation acoustics. Acoustic transmission properties of one-dimensional metamaterials with periodically and disperiodically distributed HRs are calculated based on acoustic transmission line method (ATLM). The relationship between the transmission coefficient and HR parameters is analyzed. The band gaps of 1D HR metamaterials with periodical defects are also investigated. The Bragg-scattering-type gaps exist at the frequencies associated with the effective periodic constant. The defect modes are observed in the gaps of local-resonant-type. Furthermore, another metamterial with gradually changed HRs is also researched as a disperiodical type. This work enriched the studies of HR metamaterials, which can be helpful for realizing new filters and cloaks.

6:20

2pEA13. Theoretical study of SH-wave propagation in piezoelectric/piezomagnetic layered periodic structures. Jinfeng Zhao, Zheng Zhong, and Yongdong Pan (Tongji University, 200092, zjfeng55@163.com)

The propagation of SH wave in the single and periodically-layered piezoelectric piezoelectric/piezomagnetic structures is studied. Both the dispersion equation and transmission coefficients are derived to reveal the wave behavior of the corresponding structures when the piezoelectricity/piezomagnetivity is ignored, or the electrical/magnetic circuit is open and closed. The zero-order mode of the piezoelectricity/piezomagnetivity ignored single plate is not dispersive and every higher order mode is dispersive with a cut-off frequency. Same features are found for the electrically/magnetically open and closed cases, except that the zero-order mode of the latter cases is no more non-dispersive. The pass bands of the piezoelectricity/piezomagnetivity ignored periodically-layered structure appear when the normalized frequency is an even integer under the normal incidence, and new stop bands will appear from the pass bands as the incident angle increases. Same features are observed for the band gaps of the electrically/magnetically open and closed cases, except that the zero-order mode of the latter case is dispersive.

6:40

2pEA14. Band structures of Lamb waves in one-dimensional piezoelectric composite plates: polarizations and boundary conditions. Xin-Ye Zou and Jian-Chun Cheng (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, xyzou@nju.edu.cn)

Theoretical studies are presented for the band structures of Lamb waves in a one-dimensional phononic crystal plate consisting of piezoelectric ceramics placed periodically in an epoxy substrate. The band structures for different polarizations and electric boundary conditions are calculated for the composite plates in this paper. It is found that the first band gap is always broadened by polarizing piezoelectric ceramics, and the band gap widths with short circuit boundary condition are always larger than that with open circuit situation for the same polarization. Employing harmonic frequency analysis, the numerical results show that the Lamb wave modes corresponding to certain frequencies in the composite plates can be modulated by the different electric boundary conditions. The researches show that it is possible to control the Lamb waves in the composite plates in the engineering according to need by choosing suitable polarizations and electric boundary conditions.

Session 2pHT

Hot Topics: Community Noise Policy Development II

Marion Burgess, Cochair
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Invited Papers

2:00

2pHT1. The current and future development of aircraft noise management measures in Taiwan. Cherie Lu (Department of Aviation & Maritime Management, Chang Jung Christian University, 396, Sec. 1, Changrong Rd., Gueiren Dist., Tainan City 71101, Taiwan, R.O.C., cherie@mail.cjcu.edu.tw), and Peter Hullah (EUROCONTROL Experimental Centre, Centre de Bois des Bordes - BP 15, 91222 Brétigny sur Orge cedex, France)

The geographical characteristics of Taiwan have made the country heavily dependent on air transport and this, combined with the island's high population density, has augmented the importance of aircraft noise issues. Nine of the country's seventeen airports are on the main island, with eight being on remote islands. Following the corporatization of Taiwan Taoyuan International Airport, the Taiwanese Civil Aeronautics Administration (CAA) is in charge of sixteen airports, of which six are of joint civil-military use. This paper presents the current aircraft noise management measures at Taiwanese airports: noise limits for aircraft, regulations on the reduction of aircraft noise at source, aircraft noise charges, land use planning of airport neighbourhoods, improvements to airport layouts, and noise monitoring systems. After consideration of various aspects of environmental impacts due to aircraft and airport operations, the CAA has been planning the implementation of an airport environmental management system, using Taipei Songshan Airport, in the heart of the city, as the first pilot case. In addition, having reviewed work done by the international Aircraft Noise Non-Acoustic factors (ANNA) discussion group, the paper suggests methods for handling Taiwan's noise issues in a resident-friendly manner.

2:20

2pHT2. Environmental noise situation in Bangkok. Krittika Lertsawat (Project on the Draft Law of the Environmental Adjudicating Process, Thailand, krittikanonoise@gmail.com), Lalin Kovudhikulrungsri (International Institute of Air and Space Law, Leiden University, The Netherlands), Surocha Phoolsawat (Air Quality and Noise Management Bureau, Pollution Control Department, Thailand), and Tanaphan Suksaard (Environmental Research and Training Center, Thailand)

The overview of the environmental noise management policy and regulations will be briefly discussed in this presentation, including the problem analysis on the noise management issue in Bangkok. All of the applicable policies and regulations related to environmental noise issues in Bangkok were only dominated by the road traffic noise sources. The environmental noise situation in Bangkok will be illustrated and discussed in this presentation, seeking the comparative discussion from the international experts and practitioners for the better determination methods and technologies, applied for the future policies and regulations.

2:40

2pHT3. Noise control policy in Brazil and South America. Samir N. Y. Gerges (Federal University of Santa Catarina UFSC, Brazil, samir.acustica@gmail.com)

There are noise policies and regulations in Brazil for occupational, community, and product noise. (1)Noise in the workplace, MTE 3214-1978, is a regulation of the Minister of Labor which specifies 85 dBA – 8 hours sift with 5 dBA rate (this should be changes to 3 dB). (2)Community Noise: There are two regulations of the Minister of the Environment covering noise which affects industrial, commercial, social, recreation, and political activities. A- Silence regulation NBR10151: SPL= 35-45 dBA between 10 PM to 6 AM. B- Comfort regulation NBR10152: Table of SPL (35 to 60 dBA) or NC curves for each place (hospital, hotel, residence, offices, school, etc.) range from C= 30 to 55. (3)Brazil also has Standard NBR 13910-1, 2, and 3:1999 which recommends noise labeling using the sound power level. Since 2001 the working party has been in power in Brazil, and since then industrial noise regulations have been enforced. If industrial noise cannot be reduced at the source, hearing protectors are required. All hearing protection devices must go through attenuation measurements in Brazil (ANSI S12.9-2008(B) subject fit method. Most of South America countries have similar regulations as those in Brazil. The Ibro-American Federation of Acoustics (FIA)in the last years has played an important role in noise control engineering education through its congresses in 1998 (Brazil), 2000 (Spain), 2002 (Cancun with ASA), 2004 (Portugal), and 2006 (Chile).

3:00

2pHT4. Environmental noise management in Korea. Sang Kyu Park (#303 Baekwoonkwan Maeji-Li Heungup-Myun Wonju-Si, Kangwon-Do, tankpark@yonsei.ac.kr), Jae Sik Park, Hyo Seok yun, and Soung Cheol Yoon (#304 Baekwoonkwan Maeji-Li Heungup-Myun Wonju-Si, Kangwon-Do)

More than 40% of Koreans are exposed to excessive noise levels which can lead to serious health effects, community annoyance and sleep disturbance. Government policy and management of the noise are necessary to solve these problems. In this paper, some projects carried out to mitigate the excessive noise levels under the support of the Ministry of Environment in Korea have been introduced. They include the projects such as making a long-term plan in the field of noise management, amendment of the existing legislation and noise measurement methods, establishment of noise map directive, construction of traffic noise monitoring system, and management of low frequency noise.

3:20–6:00

International Consortium on Noise Issues in Emerging and Developing Countries: Workshop on Priorities and Approaches for Noise Policies

TUESDAY AFTERNOON, 15 MAY 2012 — S228, 2:00 P.M. TO 5:40 P.M.

Session 2pMU

Musical Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Acoustics of Traditional Musical Practices and Instruments I

Jean-Pierre Hermand, Cochair
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Invited Papers

2:00

2pMU1. Culture specific approaches in music content analysis. Xavier Serra (Universitat Pompeu Fabra; Roc Boronat 138, 08018 Barcelona, Spain, xavier.serra@upf.edu)

The extraction of culturally meaningful features from audio recordings of music different from the western repertoires that are the most studied requires new signal analysis techniques and machine learning methods. The fact that many music traditions have fundamental differences from western ones, such as different musical instruments, tuning systems, performance styles, or musical forms, imply that at the level of feature analysis, most of the descriptors and extraction methodologies being used to analyze western music are not appropriate, or at least they have to be developed much further. Culture specific issues have profound research implications, offering new research problems and requiring new approaches. In this article we will present some initial results in the audio content analysis of the classical music traditions of India (Hindustani and Carnatic) and Turkey (Ottoman) especially for the issues of melodic and rhythmic description. This research is being carried out within a large project entitled CompMusic that aims to promote and develop multicultural perspectives in music computing research. In this project we want work on culture specific music problems with the goal to find new computational methodologies of interest for a wide variety of music information processing problems.

2:20

2pMU2. Computational analysis of Maqam music: from audio transcription to structural and stylistic analyses, everything is tightly intertwined. Olivier Lartillot (University of Jyväskylä, olartillot@gmail.com)

Automated transcription of audio recordings into musical scores is a very challenging problem. Robust technological solutions are so far limited to simple cases and specific conditions, such as the focus on specific tractable musical instruments. The traditional conception of transcription as the inference of a single layer of notes ignores one core characterization of music as a multi-layer encapsulation of events of various scales (notes, gestures, motifs, phrases, etc.), where higher-level structures contextually guide the progressive discovery of lower-level elements. Modeling the emergence of these multiple structural layers, although complicating the problem, is in our view the only way to obtain a robust automation of music transcription, which is modeled here in the form of a multi-layer and recursive auditory scene analysis. Additionally, culture, as the experience of previous similar types of music, plays another essential role in guiding the more ambiguous aspects of music understanding. A previously proposed modeling of the impact of culture in structural understanding, applied in particular to Arabic Maqam music, is generalized here to the study of the influence of such cultural knowledge on music analysis, and in particular on the lowest layers of note transcription.

2:40

2pMU3. Differences in instrument construction and performance practices among musical traditions reveal and guide different aesthetic attitudes towards timbre. Pantelis Vassilakis (Audio Arts and Acoustics Department, Columbia College Chicago, 33 E. Congress Parkway, Suite 601M, Chicago, IL 60605, pvassilakis@colum.edu)

Musical aesthetic judgments reflect how each musical tradition chooses to interpret and value contextual, functional, performance, formal, and sonic aspects of musical pieces. Elaborate instrument construction techniques and performance practices devoted to the exploration of timbre (sound color) variations across musical traditions indicate that timbre is a sonic aspect on which musical aesthetic judgments are often based. Intercultural differences and intra-cultural consistency of timbre interpretation illustrate the cultural bases of understanding and evaluating sound color. Close examination of musical instrument construction and performance practices, accompanied by acoustical analyses of the relevant sound signals, can reveal the types of musical timbres and timbre variation degrees a given tradition is after, providing insights on the relationship between timbre and a tradition's musical aesthetic values. The sophisticated ways devised to produce and manipulate auditory roughness within the musical contexts addressed in this presentation (Indian tambura accompaniments; Middle Eastern mijwiz improvisations; Bosnian ganga songs) will be contrasted to the limited opportunities for such explorations afforded within western art musical contexts, paralleled by equally contrasting aesthetic attitudes towards auditory roughness's meaning and value.

3:00

2pMU4. Dynamics of the Himalayan singing bowl. Brandon August, Aditya Mahara, and Thomas Moore (Department of Physics, Rollins College, Winter Park, FL 32789, baugust@rollins.edu)

The singing bowl, commonly known as the Tibetan or Himalayan singing bowl, is an idiophone indigenous to many Asian cultures. Singing bowls are usually made of brass, which is hammered into a nearly symmetrical hemispherical shell and then hand-turned on a lathe. A sound is produced by either striking the bowl or rubbing the surface with an excitation stick referred to as a puja. We report on an investigation of the sound generation mechanism of the singing bowl, with an emphasis on understanding the origin of the sound created by the stick-slip excitation mechanism that occurs while rubbing the bowl with the puja. It is shown that both the radial and tangential motion of the puja experience exponential gain during the excitation process, and the feedback mechanism required to produce this behavior is discussed. A slow oscillation in the sound produced by the bowl is explained by the dominance of a single vibrational mode that rotates at the angular speed of the puja.

Contributed Papers

3:20

2pMU5. The Ethiopian lyre *Bagana*. An ethno-acoustical study. Stephanie Weisser (Musical Instruments Museum, Rue Villa-Hermosa 1, B-1000 Brussels, Belgium, stephanieweisser@gmail.com), Jean-Pierre Hermand, and Qunyan Ren (Environmental Hydroacoustics Laboratory, Av. Fr. Roosevelt 50, CP 194/05, 1050 Brussels, Belgium)

The Ethiopian lyre *bagana* is usually finger-plucked and monodic, with a skin soundboard and ten gut strings tuned to low-frequency pitches (ca 50-150 Hz). Its most important sonorous characteristic is its buzzing sounds, produced thanks to leather pieces placed between each string and the wide bridge. This study is based on a corpus of sounds produced by eight different instruments played by a virtuoso master and recorded in situ with and without the leather pieces. The sounds have been analyzed namely through calculation of several timbre descriptors based on time-domain, sinusoidal harmonic model and short-time Fourier transform representations. These results have been confronted with ethnomusicological analyses of the repertoire and the socio-cultural aspects of the *bagana*, in order to understand how the sound qualities are dealt with by the players. These joint analyses show that the very distinctive buzzing quality of the *bagana* sounds can be linked with auditory roughness and inharmonicity descriptors (indicating that it is not due to noise but rather to quasi-harmonic components) while the significant timbre variation between sounds is mostly due to differences in string quality, location of the leather piece on the bridge and musicians' search to produce longest possible duration.

3:40

2pMU6. Shaping the resonance. Sympathetic strings in Hindustani classical instruments. Stephanie Weisser (Musical Instruments Museum, Rue VIlla-Hermosa 1, B-1000 Brussels, Belgium, stephanieweisser@gmail.com), and Matthias Demoucron (Institute for Psychoacoustics and Electronic Music (IPEM), Blandijnberg 2, B-9000 Ghent, Belgium)

Most chordophones of the contemporary classical Hindustani tradition are characterized by the presence of numerous sympathetic strings *taraf* (sometimes up to over 30). Generally tuned according to the *rag*, they are inserted within the handle of the plucked lutes *sitar* and *sarod*, and next to and below the main strings of the bowed fiddle *sarangi*. In some cases (e.g. some of *sarangi*'s, all of them in *sitar*), *taraf* are also equipped with a curved bridge, increasing the spectral richness of the sounds produced by these strings. Players consider the *taraf*'s response as fundamental to the instrument's sound. Based on field recordings realized in ITC Sangeet Research Academy (Kolkatta) this study aims to determine the contribution of these strings to the resulting sound of the different instruments and settings. Acoustical analyses are complemented with ethnomusicological analyses, in order to evaluate the *taraf*'s aesthetic, musical and perceptual role.

4:00–4:20 Break

4:20

2pMU7. Outside-instrument coupling of resonance chambers in the New-Ireland friction instrument "Lounuet". Rolf Bader (University of Hamburg, R_Bader@t-online.de)

The Lounuet is a friction instrument from New Ireland, Papua New Guinea, built from a wooden, round block, about 50 cm long, where three resonance chambers are carved below three lamellas which are played by the bare hand. As the driving mechanism is the same as with the bowed string, a perfect harmonic overtone series is produced showing frequencies up to 25 kHz. Although its fundamental is close to the lowest resonance frequency of the lamellas, the radiation nearly solely comes from the resonance chambers below the lamellas. Using a Microphone Array, back-propagating the radiated sound field to the open wholes of the resonance chambers, for different partials complex radiation patterns appear, showing clear relationships between the three chambers. Most of these standing waves between chambers show a coupling outside of the instrument for frequencies above ~ 600 Hz, where the air between the holes outside the instrument are separate anti-nodes of the standing wave fields. So these outside-instrument air resonance couplings are similar to those known from the Japanese shakuhachi wind instrument which there appear between the highest sound hole and the blowing hole.

4:40

2pMU8. Test study on the recording acoustics of Urheen. Jingying Zhang and Zihou Meng (Communication Acoustics Laboratory, Communication University of China, Beijing 100024, P.R. China, zhangjingying_29@163.com)

The acoustics of the instrument recording involves acoustic characteristics of the instrument as a sound source, the microphone characteristics for the recording, and the room acoustics of the recording studio. For the purpose to optimize the recording localization as well as the its setting and balance between instruments in Chinese orchestra, a series of test are carried out to measure the radiation directivity, the harmonic directivity and the sound power level of the urheen which is the most important instrument in Chinese orchestra. Based on the listening tests the best reverberation for urheen music is discussed to support the selection of recording environment and the post processing of the recorded music. A group of microphones are evaluated based on a listening test with the urheen music of different styles to search the appropriate microphone for urheen music recording. According to the study of recording acoustics of urheen, the guideline for a better uhreen recording is given.

5:00

2pMU9. Motion analysis for emotional performance of snare drums using a Motion Averaging Method. Masanobu Miura (Ryukoku Univ., 1-5, Yokotani, Oe-cho, Seta, Otsu 520-2194, Japan, miura@rins.ryukoku.ac.jp), Yuki Mito (Hitotsubashi University), and Hiroshi Kawakami (Nihon University)

Proposed here is a “Motion Averaging Method”, for the analysis of motion data of musical performance on expressing emotion obtained by a motion capture system. The method is made to analyze the motion on expressing any emotion on playing the snare drum. Specifically, the motions for an etude with each of emotion (tenderness/happiness/anger/fear/sadness) played by three trained percussionists were recorded by using a motion capture system. Obtained data (*.trc) were corrected and then adjusted among them by shifting and rotating player’s position, so as to ignore the difference of position of players on recording. Moreover, the difference among each performance in time is uniformed by stretching (expanding or contracting) the motion data based on the impact time of the performance. Expanded or contracted data were then averaged for each emotion. Obtained motion is called an “averaged motion”, which shows common motion of the performance on expressing each of an emotion. Features of the averaged motions for each emotion are measured by observing differences and tendencies among each player were investigated. This study shows a list of features of motion on playing the snare drum by three trained players. We also discuss about the application of the method.

5:20

2pMU10. Vibrotactile music systems for co-located and telematic performance. Deborah Egloff, Jonas Braasch, Phil Robinson (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180, eglofd@rpi.edu), Doug Van Nort, Pauline Oliveros (Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180), and Ted Krueger (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180)

Building on previous findings, the project reported here expands on the idea of how the modality of touch contributes to the sensory perception of sound in the presence of vibro-tactile events. The SenseAble1.0, a vibrotactile interface, was developed to transform the auditory parameter space into one that is adequate for haptic perception by the means of eight individually controllable actuators and machine learning algorithms. Due to the process of sensory substitution, people with hearing impairments can use the vibrotactile display to “listen” to music and perceive exterior acoustic stimuli through the sense of touch. A psychophysical pilot study investigated (i) frequency discrimination and pitch direction as well as (ii) interval size recognition of the haptic modality, and (iii) adaptation effects of the somatosensory system. Live performances using the SenseAble as a vibrotactile music system illustrate to what extent musical communication between ensemble members and a hearing-impaired musician is possible. A new prototype with enhanced spatial resolution properties for perceiving vibrotactile stimuli will be introduced as well. It was designed to improve the usability of sensory substitution systems in real and virtual environments and to maximize the efficiency of co-located and telematic performances in terms of musical communication and interaction.

Session 2pNSa

Noise: Numerical Methods in Noise II

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Contributed Papers

2:00

2pNSa1. Sound power level calculation of industry sources—simulation. Fabian Probst (DataKustik GmbH, Gewerbering 5, 86926 Greifenberg, Germany, info@datakustik.de)

Modern calculation methods enable the simulation of the sound power level measurement of complex industry sources applying the enveloping surface method. According to this procedure receiver points are created and distributed on spherical or cuboid surfaces following the relevant International Standards like ISO 3744 or ISO 3746 or on cylindrical or other regular surfaces depending on the requirements of any underlying machine specific standard. The sound pressure levels at these receivers on the enveloping surface and the resulting "effective" sound power level is calculated. The comparison of these sound power levels determined by simulation and the sound power levels of the sources in the model shows the influence of the geometric shape of the measuring surface. Such a simulation of the sound power determination with the enveloping surface method is applicable even in cases where a real measurement would not be possible, e. g. if the effective sound power level of a complete power plant or of a city with all its traffic sources shall be determined. The method and some practical examples are demonstrated.

2:20

2pNSa2. Finite-difference time-domain simulation of sound propagation through turbulent atmosphere. Loïc Ehrhardt, Sylvain Cheinet (French-German Research Institute of Saint-Louis (ISL), 5 rue du Général Cassagnou, BP 70034, 68301 Saint-Louis Cedex, France, loic.ehrhardt@isl.eu), and Daniel Juvé (Laboratory of Fluid Mechanics and Acoustics (LMFA), URA CNRS 263, Ecole Centrale de Lyon BP 163, 69131 Ecully Cedex, France)

Sound propagating outdoors is influenced by turbulent fluctuations of the atmosphere. Unfortunately, theories only exist in limited configurations and outdoor experimentation is difficult. Numerical simulation is a good alternative for fully understanding the physics in place. The Finite-Difference Time-Domain (FDTD) model has already proven to reproduce many aspects of linear acoustics. It remains to demonstrate that it catches the physics of turbulence-induced effects. This is the aim of this contribution. FDTD simulations of sound propagation through turbulent atmosphere are performed. The general behavior and the statistical characteristics of the sound field are evaluated and compared to available theories. In the limiting configurations of weak or strong (saturated) sound perturbations, the simulations are in excellent agreement with the tested theories. In the intermediate configurations, some theoretical results agree with the simulations, while others show notable discrepancies. For example, the FDTD results suggest that there is a significant correlation between phase and amplitude fluctuations. These findings generally suggest that FDTD is an appropriate modeling tool to investigate sound propagation through complex configurations of the atmospheric fluctuations.

2:40

2pNSa3. The effects of the source and ground parameters on outdoor sound propagation using the FDTD. Han Kaifeng, Zeng Xinwu, and Zhang Zhenfu (National University of Defense Technology, shunzhihan@sina.com)

The sound propagation outdoors is widely studied in several areas such as transportation noise mitigation, biological studies, security and military activities. Several special factors are important when sound waves propagate more-or-less horizontally near the ground, such as ground characteristics, the use of sources and so on. The simplest effect of the ground on the sound field is the interference between the direct and reflected sound fields. However, the ground surfaces are neither rigid nor impervious to air flow; it will cause higher sound energy loss. Secondly, the parameters of the source distribution and the frequency emission also determine the amount of loss. To quantify the influences of the source parameters and inhomogeneous ground conditions on the sound field outdoors, a practical FDTD implementation is constructed in this paper. The mixed influence of ground characteristics and source conditions are then included. The numerical results showed the source distribution and the frequency had large effects on the actual sound pressure measurement in the specific situation. The ground boundary conditions are the main effects on the basic phenomena of outdoor sound propagation.

3:00

2pNSa4. Stabilization of time-domain acoustic boundary element method for sound radiation problems. Hae-Won Jang and Jeong-Guon Ih (KAIST, 305-701, haewon@kaist.ac.kr)

Time-domain acoustic solution from the Kirchhoff integral equation for the exterior problem is not unique due to the presence of fictitious internal modes and also suffers from the instability that stems from the time marching scheme. In this work, methods to stabilize the time-domain acoustic boundary element calculation were suggested. Low-order fictitious internal modes within the effective frequency range of boundary element calculation were suppressed by the newly formulated time-domain CHIEF (Combined Helmholtz Integral Equation Formulation) method. Additional interior points were included, similar to frequency-domain problems, to satisfy the zero pressure constraint considering the shortest retarded time between boundary nodes and interior points. However, the calculation was yet unstable due to remaining unstable high-order modes beyond the effective frequency limit. To further stabilize the computation, unstable high-order internal modes were nullified using the wave vector filtering method. In comparison with the time-domain Burton-Miller formulation, the proposed method has no hyper-singular integral and the monotonically increasing instability was not observed. As a test example, sound radiation from a pulsating sphere was used and a good stabilization performance was shown. Average relative-difference norm of the stabilized time response from the analytic solution was 2.7%. (This work was partially supported by BK21 project)

3:20

2pNSa5. Development of a practical method for determining noise contribution from a large extruded panel using sound intensity method and optimized finite element model. Anne Shen, Jiumei Cheng, and Fusheng Sui (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, anne.xy.sh@gmail.com)

A practical method is proposed for determining noise contribution from a large extruded panel to the sound pressure level inside an enclosure based on acoustic measurements and numerical simulations. A finite element model is constructed for a rigid enclosure with one surface replaced with an extruded, curved panel. The interior sound pressure field is optimized using experimentally obtained reverberation time. The vibratory responses and sound intensity of the extruded panel under mechanical excitation are measured and the interior sound pressure levels are recorded. The structural-acoustic sensitivity term of the interior point is determined and used to update the vibroacoustic responses of the finite element model. The optimized numerical model can be used to predict panel noise contribution for any given excitation source.

3:40

2pNSa6. Sound propagation along a long partial enclosure. S. H. K. Chu (Department of Building Services Engineering, The Hong Kong Polytechnic University, Kowloon, Hong Kong, 09902976r@connect.polyu.hk)

The acoustical properties of sound in a long rectangular partial enclosure with various opening sizes and positions are investigated numerically in the present study. The finite element method is adopted to estimate the mode shapes across the cross section of the partial enclosure inside a free field environment. Some acoustic modes with patterns similar to some of the rigid-wall duct modes are found. The sound propagates in form of modes inside the partial enclosures as in the rigid-wall duct case. The long partial enclosure leaks sound and the opening radiates sound into a free space in the numerical model. The sound radiation is associated with the interactions between acoustic mode shape of the partial enclosure, opening size and position. Results indicate that the behaviour of acoustic pressure radiated from the opening is a significant effect on the resonance frequencies.

4:00–4:20 Break

4:20

2pNSa7. An efficient time domain solver for the acoustic wave equation. Ravish Mehra (UNC Chapel Hill, Sitterson Hall, Chapel Hill, NC 27599, ravishm@cs.unc.edu), Nikunj Raghuvanshi (Microsoft Research, One Microsoft Way, Redmond, WA 98052), Lauri Savioja (Aalto University School of Science, Department of Media Technology P.O. Box 15400 FI00076 AALTO, Finland), Ming Lin, and Dinesh Manocha (UNC Chapel Hill, Sitterson Hall, Chapel Hill, NC 27599)

An efficient numerical solver for time domain solution of the wave equation for the purpose of propagation in small and large acoustic spaces is presented. It is based on the adaptive rectangular decomposition technique that subdivides a space into rectangular partitions and within each partition utilizes the analytical solution of the wave equation for spatially invariant speed of sound. This technique allows numerical computations in kilohertz range for auralization and visualization purposes. This can help engineers to quickly locate geometric features responsible for acoustical defects in practical engineering applications like noise control. It is demonstrated that by carefully mapping all the components of the technique on the GPU architecture, significant improvement in performance can be achieved while maintaining accuracy comparable to a high-order finite difference time domain (FDTD) solver. It is an order of magnitude faster than corresponding CPU-based solver and three orders of magnitude faster than the CPU-based FDTD solver. This technique can perform a 1 s long simulation on complex-shaped 3D scenes of air volume 7500 cu m till 1650 Hz within 18 min on a desktop machine. The ideal session for this work is "Computational Acoustics".

4:40

2pNSa8. Validation of 3D numerical simulation for acoustic pulse propagation in an urban environment. Ravish Mehra (UNC Chapel Hill, Sitterson Hall, Chapel Hill, NC 27599, ravishm@cs.unc.edu), Nikunj Raghuvanshi (Microsoft research, One Microsoft Way, Redmond, WA 98052), Anish Chandak (UNC Chapel Hill, Sitterson Hall, Chapel Hill, NC 27599), Don Albert, Keith Wilson (ARO), and Dinesh Manocha (UNC Chapel Hill, Sitterson Hall, Chapel Hill, NC 27599)

Acoustic pulse propagation in an urban environment is a complicated phenomenon with various acoustic effects like scattering, diffraction and reverberation, produced due to the presence of buildings. This work models acoustic propagation in an urban environment using adaptive rectangular decomposition (ARD), a time domain numerical simulation technique for solving the acoustic wave equation. It subdivides a space into rectangular partitions and utilizes analytical solution of the wave equation inside each partition along with sixth order Perfectly Matched Layer boundary conditions. The theoretical predications of the simulation are validated with experimental measurements performed in an artificial village. The simulation captures the near-periodic reflections and reverberation effects produced at the line-of-sight positions. For non-line-of-sight positions, high-order scattering and diffraction effects are also accurately modeled. We perform a comparison with the 2D FDTD method proposed by Liu and Albert[2006]. Since ARD is a 3D simulation technique, it captures all possible wave propagation paths, including the rooftop paths that not handled by the previous method. The predicted acoustic responses produced by the simulation match well with the measured responses for most sensor locations. Disagreements between simulation and measurements are discussed as well. The ideal session for this work is "Computational Acoustics".

5:00

2pNSa9. Computation of wall pressure fluctuations and flow induced noise by large eddy simulation. Zhang Nan, Shen Hong-cui, and Tian Yu-kui (POX.116, WuXi City, Jiangsu Province, China, zn_nan@sina.com)

In industrial practice, the cavity-type oscillation is undesirable from the perspective of inducement of structure vibration and fatigue, generation of noise and drastic increase in drag on the body. A numerical work for the prediction of wall pressure fluctuations and flow-induced noise of cavity is performed in the paper. Firstly, the wall pressure fluctuations of a plate are computed and compared with experimental results of Small Anechoic Flow Facility in CSSRC. The robustness of large eddy simulation (LES) in unsteady flow calculation is analyzed. Secondly, the calculations of wall pressure fluctuations of shuttle holes are accomplished. The power spectra of wall pressure fluctuations are analyzed. The numerical predictions are compared with measured data. Finally, the flow induced noises of three cavities are predicted through LES and FW-H acoustic analogy. The computed results including flow pattern in cavity, vorticity distribution and radiated sound spectrum are analyzed. The computed results are compared with experimental data of Large Circulation Channel in CSSRC, and the numerical prediction method is validated. It shows that the numerical prediction method in the paper is credible. Key words: wall pressure fluctuations; flow induced noise; Large eddy simulation; FW-H acoustic analogy; cavity

5:20

2pNSa10. Acoustic generation of flow past an in-duct baffle. H. Y. H. Chan, R. C. K. Leung, and Y. S. Choy (Department of Mechanical Engineering, The Hong Kong Polytechnic University, Hong Kong, P. R. China, mmrleung@inet.polyu.edu.hk)

Recent research was carried out to calculate the acoustics generation induced by flow numerically. In this paper, a modified hybrid acoustics/viscous splitting technique was employed in a computational simulation to calculate the acoustic generation when flow passes through a baffle in duct. The acoustics/viscous splitting technique for Computational Aeroacoustics (CAA) was first developed by Hardin and Pope. Traditionally, CAA problems were computed by Direct Numerical Simulation (DNS) to solve the compressible Navier-Stokes equation. However DNS is very expensive in terms of computation power and time consuming. The Hybrid approach is regarded as a alternative of solving the aeroacoustics problem. Perturbation technique is employed to calculate the acoustics perturbation which

superimposes with the incompressible solution to obtain the compressible flow solution. In this paper, the hybrid CAA technique was developed and applied in a 2D duct with single baffle.

5:40

2pNSa11. Aeroacoustic source modeling for the Galbrun equation. Feng Xue and Ben Tahar Mabrouk (Laboratory Roberval UMR 6253, University of Technology of Compiègne, 60205 Compiegne cedex, France, xue.feng@utc.fr)

A new numerical technique for acoustic noise generation, based on the Galbrun equation is developed. The source term developed is from a calculation of an incompressible flow part. The acoustic equations are obtained from the numerical solution of a system of perturbed, compressible, linear equations, as the Galbrun equation. The Galbrun equation is solved by a high order mixed finite elements method in the frequency domain. Two examples to valid our source model are given. The first example is a pulsating sphere in one dimension. The result is compared with other known approaches. This example is considered as a first validation case of the code. The second example is applied for two co-rotating vortices in two dimensions. The result is compared with an analytic solution.

6:00

2pNSa12. Evaluation of barrier performance by direct environmental noise simulation. K. H. Seid, R. C. K. Leung, and G. C. Y. Lam (Department of Mechanical Engineering, The Hong Kong Polytechnic University, Hung Hum, Kowloon, Hong Kong, mmrleung@inet.polyu.edu.hk)

After a city has experienced rapid urbanization, it is usually left with many pollution problems (e.g. air, noise, etc.) that hinder its further sustainable growth. Worsen by the rapid increase in the demands of land transportation for high population mobility; the traffic noise inevitably becomes a serious environmental problem. In countermeasure to the traffic noise problem and to compromise between limited space and costing, noise barriers are commonly implemented to protect sensitive land uses from noise pollution by mean of stopping, deflecting or reducing the noise propagation. Although recent studies have shown that the use of two or more screens in the barrier profile can enhance the diffraction efficiency of plane barriers in noise reduction without increasing its height. However, ground reflections are seldom included in the analysis, which could be a critical factor in the practical point of view. In this paper, a numerical technique derived from the high-fidelity time-domain computational aeroacoustics is adapted to direct simulation of environmental noise with an illustration of noise propagation over barriers. The robustness and accuracy of the method for predicting noise propagation will be discussed.

TUESDAY AFTERNOON, 15 MAY 2012 HALL C, 2:00 P.M. TO 6:20 P.M.

Session 2pNSb

Noise: Application of Geographic Information Systems to Manage and Control Urban Noise

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Invited Papers

2:00

2pNSb1. Experiences in development and maintenance of Silence-GIS. Sven Erwin Hartog van Banda (DGMR-Softnoise, P.O. Box 370, NL-2501 CJ Denhaag, The Netherlands, ha@dgmr.nl)

Silence is a GIS (Geographical Information System) build on ArgGIS 9.3.1. This large scale noise management system has been designed for the Dutch Highway Authorities. The aim of the system is to support decisions on Dutch and European noise policy and to predict the effect of future measures on the noise exposure of the population in The Netherlands. The information needed to perform noise calculations was divided over different departments. There was a great need for standardization and integration of the different data sets. This was maybe the largest challenge of the Silence project. For the development an agile development method was chosen that involved the product owner during all the stages of the development. Terms were introduced like sprints, scrum meetings, back log, burn down charts, pigs and chickens. To keep the system up to date contracts were made between the product owner and the departments that supplied the data. This paper gives insight in the challenges and benefits of Silence, the advantages of an agile development method and gives an overview of the system hardware, the IT infra structure and the geographical database.

2:20

2pNSb2. The latest development and application of noise mapping in Hong Kong. Chi Wing Law, Aaron Shiu Wai Lui, and Maurice Kwok Leung Yeung (Environmental Protection Department, Hong Kong SAR, 26th Floor Southorn Centre Hong Kong, cwlaw@epd.gov.hk)

In recent years, there has been dramatic enhancement of computation power, rapid development in Geographic Information System (GIS), three-dimensional (3D) computer graphic and virtual reality technology. Digital topographic and mapping data are also widely available and ultilised for various applications. This had facilitated the rapid advancement in road traffic noise assessments and data presentation. Two-dimensional (2D) and even 3D noise mapping over a large geographical area has now become a much more manageable task. Hong Kong has seized the opportunity in developing advanced 3D noise modelling techniques and noise mapping in support of it's policy evaluation and formulation initiatives. State of the art 3D GIS tools, interface applications with large scale noise modelling software, sophisticated information technologies for data streaming and rendering were tried out in Hong Kong. 3D noise models were presented in meeting and uploaded to web site for consultation with local residents to encourage public participation. Also, the techniques were employed to more accurately determine the noise exposure of the population, the eligibility and overall benefits of applying low noise road surface to road sections in a large area, the determination of noise exposure of residents in support of social survey.

2:40

2pNSb3. Geographic information in noise and vibration management solutions. Douglas Manvell (Bruel & Kjaer Sound & Vibration Measurement A/S, Skodsborgvej 307, 2850 Nærum, Denmark, douglas.manvell@bksv.com)

Environmental noise and vibration management is important in enabling businesses such as construction sites, mines, airports, etc. to operate within legal limits. Increasingly, businesses understand the value of communicating impact with communities and other stakeholders. Management includes planning and control of noise and vibration levels around the site and informing stakeholders of actions to be taken. Planning is normally based around calculation software while control often must rely on monitoring. Now, real-time measured data can link with calculation models to provide automatically updated noise maps giving businesses quick feedback to help optimise their actions whilst ensuring compliance with limits. Geographic information is used in many aspects of environmental noise and vibration management solutions. Geographical data such as from AutoCAD, Geographical Information Systems (GIS) and aerial photographs are becoming more widespread and cheaper and new solutions are appearing on the market. Geo-referencing is also central for the interaction between measurements, noise calculations and sensitive receptors. This paper presents the use of geographic information in environmental noise and vibration management solutions including the use of web-based GIS services to help businesses manage their noise and vibration impact. Examples of the use of GIS solutions to directly engage with communities will be presented.

3:00

2pNSb4. Application of Geographic Information Systems (GIS) and related techniques for the management and control of noise in various urban structures and cultures. Jian Kang (Harbin Institute of Technology, Harbin 150001, China; University of Sheffield, Sheffield S10 2TN, UK, j.kang@sheffield.ac.uk)

This paper explores possibilities and potentials of integrating Geographic Information Systems (GIS), noise mapping and other related techniques for the management and control of noise in various urban structures and cultures. This includes the use of noise maps and GIS information in examining the relationships between environmental noise and social-economic factors; comparison of noise resistance of different urban structures based on micro- and macro-scale sound propagation models; effects of building arrangements in a given urban area based on genetic algorithms; and the development from urban noise maps to urban sound maps and urban soundscape maps, taking perception and cultural factors into account.

3:20

2pNSb5. A practical web-based workflow allowing authorities to meet the requirements of the END for 2012 and beyond. Chris HOAR (NGIS China Ltd., 501, Chao Building, 143-5 Bonham Strand East, Sheung Wan, HK, chris.hoar@ngis.com.hk)

Local, state and national authorities and organisations, are increasingly searching for technology-based approaches to meet the legislative and practical requirements associated with addressing community noise issues. This has been a necessity rather than luxury as the breadth and depth of activities that need to be performed by authorities has expanded to the point where it is no longer feasible to assign these to small specialised teams. The ODEN system discussed in this paper addresses these needs by providing an easy-to-use, accessible set of web-based tools that generalist and specialist users alike can use to perform required tasks. Utilising the LimA calculation engine, the system provides the ability for users to edit and manipulate model input data, perform a variety of road, rail, aircraft and industry noise calculations, generate reports and statistical analyses and visualize results as maps and 3d views. The system provides a framework which allows central mapping authorities to coordinate the activities and workflows supporting strategic noise mapping. For example by distributing workflows to stakeholder communities, the system assists authorities in preparing base data, provides tools for involved agencies to update this base data and finally tools to produce EU compliant noise maps and reports.

3:40

2pNSb6. Application of Geographic Information Systems to manage and control urban noise. Hardy Stapelfeldt (SoftNosie GmbH, D 44141 Dortmund, Germany, info@stapelfeldt.de), and Florian Pfäfflin (IVU-Umwelt GmbH, D 79110 Freiburg, Germany)

Environmental Acoustic Software products support different levels of interaction with GIS Software packages. Depending on the achieved level of interaction its value in project applications will rise. Different levels are discussed and demonstrated on practical applications: 1. Standard file exchange formats with GIS adaptation matching the needs of the Acoustic Software, e.g. QSI data format according to German DIN 45687 2. Standard file exchange format with object and attribute definition ruled by GIS software 3. Direct data communication, using e.g. geo-database of GIS application 4. "On the Fly" Pre-processing of GIS ruled attribute content ahead of noise calculation 5. Using external DLL's of Acoustic Software to check and update attribute content on GIS level 6. Supplementing

GIS ability in initial data refinement by calling specific geometric and attribute manipulation tools of the Acoustic Software. 7. Steering any automated processing of complex workflows Automated workflows have been used in EU conform Noise Mapping for the German States of Nordrhein-Westfalen (~40.000 km^2) and Thüringen (~20.000 km^2). Both example cases will briefly be lined out.

4:00–4:20 Break

Contributed Papers

4:20

2pNSb7. A comparison of surface interpolation methods for development of noise map using Geographic Information System. In Sun Park, Sang Kyu Park, Jeong Hoon Ham, and Jae Min Han (Dept. of Environmental Eng. Wonju Campus of Yonsei University, Wonju-Si, Gangwon-Do, Korea, tankpark@hdec.co.kr)

Noise map has become an important tool for the assessment and reduction of noise in cities, and can be manufactured by using ArcView GIS which is a PC-based Geographic Information System (GIS) software by Environmental Systems Research Institute, Inc (ESRI). The interpolate surface function of the ArcView GIS creates an output grid wherein each cell contains an interpolated noise value based on known noise values. In this study, several algorithms for surface interpolation, such as spline, inverse distance weighted, Kriging methods, are employed to compared the accuracy of noise levels in the downtown of a city.

4:40

2pNSb8. Current visual interfaces for acoustical modeling analysis. Dana Dorsch (DL Adams Associates, Ltd., ddorsch@dlaa.com)

With the development of advanced noise mapping software, consultants have the ability to present their clients with graphic depictions of sound level distribution in a given area. Most noise mapping software interfaces easily with Auto CAD and Google Earth, where GIS data formats can be imported and exported based on a coordinate system specified by the user. The merging of these technologies can be especially useful where the study site is located in a complex terrain or urban area. This paper describes several case studies where the development of noise maps has been improved from a basic diagram of noise contours lines to a complex, three dimensional imagery of terrain and buildings that can be rendered in Google Earth.

5:00

2pNSb9. Noise calculation method for noise mapping—what to do? Dr. Wolfgang Probst (DataKustik GmbH, Gewerbering 5, 86926 Greifenberg, Germany, info@datakustik.com)

Lots of investigations have been undertaken to find the best solution how to calculate the noise levels for noise mapping and action planning purposes. Researchers try to model the environment as detailed as possible and to include the observable phenomena like wind, temperature gradients and ground effects with more sophisticated mathematical descriptions. But many of these developments with possible improvements on one side may cause severe problems in final applications. These interdependencies are shown and demonstrated with practical examples and recommendations are given how to treat the transformation of theoretical concepts into generally applicable methodologies.

5:20

2pNSb10. Design and implementation of urban traffic noise mapping system. Tao Feng, Nan Li, Bin Liu (School of Material and Mechanical Engineering, Beijing Technology and Business University, Beijing 100048, China, fengt@th.btbu.edu.cn), and Wencheng Hu (State Environment Protection Engineering Center for Urban Noise & Vibration Control, Beijing 100054, China)

The noise mapping system is a best tool for environment noise controlling, as it can provide a graphical illustration of sound exposure levels in a region for the policy- maker. In urban construction planning, land-use definition and noise environment impact assessment, the noise map has been proved very useful for its efficiency. According to the application requirements of traffic noise prediction model, the CATNMP (Computer Aided Traffic Noise Mapping Platform) has been designed and analyzed in details. The key techniques of algorithm and procedures involved are studied and implemented. An open source topological software package of JTS is used for the development of CATNMP based on GIS data. The traffic noise prediction model is performed with the aid of the computer software system, and a noise map is drawn out. In order to improving the computation efficiency, the distributed computing technique has been implemented in the CATNMP. Finally, the road traffic noise levels in a demonstration area of Beijing city are predicted using the CATNMP and a noise map is drawn for illustration and evaluation of environmental noise. The result shows that the CATNMP can provide an efficient analytical tool for the environmental evaluation of traffic noise, and also technical bases for regional noise emission management.

5:40

2pNSb11. New town planning—noise mapping tools to facilitate evaluation of noise exposure on different traffic management concepts. Tsz Hin (Laurent) Cheung (AECOM, 15/F Grand Central Plaza, Tower 1, Shatin, N.T., Hong Kong, laurent.cheung@aecom.com)

In terms of development of modern city, there is always a great requirement on the long-term housing demand, in particular for the strategic New Development Areas (NDA) in New Territories (NT) of Hong Kong. A cross-district traveling for work is a common practice, and there is also a high percentage of Heavy Goods Vehicle (HGV) for logistic goods arrangement. In order to maximize the development potential and to guarantee a noise disturbance free ambient to the intake population, noise mapping tools is therefore selected as an evaluation tools in accordance with 2002/49/EC Directive. Traffic management concepts on (1) road against rail and (2) restricting of HGV to certain trunk roads were tested, and the pros and cons have been evaluated. The general 24-hour flow patterns of roads and rail have been discussed, and these flow natures would have great influence on the Lden and Lnight values. The difficulties and sensitive issues anticipated in the noise mapping exercises have been generally discussed which would be useful for reference.

6:00

2pNSb12. Study on noise mapping in developing Cities. Dan Xia, Yude Zhou, Wenying Zhu, and Weichen Zhang (Shanghai Academy of Environmental Sciences, 508 Qinzhou Rd., Shanghai 200233, xiad@saes.sh.cn)

Nowadays, noise mapping which can quite well reflect a long-term noise value of a particular region has been widely used in many cities. It also has a positive meaning and impact for those developed cities on noise management. But for many Chinese cities such as Shanghai, there are some differences. It is difficult to determine the long-term noise level because of more and more roads and the widely fluctuated traffic in these cities. Normal noise map does not work well any more. To solve this problem, noise mapping should be dynamic. Dynamic noise mapping updates based on the changes of geographic information and traffic flows, and shows the trend of noise level in the region. It is necessary to systematize the noise mapping, including geographic information system, acoustic model system, display system and check system. In this way, the maps can achieve a dynamic output according to the changes of input parameters. Dynamic noise mapping provides an information platform for noise management, which makes noise control measures more targeted and effective. Also, it helps environmental impact assessment in noise prediction. Meanwhile, it shows people noise level around them and enhances public awareness of noise abatement.

Session 2pPA

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

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Invited Papers

2:00

2pPA1. Correlation between oscillations and translational motion of droplets induced by surface acoustic waves: towards low power actuators. Baudoin Michael (IEMN, Avenue Poincaré, 59652 Villeneuve d'Ascq cedex, France, michael.baudoin@univ-lille1.fr), Brunet Philippe (10 rue Alice Domont et Léonie Duquet 75205 Paris cedex 13, France), Bou Matar Olivier, and Bussonière Adrien (IEMN, Avenue Poincaré, 59652 Villeneuve d'Ascq cedex, France)

Actuators based on SAW are versatile tools to achieve, atomization, jetting, oscillations, or inner mixing of droplets. Basically, acoustic waves are generated at the surface of a piezoelectric substrate by a transducer consisting of interdigitated fingers. The acoustic energy is then transmitted to the drop whose motion is induced by nonlinear acoustical effects (acoustic streaming, radiation pressure). However, an increase of the temperature occurs inside the drop due to the dissipation of acoustic energy. This could be harmful for the manipulation of biofluids. In this presentation, we will show that the amount of energy required to move or deform droplets can be drastically reduced by exciting them at their eigen frequencies. We will also exhibit and explain the correlation existing between the droplets amplitude of oscillation and their translation speed.

2:20

2pPA2. Microbubble acoustic surface cleaning. Michel Versluis (University of Twente, P.O. Box 217, 7500 AE Enschede, The Netherlands, m.versluis@utwente.nl)

Surface cleaning is accomplished by fluid mechanical forcing, often assisted by chemical and acoustical activation. Examples include the removal of nanoparticles from IC semiconductor substrates and the disruption of bacterial biofilms in dentistry. The presence of bubbles is known to greatly enhance the cleaning efficiency as it promotes mixing of the chemicals and it yields higher stresses through acoustic streaming and jetting following asymmetric collapse of the bubbles. With smaller nanofabricated structures and more delicate surgical therapies, there is a growing demand for precision cleaning with minimum damage to the surrounding media. Here we explore the concept of micro-machined cylindrical pits acting as cavitation nuclei for a continuous source of microbubbles, thereby localizing the cavitation phenomena and suppressing its inherent chaotic nature. The micropit bubble was found to be stable against dissolution, and the resonance behavior of surface mode vibrations of the cap was modeled with the unsteady Stokes equation combined with a Fourier-Bessel expansion. Above a pressure threshold, destabilization of the micromeniscus results in bubble pinch-off which was studied using high-speed imaging down to nanoseconds timescales. It was also found that the acoustic coupling and merging of cavitation clouds from neighboring micropits increasingly promote acoustic surface cleaning.

2:40

2pPA3. Using acoustic microstreaming to improve detection of gene expression in single cells. Tim Aumann, Wah Chin Boon, Mal Horne (Florey Neuroscience Institutes, The University of Melbourne, Parkville, Victoria 3010, Australia, taumann@unimelb.edu.au), Annika Axelsson, Anders Rosengren (Lund University Diabetes Centre, Lund University, Lund, Sweden), Karolina Petkovic-Duran, Yonggang Zhu (CSIRO Materials Science and Engineering, Highett, Victoria, Australia), and Richard Manasseh (Faculty of Engineering & Industrial Sciences, Swinburne University of Technology, Hawthorn, Victoria, Australia)

Functional heterogeneity among different cells of an organism (brain cell, heart cell etc.) is brought about by expression of different subsets of genes drawn from the same genetic template (DNA). Therefore, to understand the molecular basis of normal (healthy) and abnormal (diseased) cell behavior, one must measure gene expression. The typical procedure is to homogenize large numbers (>1,000) of cells together, isolate the first product of gene expression (RNA), reverse transcribe the RNA to a cDNA copy, then identify and quantify gene-specific cDNAs. Unfortunately, the gene expression profile obtained represents an average across all combined cells and cell behaviors rather than any particular cell. Ideally one would measure gene expression in a single cell; however the very small amount of labile RNA obtainable from a single cell mostly degrades before it can be measured. We have developed an acoustic microstreaming-

based device ("micromixer") which improves mixing of solutions within microliter volumes. Here we show application of "micromixing" to standard laboratory reverse transcription reactions significantly improves conversion of single-cell amounts of RNA to cDNA. Micromixing is therefore a low-cost and easy-to-use technology that is compatible with and can be added to standard laboratory hardware, software, reagents and expertise to enable better gene expression measurement from single cells.

3:00

2pPA4. Cavitation in confined spaces. Claus-Dieter Ohl (NTU & IHPC, cdohl@ntu.edu.sg), Sha Xiong, S. Roberto Gonzales Avila (NTU), Evert Klaseboer (IHPC), Ai Qun Liu (NTU), Tandiono Tandiono (IHPC), and Keita Ando (NTU)

Cavitation phenomena in real world are typically confined by one or more boundaries. Confining cavitation in small channels allows to study their interaction with cells, the formation of emulsions, and even sonochemical reactions in far greater detail as it would be possible in the bulk. However, it was expected that boundary layers will hinder bubble collapse more and more as the structure sizes are reduced. In this presentation the channel size is reduced even further, thus from microfluidic to nanofluidic channels. In microfluidic channels cavitation bubbles are generated with focused laser pulses and with acoustic waves. Acoustic cavitation in micrometer sized allows the formation of homogeneous emulsions, rapid rupture of cells (yeats and bacterias), and the dispersion of nanoparticles. While laser induced cavitation bubbles allow the study of bubble dynamics and bubble interaction in nanofluidic channels. In particular we will present experimental results on the dynamics of single bubbles and bubble-bubble interaction in nanochannels.

Contributed Papers

3:20

2pPA5. Acoustic microstreaming induced by pattern of Faraday waves on a bubble wall. Alexey Maksimov (Pacific Oceanological Institute, Far Eastern Branch of the Russian Academy of Sciences, 690041 Vladivostok, Russia, maksimov@poi.dvo.ru), Timothy Leighton (Institute of Sound and Vibration Research, University of Southampton, Southampton, SO17 1BJ, United Kingdom), and Peter Birkin (School of Chemistry, University of Southampton, Southampton, SO17 1BJ, United Kingdom)

Interest to acoustic microstreaming is supported by the variety of applications: micromixing, transferring lipid vesicles and large molecules in desired direction, and selective particle trapping which are essential to the success of lab-on-a-chip- devices. It is generally assumed that the main contributions to the streaming generated by a gas bubble come from the pulsation and translation modes. This study deals with the microstreaming which is induced when a bubble is driven acoustically in the regime of parametrically generation of Faraday waves. The greater wall displacement amplitude for $l > 1$modes means that their effect on the flux of species near the bubble wall can be much greater than that of the breathing mode. The modes with a fixed order l have a high degree of degeneracy equal to $(2l+1)$. The choice of which modes are chosen to grow to steady state, and which are selected out, determines the shape of the perturbation and hence the structures of the streaming flow. Basic features of pattern formation on the bubble wall have been recently derived by Maksimov & Leighton (doi:10.1098/rspa.2011.0366) which provides determination of streaming structures. These theoretical findings are used to interpret the experiments on selective particle trapping by oscillating microbubble.

3:40

2pPA6. 30 MHz driven fluid mixing in paper-based microfluidic systems. Amgad R. Rezk, Aisha Qi, James R. Friend (RMIT University 3001, amgad.rezk@monash.edu), Wai Ho Li (Monash University 3800), and Leslie Y. Yeo (RMIT University 3001)

Paper-based microfluidics have recently become a topic of interest due to the ease and low expense in fabrication, especially compared to traditional microfluidics fabrication materials, making them suitable for inexpensive diagnostics. We report a convective actuation mechanism in a simple paper-based microfluidic device using surface acoustic waves to drive mixing. Using a Y-channel structure patterned onto paper, the mixing induced by 30 MHz acoustic waves is shown to be consistent and rapid, overcoming several limitations associated with its capillary-driven passive mixing counterpart: the latter exhibits nonuniform and irreproducible mixing. Capillary-driven mixing offers only poor control, is strongly dependent on the paper's texture and fibre alignment, and permits backflow, all due to the scale of the fibres being significant in comparison to the microfluidics features. Using a novel hue-based colourimetric technique, the mixing speed and efficiency is computed. For the acoustically driven mixing the effects of changing the input power, channel tortuousity and fibre/flow alignment was assessed. The hue-based technique offers several advantages over grayscale pixel intensity analysis techniques in facilitating quantification without limitations on the colour and contrast of the samples, and can be used, for example, for quantification in on-chip immunochromatographic assays.

4:00–4:20 Break

4:20

2pPA7. UV epoxy bonding and enhanced SAW transmission for applications in acoustofluidic integration. Sean Langelier, Leslie Yeo, and James Friend (Monash University, Clayton 3800 VIC Australia, langelie@gmail.com)

Surface acoustic waves (SAWs) are highly attractive as a means of fluidic and particulate manipulation in Lab-on-a-Chip systems. However, standard acoustofluidic fabrication practices rely heavily on the use of elastomeric materials such as PDMS which are inherently ill-suited for conveyance of SAWs as they introduce severe acoustic attenuation. Here, we explore the use of a low-viscosity UV epoxy resin for room temperature bonding of piezoelectric SAW substrates with standard micromachined supersaturates such as Pyrex and Silicon. The bonding methodology is straightforward and allows for reliable production of sub-micron bonds that are capable of enduring the high surface strains and accelerations typical of SAW propagation. Transmission in devices prepared with this approach show as much as 20 dB of improvement compared to devices fabricated using the standard PDMS elastomer. The method is further used in the fabrication of closed-channel SAW pumping concepts for applications in microscale flow control.

4:40

2pPA8. Surface acoustic wave actuated miniaturized lab-on-a-disc (miniLOAD). Nick Glass, Richie Shilton, Peggy Chan (MNRL, Monash University, Nick.r.glass@gmail.com), James Friend (MCN and MNRL, Monash University), and Leslie Yeo (MNRL, Monash University)

Lab-on-a-chip systems offer much potential in next generation diagnostics. Miniaturizing laboratory processes can realize reductions in test times, sample size and cost. This allows for new possibilities such as real time and point of care diagnostic systems. However, lab-on-a-chip systems often require large laboratory scale equipment to drive flow for microfluidic processes. To overcome this challenge, a miniaturized centrifugal based microfluidic platform has been developed. Surface acoustic waves (SAW) are used to generate rotation in a fluid layer, which, in turn, drives the rotation of a disc. Initially, Mylar discs of 5 mm diameter were rotated to greater than 2000 rpm. SU8 discs of dimensions ranging from 250 microns to 10 mm were also actuation and speeds of the order of 10 000 rpm were recorded. The larger 10 mm discs were patterned with various microfluidic

structures through the use of photolithography. Common lab-on-a-chip processes are demonstrated including capillary valving, mixing and particle concentration. Currently, work is being carried out to do some basic biological assays on the miniaturized Lab-on-a-disc system.

5:00

2pPA9. Programmable manipulation of microbubbles in microfluidic systems with surface acoustic wave. Long Meng, Feiyan Cai, Lili Niu, Yanming Li, and Hairong Zheng (Shenzhen Institutes of Advanced Technology, Chinese Academy of Sciences, 1068 Xueyuan Ave., SZ University Town, Shenzhen 518055, China, long.meng@siat.ac.cn)

Programmable microfluidic systems for bioparticles manipulation that could enable automated biological analysis and diagnostics have the potential to revolutionize a wide range of applications in life sciences research, clinical diagnostics, and drug discovery. Several methods such as those based on hydrodynamic force, magnetic tweezers, and dielectrophoresis can manipulate the movement of the bioparticles successfully. However, these methods must be constructed the flow structure, magnetic wire, and electrode in the microchannel respectively, which not only limits the flexibility of manipulation but also makes the sample more likely to be contaminated. In this paper, a programmable microfluidic device is developed to manipulate the microbubbles along an arbitrary trajectory in the microchannel without any structure by introducing phase-shifts to a planar standing surface acoustic wave field. The microbubbles can be transported in a square trajectory, circular trajectory and even helix trajectory via adjusting the relative phase between the excitation signals controlled by LabVIEW program. The results reveal that the microbubbles can be transported over a predetermined distance continuously until they reach the targeted locations. This acoustic programmable microfluidic device would have potential applications on drug screening, cell studies, and other biomedical applications.

5:20

2pPA10. Formation of two-way Lamb waves and force potential wells using single conventional ultrasonic transducer and sheep horn shaped metal piece. Yiyang Wan (Department of Electrical, Computer & Systems Engineering, School of Engineering, Troy, NY 12180, wany2@rpi.edu), Siyuan Zhang, Yujin Zong, and Mingxi Wan (Xi'an Jiaotong University, No. 28, Xianning West Road, Xi'an, Shaanxi 710049, P.R. China)

A new method is introduced that uses single conventional ultrasonic transducer and a sheep horn shaped metal piece to induce two-way Lamb waves, and furthermore collects leaky Lamb waves to form two force potential wells. The immersion acoustical measurements show that the two-way Lamb waves with adjustable amplitudes can be induced when a 1MHz transducer radiates ultrasonic waves on to the semicircle of a metal "sheep horn". Furthermore, the results obtained by hydrophone measurement and fluorescence microscope observation verify that the force potential well can be formed in the circle of metal "sheep horn" by the leaky Lamb wave of each way and possibly used to collect the submicro particles. This designed method is intended to provide assistance in limited space for microscope observing by collecting particles within low force potential well. All experiment results also indicate that radiation force and the gradient of force potential well produced by this leaky Lamb wave device is very small due to several reasons including low generation efficiency at semicircle tip and propagation attenuation of Lamb wave in metal piece, near field properties of leaky Lamb wave in liquid, and so on.

TUESDAY AFTERNOON, 15 MAY 2012 S423, 2:00 P.M. TO 6:00 P.M.

Session 2pPP

Psychological and Physiological Acoustics: Release from Masking in Listeners with Normal and Impaired Hearing

Ruth Litovsky, Cochair
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Liang Li, Cochair
liangli@pku.edu.cn

Invited Papers

2:00

2pPP1. Effects of auditory priming on speech recognition in combination with spatial and fluctuating masker benefits. Richard Freyman (University of Massachusetts, Department of Communication Disorders, 358 N. Pleasant St., Amherst, MA 01003, rlf@comdis.umass.edu)

This presentation will discuss research on three individual causes of release from masking in speech recognition as well as the interactions among them. Spatial differences between target and masker produce release from masking through head shadow effects, binaural processing of interaural differences, and, under some specific circumstances, perceived spatial differences between target and masker. A second frequently-studied type of masking release occurs when fluctuations in a masker's amplitude envelope over time create temporal epochs in which speech-to-masker ratios are very favorable. The efficiency of masking is also reduced when a listener's prior knowledge of message content reduces uncertainty about what s/he will hear, a process often called auditory priming. This presentation will describe a series of recent experiments that consider factors that affect auditory priming and fluctuating masker benefit, individually and when combined with spatial cues. [Work supported by NIDCD DC01625.]

2:20

2pPP2. Effect of degraded binaural cues in cochlear implant users and normal hearing listeners on spatial release from masking. Ruth Litovsky, Matthew Goupell, Alan Kan, Sara Misurelli, and Corey Stoelb (University of Wisconsin, Madison, 53705, litovsky@waisman.wisc.edu)

Bilateral cochlear implants (BiCIs) are being provided to a growing number of individuals with bilateral severe-profound hearing loss and are becoming standard in many clinics worldwide, to restore spatial hearing skills and improve speech understanding in noisy environments. While patients generally perform better with BiCIs, their performance is significantly worse than that of normal hearing (NH) listeners. Spatial release from masking (SRM) in NH listeners depends on monaural (head shadow) and binaural cues. In BiCI users, SRM appears to be primarily due to head shadow; however, binaural-mediated SRM is weak or absent. Two factors are most likely responsible for this. First, bilateral CI processors are not coordinated, rendering binaural cues weak, absent or inconsistent. Second, patients' history with auditory deprivation likely results in poor neural survival at numerous cochlear locations. Our studies suggest that signal processing tools can be applied to bilateral CI users to restore binaural sensitivity. In this talk, data will be presented from studies with adults and children, in free field and using binaural research processors. Results from studies on restoration of interaural level and timing differences to BiCI users will be discussed in the context of what is needed for binaural SRM restoration. Work supported by NIH-NIDCD (grants 5R01DC003083 and 5R01DC8365)

2:40

2pPP3. The benefits of bilateral and directionally selective auditory prostheses. John Culling (School of Psychology, Cardiff University, Tower Building, Park Place, Cardiff, CF10 3AT, U.K., CullingJ@cf.ac.uk), Sam Jelfs (2Philips Research Europe, High Tech Campus 36 (WO-p.076), 5656 AE Eindhoven, The Netherlands.), Alice Talbert (School of Psychology, Cardiff University, Tower Building, Park Place, Cardiff, CF10 3AT, U.K.), and Steven Backhouse (Princess of Wales Hospital, Coity Rd. Bridgend CF31 1RQ, U.K.)

Both binaural hearing and directional microphones can improve understanding of speech in background noise if the sources of the speech and noise are spatially separated. We used a model of spatial release from masking [Jelfs, et al. (2011). Hear Res. 275, 96-104.] to predict the benefits of bilateral prostheses, directional microphones and head orientation. The model predicts large benefits of each of these factors. Measurements using selected spatial configurations in both normally hearing listeners and unilateral cochlear implantees confirmed the model's predictions. The reception thresholds for bilateral implantees were inferred using mirror-image spatial configurations to be at least 18 dB better than unilateral implantees in certain situations. Expected effects of directional microphones and head orientation were assessed through modelling spatial release from masking in a virtual restaurant situation. The model predicted marked differences between different seating positions, but in most locations, both moderate head rotations and directional microphones offered substantial benefits. Use of directional microphones generally offered larger benefits than head rotation, but there was little benefit from their combination. The addition of reverberation elevated predicted thresholds and reduced all of these effects.

3:00

2pPP4. Possible implications of interaural mismatch in the place-of-stimulation on spatial release from masking in cochlear implant listeners. Alan Kan, Corey Stoelb (Binaural Hearing and Speech Laboratory, Waisman Center, University of Wisconsin – Madison, 1500 Highland Avenue Madison, WI 53705, ahkan@waisman.wisc.edu), Matthew Goupell (Department of Hearing and Speech Sciences, University of Maryland – College Park, College Park, MD 20742), and Ruth Litovsky (Binaural Hearing and Speech Laboratory, Waisman Center, University of Wisconsin – Madison, 1500 Highland Avenue Madison, WI 53705)

Recent research suggests that bilateral cochlear implant (CI) users have improved speech intelligibility in noisy environments compared to unilateral CI users. This may not be surprising given that in normal hearing (NH) listeners, binaural hearing allows them to localize sounds and this ability allows them to benefit from a spatial release from masking (SRM). However, SRM benefits are much less in CI users and vary amongst users. This may be due to the interaural mismatch in the place-of-stimulation across the ears in CI users and occurs due to differences in neural survival and electrode implantation depth across the ears, leading to different parts of the cochlea being excited by electrodes of the same number. Data will be presented that shows that with increasing interaural mismatch, CI users typically heard lateralized or multiple sounds. In some CI users, interaural mismatches greater than 3 mm leads to an inability to use binaural cues. These effects may impact a CI user's ability to obtain SRM. Preliminary results from a CI simulation study conducted to investigate the effect of interaural mismatch on SRM will also be presented. Work supported by NIH-NIDCD R01 DC003083

3:20

2pPP5. The effects of temporal envelope confusion on release of masking. Yingjiu Nie (Department of Speech-Language-Hearing Sciences, University of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, niex0008@umn.edu), Peggy Nelson, Evelyn Davies-Venn, and Adam Svec (Department of Speech-Language-Hearing Sciences, University of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455)

Widened auditory filters in hearing impaired (HI) listeners may force them to rely more on temporal envelope (TE) cues when listening to speech. We propose that reduced masking release in HI listeners may be partially due to the confusion of the TE's of the masker and target. The current study investigates HI listener's comprehension of low- or high-pass vocoded spondees in the presence of fluctuating and stationary background noise. The spectral relationship of the target and masker was systematically varied from greater to no spectral overlap; the TE's of the masker and target were varied in similarity along two aspects — amplitude-modulation rate and shape. Preliminary data have shown the TE confusion in some HI impaired listeners results in speech understanding scores that are poorer in the presence of fluctuating noise (at a rate of 4Hz) than when the stationary noise is present. On the other hand, another group of HI listeners has demonstrated masking release. The effect of TE confusion of speech-envelope -shaped noise for understanding vocoded spondees will be discussed. Work is supported by NIH DC008306 to PB Nelson

3:40

2pPP6. The neural basis for energetic and informational masking effects in speech perception. Sophie Scott, Samuel Evans, Carolyn McGettigan (ICN, 17 Queen Square, London WC1N 3AR, sophie.scott@ucl.AC.UK), and Stuart Rosen (SHAPS, Chandler House, Wakefield Street, London)

Functional imaging studies of speech perception have revealed extensive involvement of the dorsolateral temporal lobes in aspects of speech and voice processing. In the current study, fMRI was used as a functional imaging method to address the perception of speech in different masking conditions. Throughout the scanning experiment, subjects were directed to listen to a female talker with whom they had been familiarized previously. In addition to the acoustic noise generated by the scanner, different kinds of masking sound were also presented diotically, at SPL levels determined by pilot testing. The masking sounds were signal correlated noise, spectrally rotated speech, and a second female talker. The results show a network of activation in the bilateral temporal lobes, prefrontal cortex and parietal lobes that were commonly activated across all masking conditions. Controlling for the effects of energetic masking, informational masking gave rise exclusively to clusters of activity in bilateral STG, with peak level activations in left anterior-mid STG and posterior STS/STG and right primary auditory cortex. Post-test intelligibility measures were used to reveal greater activation in the left temporal lobe, which was associated with higher performance scores.

4:00–4:20 Break

4:20

2pPP7. Noise-induced increase in human auditory evoked fields. Claude Alain (Rotman Research Institute at Baycrest Centre, 3560 Bathurst Street, Toronto, ON M6A 2E1, Canada, calain@rotman-baycrest.on.ca)

Background noise is usually detrimental to auditory perception. However, psychophysical studies have shown that in some circumstances low levels of broadband noise may improve signal detection. Here, we measured auditory evoked fields (AEFs) while participants listened passively (no response required) to stimuli embedded in low or moderate levels of continuous Gaussian noise. As a control condition, the stimuli were also presented without background noise. The AEFs were modeled with a pair of dipoles in the superior temporal plane and the effects of noise on the amplitude and latency of the resulting source waveforms were examined. The results show that low-level background noise enhanced AEF amplitude. The effects of continuous low level Gaussian noise on AEF amplitudes were comparable for young and older adults. Possible explanations for the noise-induced increase in AEF amplitude are a lengthening of the temporal integration window and/or efferent feedback connections between the auditory cortex and lower auditory centers to enhance the signal-to-noise ratio.

4:40

2pPP8. Prediction of speech masking release for fluctuating interferers based on the envelope power signal-to-noise ratio. Søren Jørgensen and Torsten Dau (Center for Applied Hearing Research, Department of Electrical Engineering, Technical University of Denmark, Orsteds Plads, Building 352, DK-2800 Lyngby, sjor@elektro.dtu.dk)

The speech-based envelope power spectrum model (sEPSM) presented by Jø rgensen and Dau [(2011). J. Acoust. Soc. Am. **130**, 1475-1487] estimates the envelope signal-to-noise ratio (SNR_{env}) after modulation-frequency selective processing, which accurately predicts the speech intelligibility for normal-hearing listeners in conditions with additive stationary noise, reverberation, and nonlinear processing with spectral subtraction. The latter condition represents a case in which the standardized speech intelligibility index and speech transmission index fail. However, the sEPSM is limited to conditions with stationary interferers due to the long-term estimation of the envelope power and cannot account for the well known phenomenon of speech masking release. Here, a short-term version of the sEPSM is presented, estimating the envelope SNR in 10-ms time frames. Predictions obtained with the short-term sEPSM are compared to data from Kjems *et al*. [(2009). J. Acoust. Soc. Am. **126** (3), 1415-1426] where speech is mixed with four different interferers, including speech-shaped noise, bottle noise, car noise, and a highly non-stationary cafe noise. The model accounts well for the differences in intelligibility observed for the stationary and non-stationary interferers, demonstrating further that the envelope SNR is crucial for speech comprehension.

5:00

2pPP9. Fine-structure storage, correlation computation, attribute capture, perceptual integration, and perceived spatial separation: an auditory chain in the reverberant environment. Liang Li (Department of Psychology, Peking University, Beijing 100871, China, liangli@pku.edu.cn)

In a reverberant environment with multiple-people talking, listeners are still able to recognize attended speech to a remarkable degree. The tolerance to disruptive stimuli largely depends on perceptual integration between direct and reflection waves of the target speech, because the perceptual integration facilitates the listener's selective attention to the target. In this presentation I show that under simulated noisy, reverberant conditions, recognition of target speech is not only dependent on the direct-reflection perceptual integration, but also functionally related to the lower-level ability to briefly store low-frequency acoustic fine-structure signals (i.e., auditory primitive memory). It is suggested that the auditory chain from the fast-fading primitive memory to correlation computation, reflection-attribute capture, lead-lag integration, perceived spatial separation between sources, and attention facilitation represents the feature-and-spatial parallel-processing strategy of the auditory system for dealing with input "flooding". Also, there is an age-related decline in both the auditory primitive memory and the perceptual integration-induced release of speech from masking.

5:20

2pPP10. Age-differences in the time course of stream segregation in informational masking of speech by speech. Bruce A. Schneider, Payam Ezzatian (University of Toronto Mississauga, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, bruce.schneider@utoronto.ca), Liang Li (Peking University, Beijing 100871, China), and M Kathleen Pichora-Fuller (University of Toronto Mississauga, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada)

Ezzatian et al. (*Lang. Cognitive Processes*, 2011) determined thresholds for detecting three target-words(target-words in italics) in semantically-anomalous but syntactically-correct sentences (e.g., “A *rose* could *paint* a *fish*”) masked by either speech-spectrum noise or by two other females talkers. When both masker and target originated from the same loudspeaker, speech recognition was independent of word position for the noise masker, but improved as a function of word position for the speech masker, suggesting that informational masking prolongs the time it takes to perceptually segregate the talker from the competing speech. Spatially separating the masker from the target sentence removed this word-position effect, as did vocoding the masker, presumably because both operations permitted more rapid segregation of the target from competing speech. The older adults in this study displayed the same pattern of results as the younger adults in Ezzatian et al. when both target and masker were presented over the same loudspeaker. However, in older adults, the word position effect found for the two-talker masker remained after either spatial separation or vocoding. This suggests that the segregation of speech from speech is more “sluggish” in older than in younger adults. Supported by Canadian Institutes of Health Research.

5:40

2pPP11. Speech in noise and ease of language understanding: when and how working memory capacity plays a role. Jerker Rönnberg (Linnaeus Centre HEAD, Swedish Institute for Disability Research, Department of Behavioural Sciences and Learning, Linköping University, Sweden, jerker.ronnberg@liu.se), Patrik Sörqvist (Linnaeus Centre HEAD, Swedish Institute for Disability Research, Department of Building, Energy and Environmental Engineering, University of Gävle, Gävle, Sweden), Örjan Dahlström, Mary Rudner (Linnaeus Centre HEAD, Swedish Institute for Disability Research, Department of Behavioural Sciences and Learning, Linköping University, Sweden), Ingrid Johnsrude (Linnaeus Centre HEAD, Swedish Institute for Disability Research, Department of Psychology, Queen’s University, Kingston, Ontario, Canada), and Stefan Stenfelt (Linnaeus Centre HEAD, Swedish Institute for Disability Research, Department of Clinical and Experimental Medicine, Linköping University, Sweden)

A working memory based model for Ease of Language Understanding (ELU) has been developed (Rönnberg, 2003; Rönnberg et al., 2008; Rönnberg et al., 2011). It predicts that speech understanding in adverse, mismatching noise conditions is dependent on explicit processing resources such as working memory capacity (WMC). This presentation will examine the details of this prediction by addressing some recent data on (1) how brainstem responses are modulated by working memory load and WMC, (2) how cortical correlates of speech understanding in noise are modulated by WMC, and (3) how WMC determines episodic long-term memory for spoken discourse masked by speech.

TUESDAY AFTERNOON, 15 MAY 2012 S222, 2:00 P.M. TO 7:00 P.M.

Session 2pSA

Structural Acoustics and Vibration and Noise: Machinery Noise and Vibration II

Zhuang Li, Cochair
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Wilson Ho, Cochair
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Invited Paper

2:00

2pSA1. A method to determine the road traffic flows for installing the noise barriers along a surface and viaduct combined highway. Jiping Zhang (Zhejiang Research & Design Institute of Environmental Protection, 109 Tian Mu Shan Road, Hangzhou 310007, China; State Key Lab. of Subtropical Building Science, South China University of Technology, China, jpzhang@mail.hz.zj.cn), Paul D. Schomer (Schomer and Associates, Inc., 2117 Robert Drive Champaign, IL 61821), Qun LI, Fei Chen (Zhejiang Hangzhou-Anhui Expressway Co., Ltd, Hangzhou 310004, China), Xilu Zhou (Hangzhou Traffic Police Detachment of Public Security Bureau, Hangzhou 310014, China), and Xioulong Han (Hangzhou Hanks noise Control Technology Co., Ltd)

A highway composed of surface and elevated sections opened with very little traffic and without noise barriers along the elevated sections to help meet the relatively stringent noise control targets. However, some noise control measures that were added to the basic design included increased height to the guard barrier, low noise road surface pavement, and green buffer zones. Now authorities wish to

determine the traffic flow conditions for which noise barriers would be required on the elevated sections. Of course this result can be determined by iteratively stepping through the variables, but this would take great effort. This paper presents a semi-theoretical semi-empirical method that simplifies the problem. Let K be the difference between noise from surface and elevated sections of the road. The noise level is calculated for the surface sections including such factors as speed, percent buses, percent of traffic at night, and other noise related factors from transportation engineering. Then, the noise level of the elevated sections is estimated from the levels calculated for the surface sections added to K. This estimated level is compared to the noise limit, and from this one can establish the traffic flow which would require noise barriers on the elevated sections.

Contributed Papers

2:20

2pSA2. Vibration analysis of beams with arbitrary elastic boundary conditions excited by a moving mass. Binglin Lv, Wanyou Li, Haijun Zhou, and JIngtao Du (College of Power and Energy Engineering, Habin Engineering University, 150001, lvbinglin@yahoo.com.cn)

In this paper, one newly developed method named the Improved Fourier Series method is applied to the vibration of a beam elastically supported at the both end excited by a moving mass. The flexural displacement of the beam is supposed to be one set of Fourier Series coupled with several appended terms. Based on the energy principle, the mass and stiffness matrix of the beam system are obtained. The mass is added to the model with its gravitation treated as the excitation to the dynamic system. In the end, the effect of the moving mass to the vibration of the beam is analyzed.

2:40

2pSA3. A new developed method for the vibration analysis of a beam with variable cross section. Gang Wang (No. 145, Nantong Street, Harbin Engineering University, wanggang_16@yahoo.cn), Wanyou Li, Wenlong Li, and Binglin Lv

In this paper, one method which combined the Improved Fourier series method and Differential Quadrature method (DQM) is firstly introduced to analyze the vibration of a beam with variable cross section. For DQM, the weighting coefficients is traditionally obtained by Lagrange interpolation, which would lead the equation system to be ill-conditioned when points are equidistantly chosen. In this paper, Fourier series with several auxiliary functions is applied to obtain the weighting coefficients which would converge at a fast speed and the more points included, the more accurate would be the results. Application of Fourier series to the DQM would avoid the appearance of the ill-conditioned equation system when the points are chosen equidistantly.

3:00

2pSA4. Nonlinear interactions as trigger for chaotic vibrations in a simplified brake system. Sebastian Oberst and Joseph Lai (UNSW Canberra (ADFA), SEIT, Canberra ACT 2600, Australia, s.oberst@adfa.edu.au)

In automotive disc-brake squeal, much of the focus in the past two decades has been directed to the analysis of a brake system's vibration response in the frequency domain using the complex eigenvalue analysis (CEA) to predict the number of unstable vibration modes. Unfortunately, it is well known that not all predicted unstable vibration modes will lead to squeal and the magnitude of negative damping does not always indicate squeal propensity. In the complex eigenvalue approach, only linearised equilibrium is analysed and non-linear behaviour of the brake system is not taken into account. On the other hand, non-linear (transient) time-domain analysis simulates the dynamic behaviour closer to a real brake system, but is rarely applied as it is computationally expensive and frequency domain analyses are very popular in industry practice. In this paper, a simplified brake system in the form of an isotropic pad-on-disc system is considered. Specifically, the pad motion and a related instability are investigated using a nonlinear finite element time-domain analysis (ABAQUS 6.8-4). The complex eigenvalue method fails to detect in-plane acting pad-mode instabilities which initiate in-plane intermittent out-of-plane impulsive excitation. This excitation leads to intermittent chaotic behaviour of the pad and quasi-periodic out-of-plane disc vibration. If the pad behaves in a turbulent fashion, the disc's out-of-plane motion shows toroidal chaos. This chaotic disc vibration could be the cause of the instantaneous brake squeal described in the literature.

3:20

2pSA5. Active vibration control experiments of a large flexible vibration isolation structure. Liubin Zhou, Tiejun Yang, Hui Shi, Wanpeng Yuan, and Zhigang Liu (Research Institute of Power Engineering Technology, Harbin Engineering University, Heilongjiang, Harbin 150001, China, liubin.zhou@gmail.com)

An active control system based on Filtered-x LMS algorithm with off-line secondary path modeling is proposed and applied in active vibration control of a large flexible vibration isolation structure, which is an isolation system attached on a large flexible structure supported by twenty-six air springs. Details of the large flexible vibration isolation system, the features of the active system, and experimental investigation including the single-input single-output and multi-input multi-output experiments are presented in this paper. The experimental results show that vertical vibration levels of error sensor positions on the flexible foundation can be reduced greatly. At last some results are presented and discussed.

3:40

2pSA6. Investigation of active vibration isolation with inertial actuators. Hui Shi (Research Institute of Power Engineering Technology, Harbin Engineering University, Harbin 150001, shmily_hui@msn.com)

A single stage active vibration isolation system with inertial actuators is presented in this paper. The system frequency responses were obtained from the finite element model of the vibration isolation system. By using curve fitting method, the transfer functions of primary paths and secondary paths were derived, which were defined as transfer functions from excitation forces to error sensors and from secondary forces to error sensors respectively. Then multi-input multi-output active vibration control simulation and experiment were conducted in the Lab. The results show that good vibration attenuations were achieved. Some conclusions were given and discussed at last. Key words: vibration isolation; inertial actuator; multi-input multi-output; active control

4:00–4:20 Break

4:20

2pSA7. Experimental analysis on barrel zoom module of digital camera for noise source identification and noise reduction. Un-Chang Jeong, Ji-Hyun Yoon, Jae-Eun Jeong (Hanyang Univ., Unchang.jeong@gmail.com), Jung-Youn Lee (Kyonggi Univ.), and Jae-Eung Oh (Hanyang Univ.)

Noise of digital camera has been noticeable to its users. Particularly, noise of a barrel assembly module in zoom in/zoom out operation is recorded while taking a video. Reduction of barrel noise becomes crucial but there are not many studies on noise of digital camera due to its short history of use. In this study, experiment-based analyses are implemented to identify sources of noise and vibration because of complexity and compactness of the barrel system. Output noise is acquired in various operation conditions using synchronization for spectral analysis. Noise sources of a barrel assembly in zoom operating are first identified by the comparison with gear frequency analysis and then correlation analysis between noise and vibration is applied to confirm the generation path of noise. Analysis on noise transfer characteristic of zoom module is also carried out in order to identify the most contributing components. One of possible countermeasures of noise in zoom operating is investigated by an experimental approach

4:40

2pSA8. Evaluation of vehicle seat rattle noise using coherence function technique. Jin-su Park, Jae-Eun Jeong, Jong-Won Lee, In-Hyung Yang, and Jae-Eung Oh (Hanyang University, Mechanical Engineering 17 Haengdang-Dong, Seongdong-Gu, Seoul 133-791, Korea, jeoh@hanyang.ac.kr)

Recently, customers have been concerned about vehicle NVH depending on vehicle designing and manufacturing technologies development. In choosing vehicle, vehicle NVH is becoming the most important factor to customers. Especially, a seat is the final stage of vibration transfer path to passengers from all sources of vibration like engine, transmission and etc. And seat is the nearest component from driver's ear. For this reason, seat is the most important component that directly related to ride comfort for passengers. And driver can be influenced sensitively by BSR caused by seat. Thus, evaluating the vibration characteristics of vehicle seat and BSR caused by vehicle seat is necessary to reduce the seat BSR. The rattle noise occurred from seat has evaluated through sound source visualization and multi-dimensional spectral analysis - coherence function technique in this paper. Vibration characteristics of the seat has verified through modal test.

5:00

2pSA9. Acoustic radiation of stiffened cylinder with double shells. Libo Qi (China Ship Scientific Research Center, 222, Shangshui East Road, Binhu District, Wuxi, Jiangsu, China, qilibo1984@163.com)

This paper is subject to respectively calculate the acoustic radiation curves of stiffened cylinders with single shell and double shells under the two conditions of longitudinal unit force excitation and vertical unit force excitation. The models are applied both sides of the free boundary conditions. Three-dimensional hydroelastic theory is applied to solve the fluid-structure interaction problems. Through analyzing the peak values and their corresponded characteristic modes to obtain the structure key models that mainly contribute to the acoustic radiation. As well to instruct the acoustic optimization design of stiffened structures by comparing the frequency spectrum of acoustic radiation of the two structures. Keywords: acoustic radiation; stiffened cylinders; single shell; double shells

5:20

2pSA10. Tuned viscoelastic damper for hollow shaft's torsional vibration control. Yong Duan and Wenwei Wu (China ship scientific research center, Mailbox 116 Wuxi, 214082, China, yduan.detec@nuaa.edu.cn)

This paper presents a piece of work on hollow shaft vibration control using viscoelastic materials in torsional directions. Columned viscoelastic damper is designed as a capsule containing viscoelastic material and elastic material, the outer layer of the capsule is viscoelastic material, the inner layer is elastic material. The damper is mounted somehow into the void of the shaft. When the shaft is vibrating, the damper can provide damping for the passive control of torsional vibrations. The viscoelastic damper is designed based on the principle of dynamic absorber in theoretically. When the damper has been designed, the frequency response functions of the damping system include damper and shaft can be calculated by solving the dynamic equations, then, the damping effect of the viscoelastic damper on the hollow shaft can be obtained. It's shown that the damping effect depends on the loss factor of the viscoelastic material, and the damper can provide large damping for the hollow shaft when the loss factor of the viscoelastic material achieves its optimal value. And these conclusions can also be obtained by ANSYS software.

5:40

2pSA11. Noise and vibration from railway inclined turnout. Wilson Ho, Cheuk Yin Chan (Wilson Acoustics Limited, Unit 616, Technology Park, 18 On Lai Street, Shatin, Hong Kong, China, who@wal.hk), Tim Chan, and Richard Kwan (MTR Corporation, MTR Headquarters Building, Telford Plaza, Kowloon Bay, Hong Kong)

Railway noise and vibration control at turnouts and the associated groundborne noise and viaduct re-radiated noise are major environmental concerns for railway projects in urban area. For new railway lines in Hong Kong, inclined turnouts are used in mainlines (instead of vertical turnout) for smooth load transfer at turnout, and reduction of noise and vibration from the turnout. According to the practice in the world (Calculation of Railway Noise by Dept of Transport in UK, Transit Noise and Vibration Impact Assessment by Federal Transit Administration in USA, etc.), noise and vibration levels from turnout would be higher than that from normal railway track in the range of 2.5dB to 10dB. According to the practice in Hong Kong, +7dB turnout correction is adopted for airborne noise calculation, and +5dB and +10dB turnout corrections are adopted for groundborne noise calculation for inclined and vertical turnouts respectively. This paper presents noise and vibration measurement results at various setback disances from turnouts and discusses the turnout corrections for various operation conditions.

6:00

2pSA12. Vibration isolation performance of isolated slab track. Wilson Ho, Banting Wong, Isaac Chu (Wilson Acoustics Limited, Unit 616, Technology Park, 18 On Lai Street, Shatin, Hong Kong, China, who@wal.hk), Calvin Kong, and Richard Kwan (MTR Corporation, MTR Headquarters Building, Telford Plaza, Kowloon Bay, Hong Kong)

Isolated slab tracks (IST) have been used in a number of existing railway lines in Hong Kong to reduce viaduct re-radiated noise and groundborne noise. The design has adopted both profiled-type and full-space elastomeric mat, placed underneath the concrete track slab to reduce trackform vibration transmission to the supporting structure. The profiled-type mat was installed on the Tseung Kwan O line while a full-space mat on the Lok Ma Chau line in Hong Kong. This paper presents the latest vibration measurement results of the full-space elastomeric mat IST and compares the vibration isolation performance with other resilient trackform types. The advantages and disadvantages of using profiled mat IST and full-space mat IST are discussed from the view angle of trackform engineers and acousticians.

6:20

2pSA13. Noise and vibration control by rail dampers. Wilson Ho, Banting Wong (Wilson Acoustics Limited, Unit 616, Technology Park, 18 On Lai Street, Shatin, Hong Kong, China, who@wal.hk), David England, and Alson Pang (MTR Corporation, MTR Headquarters Building, Telford Plaza, Kowloon Bay, Hong Kong)

MTR Corporation and Wilson Acoustic Limited launched a project to investigate the effect of rail dampers on noise reduction and rail corrugation growth in an operational metro railway at a 300m radius curved track. Corrugation was measured by Corrugation Analysis Trolley (CAT) every month throughout a 6-month grinding cycle. Corrugation of wavelength 50-80mm was observed at the low rail, corresponding to 250-400Hz at train speed of 70km/hr. No corrugation was observed at the high rail. Corrugation growth rate was reduced by 45% after installation of rail dampers. In contrary to rail corrugation models at tangent track which predict exponential growth with time, corrugation growth at the test site is approximately linear. Corrugation at the test site was thought to be the combined effect of low vertical response and high lateral response at 250–400Hz, resulting in periodic fluctuation of normal contact force, minimal normal contact force is accompanied by lateral sliding causing frictional wear. Rail dampers have no apparent effect on the vertical response at anti-resonance 250–400Hz, but significantly reduce the lateral response. Measurement results and literature reviews suggest that using rail dampers to decrease lateral response could reduce growth of short pitched corrugation at sharp curves.

6:40

2pSA14. Influential parameters in system damping performance of viscoelastically damped plates. André Verstappen and John Pearse (University of Canterbury, 69 Creyke Rd, Ilam, Christchurch, NZ, andre.verstappen@pg.canterbury.ac.nz)

Vibration in metal panels is often undesirable as it can lead to high noise levels, reduced equipment lifetime and physical discomfort. For this reason, vibration and acoustic performance is an important aspect to consider during the design stage. Unconstrained viscoelastic coatings can provide a simple solution for damping vibrations of metal panels. It is well known that the damping performance of viscoelastic materials is dependant on the ambient

temperature and frequency of excitation. It is also known that system damping performance is affected by plate boundary conditions, substrate material and the thickness ratio between the substrate and damping material. A fractional factorial experimental design was employed to determine the interactions of these variables and their relative influence on the system damping performance. Fully clamped and simply supported aluminium and steel plates of two sizes were examined. Temperatures and damping layer thickness ratios typical of normal use were used. Performance was measured by the system loss factor which was obtained using a sound intensity method [1]. 1. Lim, M. K., A sound intensity technique for determining structural damping of a panel exposed to noise. Applied Acoustics, 1991. 32(Copyright 1991, IEE): p. 311-19.

TUESDAY AFTERNOON, 15 MAY 2012 S428, 2:00 P.M. TO 5:00 P.M.

Session 2pSC

Speech Communication: Articulation and Acoustics in Typical and Atypical Speakers (Poster Session)

Charles B. Chang, Chair
cbchang@umd.edu

Contributed Papers

All posters will be on display from 2:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:00 p.m. to 3:20 p.m. and contributors of even-numbered papers will be at their posters from 3:20 p.m. to 5:00 p.m.

2pSC1. Producing whole speech events: differential facial stiffness across the labial stops. Bryan Gick, Naomi Francis, Chenhao Chiu (Department of Linguistics, University of British Columbia, Vancouver, BC V6T1Z4, Canada, gick@interchange.ubc.ca), Ian Stavness, and Sidney Fels (Department of Electrical and Computer Engineering, University of British Columbia, Vancouver, BC V6T1Z4, Canada)

It has long been assumed that the labial stops (e.g., [p], [b], [m]) are articulatorily identical. However, recent evidence [Abel et al. ISSP, 2011] shows that these labial stops are visually distinct. This distinction could result from differential passive responses to air pressure differences across the stops, or could reflect an active difference in facial muscle activation. An active difference would challenge the simplicity of unidimensional physical target-based speech production models. A pilot study was conducted in which air was blown simultaneously into a speaker's mouth and nose just at the onset of /p/ and /m/ closures. Preliminary results show displacement of the cheeks and lips at /m/ onset, but not at /p/ onset. These results indicate different initial muscular settings for these sounds, presumably to stiffen the face in anticipation of the increased oral air pressure for /p/. Biomechanical simulation using ArtiSynth (www.artisynth.org) confirms that this outcome is consistent with activation of distinct muscle sets across the stops. These findings suggest that speech tasks include aspects of the "whole" event, including aerodynamics, rather than being determined by unimodal spatial targeting.

2pSC2. Vowel onset as a rhythmic marker in Japanese single and geminate stop distinctions. Yukari Hirata and Carmen Lin (13 Oak Dr., EALL, Hamilton, NY 13346, U.S.A., yhirata@colgate.edu)

Hirata and Whiton (2005) found that the durational ratio of stop closure to a disyllabic word was the best parameter to distinguish singleton and geminate stops in spoken Japanese. However, given that vowel onset has perceptual and psychoacoustic importance as a temporal marker (Kato et al., 2003), the best unit may be from the first vowel onset of the target word to that of the following word in a carrier sentence (= Vword). In this study, 36 target disyllabic pairs including singletons and geminates (e.g. [buka] and [buk:a]) were embedded in the sentence [sokowa ____to jomimasu], spoken by four native Japanese speakers at three rates (Hirata and Whiton, 2005). The duration of Vword (e.g. the beginning of 'u' in [buka] to the beginning of 'o' in [to]) and of Vc(c) (e.g. 'uk' from [buka] and 'ukk' from [buk:a]) were measured, and the Vc(c)/Vword ratio was calculated. We examined how accurately this ratio classified singleton and geminate tokens across all rates and speakers. The classification accuracy was found to be 97.3-98.0%. This indicates that the duration of Vc(c) relative to Vword is a stable parameter for quantity distinction, and that the Vc(c) unit may constitute a meaningful rhythmic marker in production.

2pSC3. Speech rate influences categorical variation of English flaps and taps during normal speech. Donald Derrick (MARCS Auditory Laboratories, Bankstown Campus, Bldg 1, MARCS, Locked Bag 1797, Penrith, NSW 2751, Australia, donald.derrick@gmail.com), and Bryan Gick (University of British Columbia)

This research demonstrates that North American English speakers with faster syllable iterance rates are more likely to produce reduced /t/ and /d/ as taps, which have two directions of motion, than flaps, which have one. Using B/M mode ultrasound to capture tongue motion direction, coupled acoustic recordings to capture syllable speed, we identified categorical variants of flaps and taps produced by 18 speakers and compared the average likelihood of each variant in relation to the rate at which speakers can repeat 'ta'. Faster iterators produce more alveolar taps in single flap phrases (e.g. 'autumn), and double flap phrases (e.g. 'edit a'). Alveolar taps require more changes in direction of tongue motion, and are often produced to avoid articulatory conflicts with neighbouring vowels. The results support an argument that individual variation in flap/tap production is partly due to a motor skill constraint limiting cycles of tongue motion, weighed against avoiding articulatory conflict. Currently a second experiment, testing whether changing actual speech rates will generate categorical shifts in flap variant production, is in progress. Preliminary results are expected by the conference date.

2pSC4. Acoustic characteristics of coronal fricatives in Seoul Korean. Charles Chang (University of Maryland, College Park, Center for Advanced Study of Language, 7005 52nd Avenue, College Park, MD 20742, cbchang@umd.edu)

This study presents a detailed acoustic characterization of the contrast between the two voiceless coronal fricatives of Korean, variously described in the literature as a lenis/fortis or aspirated/fortis contrast. In utterance-initial position, the fricatives were found to differ in centroid frequency, frication duration, aspiration duration, following vowel duration, and several aspects of the following vowel onset, including intensity profile, spectral tilt, and F_1 onset. Between-fricative differences varied across vowel environments, and spectral differences in the vowel onset especially were more pronounced for /a/ than for /i, ɨ, u/. However, differences between the fricatives in f_0 onset consistently failed to be found. Taken together, the acoustic

data showed that the ‘non-fortis’ fricative resembles both the lenis stops and the aspirated stops of Korean in a principled way: properties related to articulatory tension are similar to those of the lenis stops, while properties related to glottal width are similar to those of the aspirated stops. Given this dual patterning, it is argued that the ‘non-fortis’ fricative is best characterized not in terms of the lenis or aspirated categories for stops, but in terms of a unique representation that is at once both lenis and aspirated.

2pSC5. Acoustic properties of Turkish voiceless fricatives. Esra Ertan (Anadolu Universitesi DİLKOM 26470 Eskisehir Turkey, esraertan@anadolu.edu.tr), and Handan Kopkallı Yavuz

The investigation of acoustic parameters which differentiate place of articulation for fricatives has been limited to few languages. This study investigates the acoustic properties of Turkish voiceless fricatives representing three different places of articulation, /f, s, ʃ/. For each fricative, spectral properties, duration, overall amplitude, F2 transition, and center of gravity were measured. Each fricative occurred in different vowel environments, in one and two word syllables and in different positions within a word. The participants were 6 (3 male, 3 female) native Turkish speakers. A total of 4320 tokens were analyzed. The findings showed that frequency range, spectral peak location, F2 transition and center of gravity are cues to place of articulation in Turkish. The findings showed that the frequency range is 1500-10500 Hz for /f/, 3500-10800 Hz for /s/, 1800-9400 Hz for /ʃ/. The spectral peak location measurements were; /s/ > /f/ > /ʃ/. The mean duration of fricatives were /ʃ/ >/s/ >/f/. The amplitude for fricatives are as follows; /ʃ/ >/s/> /f/. The onset of F2 transition were; /ʃ/ >/s/> /f/ and the center of gravity measurements were; /s/ > /ʃ/ > /f/.

2pSC6. Motion of apical and laminal /s/ in normal and post-glossectomy speakers. Rachel Reichard (University of Maryland SOD, Orthodontics, 650 W. Baltimore St., Baltimore, MD 21201, rachelhepfer@gmail.com), Maureen Stone, Jonghye Woo (University of Maryland SOD, NPS, 650 W. Baltimore St., Baltimore, MD 21201), Emi Z Murano (Johns Hopkins University School of Medicine, 720 Rutland Ave, Ross 528, O-HNS, Baltimore, MD 21205), and Jerry L Prince (Johns Hopkins University, ECE, 1800 N. Charles St., Baltimore, MD)

There are two ways to produce an ‘s’ in English: apical and laminal. They are almost identical perceptually and the reason for choosing one type is not well understood. This study questions whether one type is preferred in certain conditions, such as high vs low palate height or in post surgical tongue adaptation. This study examines palate height, and motion of critical and non-critical tongue regions in 8 normal speakers and 8 post-glossectomy patients who have had surgical resection of squamous cell carcinomas of the tongue. Speech was recorded with tagged-MRI and processed to track displacement and velocity of 2-D midsagittal tongue tissue points in the tongue tip and body during the utterance of the word “a souk”. Results indicate that subject category had a greater effect on s motion than palate height. Both critical and non-critical articulators in subjects with apical s used higher velocity and displacement on average than subjects with laminal s. Compared to patients with larger resections, patients with smaller resections had larger velocities and displacements in both tongue areas suggesting less debility. The single patient with flap reconstruction showed the highest velocity and greatest displacement, suggesting a different type of articulatory compensation. This research was supported in part by NIH R01 CA133015

2pSC7. Articulatory-acoustic relations in Cantonese vowel production. Wai-Sum Lee and Eric Zee (Dept. of CTL, City University of Hong Kong, 83 Tat Chee Avenue, Kowloon, Hong Kong, w.s.lee@cityu.edu.hk)

The study investigates the quantal nature of two sets of Cantonese point vowels, the long [i: u:] in CV open syllables and the medium-long [i u] in CVS closed syllables, where S = stop consonant, by determining the articulatory-acoustic relations in the production of the four Cantonese point vowels. It evaluates Stevens’ quantal theory (1972, 1989) by analyzing variability in articulation and acoustics of the four Cantonese point vowels. It also tests the sensitivity of formant frequencies of these point vowels to variation in linguo-palatal constriction location and degree of constriction along the vocal tract length. The investigation obtains variations in location and degree of linguo-palatal constriction along the vocal tract length during the point vowels, using EMMA AG500, and it relates the articulatory measurements to the corresponding vowel formant frequencies. Preliminary data show that the acoustic consequences of a shift in forward or backward direction of the constriction away from the target position for the point vowels are non-linear. The results appear to support the quantal nature of vowel production (Stevens, 1972, 1989). Furthermore, similar to the findings reported in Beckman, et al. (1995) formant frequencies are relatively less sensitive to variation in constriction location than degree of constriction.

2pSC8. A spectral analysis of the apical vowel in Yongding Hakka. Wai-Sum Lee and Eric Zee (Dept. of CTL, City University of Hong Kong, 83 Tat Chee Avenue, Kowloon, Hong Kong, w.s.lee@cityu.edu.hk)

The so-called apical vowels occur in a large number of Chinese dialects, including Yongding Hakka spoken in the southwest of Fujian Province in southeastern China. A main difference between apical vowel and dorsal vowels is in the fact that the former is a syllabic approximant with a linguo-palatal contact pattern of either laminal alveolar or apical postalveolar. Phonologically, both vowel types function as the syllable nucleus. The paper analyzes the spectral characteristics of the apical vowel [ɿ] in Yongding Hakka. Preliminary results show that the F-pattern of the apical vowel differs from that of the dorsal [i] substantially. The average F1, F2, and F3 values of five repetitions of [ɿ] from 10 male speakers are 500.9 Hz, 1323.4 Hz, and 2722.9 Hz, but 339.8 Hz, 2362.6 Hz, and 3269.6 Hz for [i]. Thus, a large difference is in F2. This is also true for female speakers of Yongding, in that the average F-values are 567.9 Hz, 1473.2 Hz, and 3242.6 Hz for [ɿ] and 405.4 Hz, 2362.1 Hz, and 3269.6 Hz for [i]. The spectral data for the apical vowel in Yongding Hakka will be compared with those for the apical vowels in other Chinese dialects, such as Beijing Mandarin.

2pSC9. Modeling Mandarin Tone 2 and Tone 3 from natural productions with variability. Tian (Christina) Zhao (University of Washington, ILABS, Box 357988, Seattle, WA 98195, zhaotc@uw.edu), Richard Wright (University of Washington, Box 354340, Linguistics, Seattle, WA 98195), and Patricia Kuhl (University of Washington, ILABS, Box 357988, Seattle, WA 98195)

Naturally produced speech is highly variable. Previous descriptions of lexical tone contours tend to be based on a single production or productions from a single speaker. Here we describe a method to model Mandarin Tone 2 and Tone 3 accounting for both intra- and inter-speaker variability. Five female speakers, born and raised in Beijing, were recorded reading 60 syllables both in Tone 2 and Tone 3 (real Chinese words) at two different speeds. The pitch contour of each production was first extracted using Praat. All the contours for Tone 2 and Tone 3 were then plotted in normalized time against frequency values. As expected, the normalized time vs. frequency plot for each tone exhibited large variability. Using the Matlab curve fitting toolbox, the 6th polynomial function was chosen to achieve the best goodness of fit with the current data. [Research supported by NSF.]

2pSC10. Effects of prosodic position on the production of Si-Xien Hakka tones at phrase level. Hsiu-min Yu (Language Center, Chung Hua University, 707, Sec.2, Wu Fu Rd., Hsinchu 30012, Taiwan, R.O.C. and Graduate Institute of Taiwan Languages and Language Education, National Hsinchu University of Education, Taiwan, kuo@chu.edu.tw), Chen-yu Chiang, Yih-ru Wang, and Sin-horng Chen (Institute of Communications Engineering, National Chiao Tung University, 1001 Univ. Rd., Hsinchu 300, Taiwan, ROC)

This study examined the effects of prosodic position on duration and F0 of the six tones in Si-Xien Hakka. Each of the tones was placed in the initial, medial, and final positions of a three-syllable phrase/clause, which constituted the first part of a sentence with the [$\delta\delta\delta$, $\delta\delta\delta$] structure. The results showed that F0 lowering and lengthening were found at domain final position. The final lengthening allowed the offset of the low-rising tones to

reach a higher F0 target, not attainable by those in other prosodic positions. Also in this final position the high level tones exhibited a real flap shape due to the greater coarticulation resistance caused by the following boundary breaks. Furthermore, the F0 decreasing rate of the falling tones was found to vary according to domain position: the F0 descended the fastest at domain medial position and the slowest at final position, except for the high checked tones, which showed the same rate at domain-initial and –final positions. This reveals that the vibration of the vocal folds, reflected by the F0 decreasing rate, started from the initial position, gradually speeded up to the highest rate at medial position and then slowed down till the pre-boundary domain-final one.

2pSC11. Influence of medialization depth, implant shape, and implant stiffness in medialization thyroplasty. Dinesh Chhetri (62-132 CHS, UCLA Medical Center, Los Angeles, CA 90095, dchhetri@mednet.ucla.edu), and Zhaoyan Zhang (31-24 Rehab center, 1000 Veteran Avenue, Los Angeles, CA 90095-1794.)

Medialization thyroplaty (MT) is a widely used treatment modality for vocal fold paralysis. The goal of MT is to improve glottal closure, especially as viewed from the superior surface, by surgically placing an implant in the paraglottic space. However, phonatory improvement from the MT procedure is variable. In this study, an excised larynx model was used to investigate the effects of the following ML implant parameters on laryngeal acoustics, aerodynamics, and vibration: medialization depth, medial surface shape of implant, and implant stiffness. Phonation threshold pressure and flow rate, H1-H2, spectral slope, and high-speed recordings of vocal fold vibration were measured as the implant parameters were varied. The relative importance of the implant parameters (depth, shape, and stiffness) to phonatory improvement and the clinical implications will be discussed.

2pSC12. Observations of phonation threshold pressure and fundamental frequency in the in vivo canine larynx using graded neuromuscular stimulation. David A. Berry, Dinesh K. Chhetri, and Juergen Neubauer (UCLA Head & Neck Surgery, 1000 Veteran Ave, Suite 31-24, Los Angeles, CA 90095-1794, daberry@ucla.edu)

Phonation threshold pressure is an objective measure of the relative ease of phonation. Previous studies of the in vivo canine larynx have reported values of phonation threshold pressure in the range of 3.5-5.5 kPa. Such studies are often cited to argue that canine and human vocal fold physiology may differ significantly, since threshold pressures in human phonation are often measured to be as low as 0.2-0.3 kPa. Similarly, the frequency range observed in canine phonation has generally been reported to be more limited than that found in human phonation. In this study, our hypothesis was that through the use of a newly-developed method of graded stimulation to both the superior laryngeal nerve and the individual branches of the recurrent laryngeal nerve, the ranges of phonation threshold pressure and fundamental frequency observed in the in vivo canine model would overlap significantly with that of human phonation. This hypothesis was confirmed. In particular, phonation threshold pressures were observed to be as low as 0.2-0.3 kPa, and the fundamental frequency range was observed to span nearly four octaves. Previous studies of in vivo canine phonation may have been limited by an inadvertent, exclusive focus on hyper-stimulation of the thyroarytenoid muscle.

2pSC13. Experimental and theoretical investigation of self-sustained vibration in a two-layer vocal fold model with left-right stiffness asymmetry. Zhaoyan Zhang and Trung Hieu Luu (UCLA School of Medicine, 31-24 Rehab Center, 1000 Veteran Ave., Los Angeles, CA 90095-1794, zyzhang@ucla.edu)

The vibratory characteristics of a self-oscillating two-layer vocal fold model with left-right asymmetry in body-layer stiffness were experimentally investigated, with the goal of better understanding the relative difference in vibration amplitude and phase between the two folds. Two regimes of distinct vibratory pattern were identified. In the first regime with extremely large stiffness mismatch (larger than a factor of ten), vocal fold vibration was dominated by the vibration of the soft fold only, while the other fold was enslaved to vibrate at the same frequency. The fundamental frequency was close to that of the soft fold in a symmetrical condition. In the second regime when the left-right stiffness mismatch was reduced, both folds vibrated with comparable amplitude. In this regime, either fold can exhibit a relatively larger amplitude but the stiff fold consistently led the soft fold in phase for all conditions. The experimental results were compared to predictions from a two-dimensional plane-strain vocal fold model, and qualitatively good agreement was obtained between experiment and simulation. The clinical implications of the results of this study are also discussed. [Work supported by NIH.]

2pSC14. Phonation in nine languages. Patricia Keating (Dept. Ling., UCLA, Los Angeles CA 90095-1543, keating@humnet.ucla.edu), Jianjing Kuang (Dept. Ling., UCLA), Christina Esposito (Dept. Ling., Macalester College), Marc Garellek (Dept. Ling., UCLA), and Sameer ud Dowla Khan (Cog. Ling. Psych. Sci., Brown University)

This study compares the phonations of 9 languages. Some of the languages use phonation types contrastively, independently of any pitch contrasts (Gujarati: modal, breathy; White Hmong and Black Miao: modal, breathy; Jalapa Mazatec: modal, breathy, creaky; Southern Yi, Bo, and Hani: tense, lax), while some use phonation as correlates of pitch contrasts (White Hmong: creaky low tone; Black Miao: creaky low tone and pressed high tone; Mandarin: creaky low and falling tones; Santiago Matatlán Zapotec and San Juan Guelavia Zapotec: creaky large-falling tone and breathy small-falling tone). Acoustic measures of phonation are compared for all 9 languages, and electroglottographic measures are compared for all but Mazatec. Multi-dimensional scaling of the production measures is then used to derive a lower-dimension phonation space, and the use of that production space by the different languages is compared. [Work supported by NSF]

2pSC15. Properties of the duration of English rhythm segments. Shizuka Nakamura (Waseda University/1-6-1 Nishi-Waseda, Shinjuku-ku, Tokyo 169-8050, Japan, shizuka@akane.waseda.jp)

The properties of the duration of English rhythm segments, which comprised a set of stressed and unstressed syllables, were investigated. The speech sounds of short sentences, each including three to five stressed syllables, spoken by 20 native speakers were used. The following definitions of the rhythm segment were adopted for comparative judgment: a stressed syllable and succeeding weak syllable sequence; preceding weak syllable sequence and a stressed syllable; a stressed syllable and a half of preceding and succeeding weak syllable sequences; and the interval between adjacent stressed syllables. The measurement based on the detailed acoustical analysis showed that the duration of each segment by the native speakers was distributed around a peak at about 0.7 second, which was considered to be a target rhythmic period. Some segments, which include the secondary stressed syllables, were distributed separately around a half of the target period. However, they could not be put closer to the main distribution, by exchanging the secondary stressed syllable for the primary one or weak syllable. For the universal description of the rhythm structure, the duration of the rhythm segment was approximated by a series of the target periods into which a half period was interpolated irregularly.

2pSC16. Properties of the duration of pauses in the recitation of a Japanese text. Shizuo Hiki (Waseda University, 1-104 Totsuka-machi, Shinjuku-ku, Tokyo 169-8050, Japan, hiki@waseda.jp), and Kuniko Kakita (Toyama Prefectural University, 5186 Kurokawa, Imizu-city, Toyama-Pref., 939-0398, Japan)

The present study investigated the properties of the duration of pauses in the recitation of a Japanese text. The text employed was the "panphonic" version of 'The North Wind and the Sun' created for the illustration of the IPA of Japanese (Tokyo dialect) consonants (S. Hiki, K. Kakita and H. Okada, Proc. 7th Int'l Cong. on Phonetic Sciences, Hong Kong, 2011, 871-873). The text consisted of eight sentences (294 syllables), was easy to read aloud, and could be read in one minute at normal speaking rate. In the text,

tonal segments were specified by boundary symbols. On the basis of these boundary symbols, as well as other factors such as breathing, possible pause locations were determined. A native speaker of Japanese read the text, inserting pauses at these locations. The speaker chose the pause duration that was most natural in each location. The relationship between the pause duration and the preceding utterance duration was analyzed. It was revealed that the accumulated duration of pause segments converges at about 40% of that of speech segments towards the end of one minute recitation. The present result provides a useful framework for extending the analysis to spontaneous speech.

2pSC17. Acoustic analysis of adults imitating infants: a cross-linguistic perspective. Johan Engdahl, Johannes Bjerva, Ellen Marklund, Emil Byström, and Francisco Lacerda (Department of Linguistics, Stockholm University, Universitetsvägen 10C, 10691, Stockholm, johan@ling.su.se)

The present study investigates adult imitations of infant vocalizations in a cross-linguistic perspective. Japanese-learning and Swedish-learning infants were recorded at ages 16-21 and 78-79 weeks. Vowel-like utterances (n=210) were selected from the recordings and presented to Japanese (n=3) and Swedish (n=3) adults. The adults were asked to imitate what they heard, simulating a spontaneous feedback situation between caregiver and infant. Formant data (F1 and F2) was extracted from all utterances and validated by comparing original and formant re-synthesized utterances. The data was normalized for fundamental frequency and time, and the accumulated spectral difference was calculated between each infant utterance and each imitation of that utterance. The mean spectral difference was calculated and compared, grouped by native language of infant and adult, as well as age of the infant. Preliminary results show smaller spectral difference in the imitations of older infants compared to imitations of the younger group, regardless of infant and adult native language. This may be explained by the increasing stability and more speech-like quality of infants' vocalizations as they grow older (and thus have been exposed to their native language for a longer period of time), making their utterances easier for adults to imitate.

2pSC18. The effects of jaw surgery on bilingual speech. Alan Mishler and Charles Chang (University of Maryland Center for Advanced Study of Language, 7005 52nd Ave, College Park, MD 20742, United States of America, amishler@umd.edu)

Surgery designed to correct misalignments of the jaw may lead to changes in the fine phonetic detail of speech segments. In some cases, these changes are unstable, with the acoustic parameters of post-surgical speech gradually reverting toward pre-surgical values, even when this causes speech to become less perceptually or acoustically "normal" (Lee et al, 2002; Niemi et al, 2006). This process might be driven primarily by tactile feedback, which could have differential effects on consonants and vowels, since consonants involve greater contact between articulators; alternatively, it might be driven primarily by auditory feedback, which might be expected to affect consonants and vowels to a similar degree. Previous studies have examined only vowels or consonants in isolation, and they have examined only a single language at a time. Here, a case study of a Korean-English bilingual's speech over the course of a year before and after surgery is presented. The subject's entire phoneme inventory in both languages is examined in order to determine the effects of surgery on different segment types and to determine to what extent the effects are language-specific.

2pSC19. Temporal structure in the speech of persons with Dementia of the Alzheimer's Type (DAT). Linda Carozza (St. John's University, 300 Howard Avenue, Staten Island, NY 10301, carozzal@stjohns.edu), Pamela Cantor, Margaret Quinn, and Fredericka Bell-Berti (St. John's University, Queens, NY 11439)

Although cognitive and language processes in dementia have been studied extensively, the question of motor speech degeneration in the course of dementing illness is a relatively unexplored area. The potential for early dissociation of motor functions of language at the level of speech production has not been explored; an interaction between motor speech and language production and perception changes should inform our understanding of the deterioration in dementia. In previous reports on two persons with DAT, we have shown inconsistent final lengthening and effects of syllable-final consonant voicing on vowel duration for one of the two speakers. We recorded one of the speakers again, and his speech was markedly slower. In this report, we expand our analysis to include three additional persons with mild-to-moderate DAT, from whom a series of 4-word phrases containing a target word occurring in phrase-medial or phrase final position was elicited. We will present the results of our analysis of final lengthening, compensatory shortening, and the effects of final consonant voicing on vowel duration.

2pSC20. Acoustic effects of neuromuscular electrical stimulation after a vocal loading task. Mary Gorham-Rowan (Valdosta State University, 1500 N Patterson St., Valdosta, GA 31698, mmgorhamrowan@valdosta.edu), Richard Morris (Florida State University, 127 Honors Way, Tallahassee, FL 32306-1200), and Linda Fowler (Georgia State University, 850 Suite 30 Pryor St. SW, Atlanta, GA 30303)

Recent studies have examined the application of neuromuscular electrical stimulation (NMES) to treat individuals with voice disorders. Appropriate protocols should be established for effective use of NMES. The purpose of this study was to use acoustic data to evaluate the effectiveness of NMES to the anterior neck in reducing vocal fatigue and muscle soreness following a prolonged reading task. Recordings were taken from two groups of talkers. Both groups completed a 45 minute oral reading task, then 11 subjects underwent a 15 minute NMES session, another group of 11 subjects completed the reading task only. Audio recordings were made before the reading task, after the reading task, and after the final 15 minutes. Acoustic measures of recordings included fundamental frequency, relative speech amplitude, cepstral level, H1-H2, H1-A1, and H1-A3. Preliminary results indicate increases in fundamental frequency and relative speech amplitude at the end of the reading task relative to pre- and post-session levels. The increases were greater among subjects receiving NMES.

2pSC21. Dynamical behavior in hemilarynx experiments with the false vocal folds attached. Michael Döllinger (University Hospital Erlangen, Medical School, Laboratory for Computational Medicine, Department for Phoniatrics and Pediatric Audiology, Bohlenplatz 21, 91054 Erlangen, Germany, michael.doellinger@uk-erlangen.de), and David A. Berry (The Laryngeal Dynamics Laboratory, Division of Head & Neck Surgery, UCLA School of Medicine, 31-24 Rehab Center, 1000 Veteran Ave., Los Angeles, CA 90095-1794)

Previously, changes in vibrational parameters were reported as a function of varying glottal conditions in hemilarynx experiments (Döllinger and Berry, J Acoust Soc Am, 130(4):2440). Now, dynamical changes will be reported with the false vocal folds still attached, and the results will be contrasted with the results of the previous study. For two different larynges in which the false vocal folds remained intact during sustained phonation, vibrational output was statistically analyzed as a function of glottal airflow, adduction forces, and elongation forces. Global parameters were computed, including empirical eigenfunctions, fundamental frequency, subglottal pressure, and sound intensity level. Similarly, local parameters were analysed at individual locations, including displacements, velocities and accelerations. The recordings were obtained using a digital high-speed camera. Increased airflow resulted in significant statistical changes in all parameter values except the empirical eigenfunctions. For increased adduction and lower elongation forces, the local parameters increased more than for the higher elongation forces. As compared to experiments without false vocal folds, these experiments exhibited more marked and consistent dynamical changes. This result suggests that the false vocal folds may have a stabilizing influence on the vibratory behavior of the vocal folds, perhaps due to nonlinear feedback with the supraglottal airflow.

Session 2pSP

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

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Invited Papers

2:00

2pSP1. Bayesian model-based filter design in acoustical signal processing applications. Jonathan Botts (Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180, bottsj@rpi.edu), and Ning Xiang (Rensselaer Polytechnic Institute, Graduate Program in Architectural Acoustics, 110 8th Street, Troy, New York, NY 12180)

Digital filter design finds applications across many disciplines. Acoustic applications include, but are not limited to, impedance boundary conditions in time-domain wave-based room-acoustics modeling, head-related transfer functions, and loudspeaker equalization. The process of filter design can be approached in a variety of ways. To design a non-standard filter, optimization methods may be used to solve for the coefficients that minimize the difference from a specified transfer function of a prescribed order. Interpreted as a model-based inference problem, a Bayesian framework, realized through the nested sampling algorithm, provides simultaneous optimal coefficient estimates and a filter order selection criterion. The selection criterion implicitly favors simpler models over more complex models, in this case lower-order filters over higher-order filters. Many acoustical applications are well suited to this type design, where low filter order is important, and the desired system response is unattainable with classical, closed-form design techniques. This paper formulates filter design problem as a problem in model-based inference. Then several illustrative examples from acoustical signal processing are used to demonstrate the flexibility of Bayesian digital filter design.

2:20

2pSP2. Modified S transform for time-frequency analysis of borehole flexural waves dispersion. Said Assous and Peter Elkington (Weatherford, Geoscience, East Leake, United Kingdom, LE6JX, said.assous@eu.weatherford.com)

Guided wave propagation usually exhibits dispersive behaviour. The time-varying spectral components of dispersive waves have been characterized effectively using the time-frequency approach; the short-time Fourier transform (STFT) and the continuous wavelet transform (CWT) are commonly used for this purpose. However, the resulting energy distributions suffer from poor resolution related to the uncertainty principle, and this complicates the allocation of energy to individual propagation modes especially when the dispersion curves of these modes are close to each other in the time-frequency domain (in which case the separation becomes a challenge). Therefore there is a need for high resolution time-frequency techniques. To meet this challenge we introduce a modified version of the S transform, a method which combines the advantage of the STFT and CWT but with greater resolution. An adaptive algorithm is presented which identifies frequency regions of interest related to different energy modes, and employs the modified S transform to a dispersion curve model to extract the proportional energy distribution of a specific mode from a multimode dispersive wave signal. The effectiveness of this approach is demonstrated on dispersive flexural waves obtained from an acoustic borehole logging tool.

2:40

2pSP3. Model-based geoacoustic inversion. Peter Gerstoft (Scripps Institution of Oceanography, UCSD, La Jolla, CA 92093, gerstoft@ucsd.edu)

The unknown geoacoustic environment often limits sonar-processing performance. Thus, there is a need for better estimates of the geoacoustic parameters. From the start of geoacoustic inversion more than two decades ago this has been model-based. Advanced propagation codes are based on ray tracing, wavenumber integration, normal modes or parabolic equations. The environment model is parameterized using ocean sound-speed and sediment sound-speed, density and attenuation. In some cases range-dependent models has been used. Initially classical least squares were used. But early on stochastic optimization methods as simulated annealing and genetic algorithms became popular. Understanding of uncertainties is important for the development of the methods. A major effort has been addressed using Bayesian methods and Markov chain Monte Carlo methods. Uncertainties are used to find the most important parameters and the obtained geoacoustic inversion uncertainties can be mapped into sonar performance assessment. In a typical experimental

scenario, the data arrives sequentially and new improved estimates can be obtained by updating previous estimates at significantly less computational expense than if each data observation were treated independently. Current efforts are focused on model-based geoacoustic inversion sequentially using sequential methods such as particle filters or Kalman-family of filters.

Contributed Papers

3:00

2pSP4. Time reversal mirror localization technology based on a high-resolution MVDR algorithm. Yan-yi Yuan (Science and Technology on Complex Ship Systems Simulation Laboratory, Beijing 100000, China, shengxueli@yahoo.com.cn), Xue-li Sheng (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001, China), Qing Ling (Science and Technology on Complex Ship Systems Simulation Laboratory, Beijing 100000, China), Jia Lu, Ye Bai, and Mei-ren Jiang (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001, China)

Minimum variance distortionless response (MVDR) algorithm has broken through the Rayleigh limit restrictions and makes higher resolutions possible. However, they are easily affected by complex underwater acoustic environments and are not sufficiently robust, so it is difficult to use them in real underwater systems. Time reversal mirror technology is a new method of localization, which may be effectively applied to target detection and localization. To improve its robustness, the MVDR algorithm is combined with time reversal mirroring for passive detection and location. This method makes full use of underwater multipath channel information and reduces the effects of complex underwater environments. The performance of time-reversal MVDR that is based on fixed diagonal loading and robust capon beamforming (RCB) was also studied. Simulations, water tank experiment and sea trial proved it is feasible and practical Keywords: MVDR; time reversal mirror; robust capon beamforming; passive location

3:20

2pSP5. Application of time reverse mirror in underwater acoustic networks communication. Yin Jingwei, Zhang Xiao, Guo Longxiang, and Sheng Xueli (College of Underwater Acoustic Engineering, Harbin Engineering University, Harbin 150001, China, yinjingwei@hrbeu.edu.cn)

Time reversal mirror (TRM) could match the acoustic channel impulse responses (CIR) automatically to achieve channel equalization without any transcendental knowledge. Active time reversal mirror (ATRM) needs monostatic sensor, and the signal propagates through the acoustic channel twice leading to inefficient. Further, the array processing enhances the complexity. In order to overcome disadvantages, the single sensor passive time reversal mirror (PTRM) and virtual time reversal mirror (VTRM) are presented. Based on the properties of underwater acoustic channel, using the uncorrelated character among different users' CIR, the single-element time reversal mirror is proposed to apply to underwater acoustic networks. The scheme could focus the desired user's information and suppress the undesired users' information. Furthermore, we have performed results of numerical simulations. Key words: time reversal mirror (TRM); channel equalization; underwater acoustic communication; networks

3:40

2pSP6. Own voice detection with near field head related transfer function based on frequency domain binaural model. Taira Onoguchi, Yoshifumi Chisaki, and Tsuyoshi Usagawa (Kumamoto University, 2-39-1 Kurokami, Kumamoto, Japan, taira@hicc.cs.kumamoto-u.ac.jp)

On the binaural hearing assistance systems with directivity control, it is difficult to distinguish user's own voice from the target speech located in front of the user. Therefore, emphasized user's own voice degrades the quality of hearing assistance. Because the own voice arrives from the specific angle in the median plane in the near field, it is difficult to distinguish whether the incoming sound is own voice or not. However, in the previous works, the elevation angle of signal source can be estimated even on median plane by means of artificial neural network based on the interaural level and phase differences. In this paper, a new method to detect own voice is proposed. This method utilizes the artificial neural network to discriminate own voice from any of signal sources located far than 1.5m from the user. The performance of this system is examined by simulation.

4:00–4:20 Break

4:20

2pSP7. Array signal Processing for noise control and signal separation based on time-reversal and impulse response. Yu-Hao Hsieh and Gee-Pinn Too (Dept. of Systems and Naval Mechatronic Eng., Natl. Cheng Kung Univ., 1 University Rd., Tainan 70101, Taiwan, p1894105@mail.ncku.edu.tw)

Noise control and signal separation are important purposes for acoustic signal processing. This study presents a detailed analysis for designing an acoustic signal processing procedure based on time-reversal method, which has been widely used for capable to compensate distortion due to multipath effect. However, setting transducers to retransmit at source places is impracticable for some applications. A way to overcome such limitation is to model wave propagation path between two points using impulse response function. Adaptive digital filter, deconvolution with singular value decomposition and Tikhonov regularization, and correlation function, are chosen to calculate impulse response function. All three different techniques above are tested in details with various array properties. The conclusion is made according to the level of accuracy using signal-to-noise ratio and correlation coefficient as indicators, and the computation time under the same controlled parameters as well.

4:40

2pSP8. Group modes based time reversal imaging algorithms for pipeline inspection using ultrasonic guided waves. Weichang Li (ExxonMobil Corporate Strategic Research, 1545 Route 22 East, LC326, Annandale, NJ 08801, weichang.li@exxonmobil.com), Shuntao Liu, and Limin Song

This paper presents a time reversal imaging technique utilizing a group of guided wave modes over a broad frequency band. This increases both the bandwidth integrated power and the mode diversity of the propagating wave. Typically, dispersion is minimized by selecting a single mode. Instead, we develop algorithms to coherently combine the group of modes to obtain increased signal to noise ratio and sensitivity, and we compensate for dispersion using time reversal to improve image quality and extend the inspection range. Imaging is computed via an efficient angular spectrum propagation algorithm without numerically computing the Greens functions. The performance improvement is demonstrated via signal processing results based on numerically simulated data.

5:00

2pSP9. A maximal mutual information based feature selection method in sound classification. Xueli Fan and Haihong Feng (Shanghai Acoustic Laboratory, Institute of Acoustics, Chinese Academy of Sciences, Xiao Mu Qiao Road 456, Shanghai, China, shirleyfan916@gmail.com)

In sound classification problem, a feature selection method based on improved maximum mutual information is proposed. Better sound classification performance is expected with smaller computational effort by evaluating the "information content" of classification features, only selecting the relevant features of a classifying system, and excluding redundant ones. The proposed method is based on a "greedy" selection of the classification features. The mutual information is measured based on both the output class and the selected features. Simulation experiment was carried out to evaluate performance of sound classification with the proposed feature selection method. The selected features were feed into neural network for sound classification. The classification accuracy was compared with the mutual information feature selector (MIFS). Experimental results showed that the proposed method

can produce better sound classification than the classical method. This work has been sponsored by Jiaxing Engineering Center, Institute of Acoustics, Chinese Academy of Sciences and Jiaxing Earelectric Co., Ltd.

5:20

2pSP10. Design and deployment of infrasonic sensor monitoring network in China. Haonan Feng, Yichun Yang, and Chunlian Men (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, fhn02212005@163.com)

The Infrasonic Monitoring Network (IMN) in China is under construction to detect, identify and located the signals generated by natural disasters. Presently, fifteen sensors are operational and transmitting real time data to Central Data Centre (CDC) in Beijing. Each sensor station is composed of one infrasonic sensor, one humidity and temperature sensor, one digital sampling and transmitting device, which collects and delivers real-time continuous infrasonic, temperature, humidity, GPS raw data to CDC. IMN is designed as a robust sensor network optimized for rapid deployment and provides long-term information monitoring; the system supports data acquisition with GMT time and is remotely configurable. Its high performance has been demonstrated in the initial deployment experience and preliminary analysis of sampling data.

5:40

2pSP11. Third-order nonlinear IIR filter for compensating nonlinear distortions of loudspeaker system. Kenta Iwai and Yoshinobu Kajikawa (Faculty of Engineering Science, Kansai University, 3-3-35, Yamate-cho, Suita-shi, Osaka 564-8680, Japan, kenta1986@gmail.com)

In this paper, we propose a 3rd-order nonlinear IIR filter for compensating nonlinear distortions of loudspeaker system. The proposed filter is derived from the nonlinear differential equation of the loudspeaker system. The conventional filter, which is called the 2nd-order nonlinear IIR filter, cannot reduce nonlinear distortions at high frequencies because this filter does not include the nonlinearity of the self-inductance of the loudspeaker system. On the other hand, the proposed filter includes the effect of the self-inductance and can reduce nonlinear distortions at high frequencies. Experimental results demonstrate that the proposed filter can reduce intermodulation distortions by 2 dB at high frequencies.

6:00

2pSP12. Indoor beamformer design using room simulators. Zhibao Li, Cedric Yiu (Department of Applied Mathematics, The Hong Kong Polytechnic University, zbli0307@163.com), Randolph Leung (Department of Mechanical Engineering, The Hong Kong Polytechnic University), and Sven Nordholm (School of Electrical and Computer Engineering, Curtin University, Perth, Australia)

Broadband microphone array provides an important means of hands-free speech acquisition via spatial beamforming techniques. There are many different approaches for the design of near-field broadband beamformers. One method is based on the wave propagation model using a direct path transfer function. However, as signals are corrupted by different interfering noise, room reverberation plays a particular important role for indoor applications even if there is no another speaker around. Image-source method is one simple but effective approach for room acoustic simulation. It is also possible to employ a wave-based model for the simulation. In this paper, we will study different room simulators and employ them for the design of indoor beamformers.

6:20

2pSP13. Study on localization of infrasound waves radiated by natural events. Jun Lu and Yichun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, lvjun@mail.ioa.ac.cn)

In recent years, there happened several large earthquakes which have made very serious destruction to local people and society. Among these terrible events' gestation, occurring, and developing, infrasound waves radiated in these processes were often observed. According to the signals received by monitoring sensors, the infrasound sources should be located accurately by an infrasound microphone array. The array was constructed by several sensors of InSAS2008 developed by IACAS, distributed in the North China area. In near future, the sensors will be constructed in all China. Data of signals received by this network could be processed by some methods such as Progressive Multi Channel Correlation (PMCC), sound imaging, rainbow algorithm, etc., to get useful information of nature events. Depending upon this work, some natural events, such as earthquake, debris flow, tsunami, volcanic eruption, etc., could be forecasted under certain conditions, or get more knowledge of them.

6:40

2pSP14. Dialectal alarm words recognition based on a hybrid model of Hidden Markov Models and the BP Neural Network. Ling Lu (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, luling12345678@yahoo.cn), Xiangyang Zeng (Institute of Environmental Engineering, College of Marine Engineering, Northwestern Polytechnical University, Xi'an 710072, China), Xiaobin Cheng, and Zhaoli Yan (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)

This paper explores the small-vocabulary speaker-independent isolated-word speech recognition technology. A recognition system based on a hybrid model of Hidden Markov Model and BP Neural Network is constructed. The HMM is used for computing the Viterbi output score which is inputted into the BP network to acquire the classification information. A corpus with more than 2500 speech samples is created, which includes three dialects (mandarin, shannxi dialect, English), and each kind of dialect contains four alarm word ("help", "hijack", "murder", fire"). Recognition experiments are carried out using the corpus. The results show that the hybrid model has higher performance than the hidden Markov model.

Session 2pUWa

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

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Peter Worcester, Cochair
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Invited Paper

2:00

2pUWa1. The North Pacific Acoustic Laboratory (NPAL) deep-water acoustic propagation experiments in the Philippine Sea.
Peter F. Worcester (Scripps Institution of Oceanography, UCSD, La Jolla, CA 92093, pworcester@ucsd.edu), Rex K. Andrew (Applied Physics Laboratory, University of Washington, Seattle, WA 98105), Arthur B. Baggeroer (Massachusetts Institute of Technology, Cambridge, MA 02139), John A. Colosi (Naval Postgraduate School, Monterey, CA 93943), Gerald L. D'Spain, Matthew A. Dzieciuch (Scripps Institution of Oceanography, UCSD, La Jolla, CA 92093), Kevin D. Heaney (OASIS, Inc., Fairfax Station, VA 22039), Bruce M. Howe (University of Hawaii, Honolulu, HI 96816), John N. Kemp (Woods Hole Oceanographic Institution, Woods Hole, MA 02543), James A. Mercer (Applied Physics Laboratory, University of Washington, Seattle, WA 98105), Ralph A. Stephen (Woods Hole Oceanographic Institution, Woods Hole, MA 02543), and Lora J. Van Uffelen (University of Hawaii, Honolulu, HI 96816)

The North Pacific Acoustic Laboratory (NPAL) Group has performed a series of experiments to study deep-water acoustic propagation and ambient noise in the northern Philippine Sea: (i) 2009 NPAL Pilot Study/Engineering Test (PhilSea09), (ii) 2010–2011 NPAL Philippine Sea Experiment (PhilSea10), and (iii) Ocean Bottom Seismometer Augmentation of the 2010–2011 NPAL Philippine Sea Experiment (OBSAPS). The goals are to (i) understand the impacts of fronts, eddies, and internal tides on acoustic propagation in this oceanographically complex and dynamic region, (ii) determine whether acoustic methods, together with other measurements and ocean modeling, can yield estimates of the time-evolving ocean state useful for making improved acoustic predictions and for understanding the local ocean dynamics, (iii) improve our understanding of the physics of scattering by small-scale oceanographic variability, and (iv) characterize the depth dependence and temporal variability of the ambient noise field. In these experiments, moored and ship-suspended low-frequency acoustic sources transmitted to a newly developed Distributed Vertical Line Array (DVLA) receiver capable of spanning the water column in deep water. The acoustic transmissions and ambient noise were also recorded by the towed Five Octave Research Array (FORA), by acoustic Seagliders, and by ocean bottom seismometers during OBSAPS. [Work supported by ONR.]

Contributed Papers

2:20

2pUWa2. The 2009 Philippine Sea Pilot Study/Engineering Test and the 2010 Philippine Sea Experiment: University of Washington cruises.
James Mercer, Rex Andrew, Linda Buck (APL-UW, 1013 NE 40th St., Seattle, WA 98105, mercer@apl.washingtonl.edu), Gerald D'Spain, Mathew Dzieciuch (SIO, 9500 Gilman Dr., La Jolla, CA 92093), Andy Ganse, Frank Henyey, Andrew White (APL-UW, 1013 NE 40th St., Seattle, WA 98105), and Peter Worcester (SIO, 9500 Gilman Dr., La Jolla, CA 92093)

Investigators at the University of Washington's Applied Physics Laboratory collaborated with scientists from the Scripps Institution of Oceanography during the 2009 Philippine Sea Pilot Study/Engineering Test and the 2010 Philippine Sea Experiment. The focus of both efforts was to collect well controlled low-frequency acoustic propagation data and detailed environmental information. The data from these cruises are presently being analyzed in the interests of: horizontal statistics of ocean spice as measured on a towed conductivity-temperature-depth (pressure) chain, fluctuation measures of low-frequency broadband ocean acoustic signals, bottom properties, and associated theoretical developments. This presentation will outline the experimental plans for each year, discuss preliminary analysis results, and provide an introduction for more detailed presentations in the remainder of this session. [Work supported by ONR.]

2:40

2pUWa3. The ocean bottom seismometer augmentation of the Philippine Sea experiment.
Ralph Stephen, Tom Bolmer (Woods Hole Oceanographic Institution, Woods Hole, MA 02543-1542, rstephen@whoi.edu), Peter Worcester, Matt Dzieciuch, Scott Carey, Brianne Moskovitz (Scripps Institution of Oceanography, La Jolla, CA 92093-0225), Sean McPeak (University of Washington, Seattle, WA 98105-6698), Richard Campbell (OASIS, Inc., Annapolis, MD 21409), Ernest Aaron (Scripps Institution of Oceanography, La Jolla, CA 92093-0225), and John Kemp (Woods Hole Oceanographic Institution, Woods Hole, MA 02543-1542)

The goal of the Ocean Bottom Seismometer Augmentation of the Philippine Sea Experiment (OBSAPS) was to study the coherence and depth dependence of deep-water ambient noise, in the band from 0.03 - 80Hz, and signals, in the band from 50-400Hz. The cruise sailed Kaohsiung to Kaohsiung, April 29 to May 16, 2011. A fifteen element OBSAPS - Distributed

Vertical Line Array (O-DVLA) with hydrophone modules from 12 to 852m above the seafloor was deployed in the Philippine Sea near 21degN, 126degE. Four short-period Ocean Bottom Seismometers (OBSs) and two long-period OBSs were deployed at 2km range from the O-DVLA. All of the OBSs had three-component inertial sensors and an acoustic pressure sensor. Three of the short period OBSs also had an external, autonomously recording hydrophone module identical to the hydrophone modules on the O-DVLA. The 12 day transmission program, using a J15-3 acoustic source, consisted of various M-sequences from 75.5 to 310Hz. Eight radial lines, a Star of David pattern, and nine station stops were transmitted out to 50km range. One radial line and nine more station stops were transmitted out to 250km range. Examples of possible Deep Seafloor Arrivals will be discussed. [Work supported by ONR.]

3:00

2pUWa4. Towed array propagation measurements and modelling in the Philippine Sea. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton, Dr. Fairfax Station, VA 22039, oceansound04@yahoo.com), Richard L. Campbell (OASIS Inc., Fairfax Station, VA 22039), Arthur B. Baggeroer (5 Mass Ave, Cambridge, Massachusetts 02139), Ralph Steven, Edward Scheer (Woods Hole Oceanographic Institution 93 Water Street — MS #16. Woods Hole, MA 02543), Peter Worcester, and Matthew Dzieciuch (SIO-UCSD 9500 Gilman Dr., La Jolla, CA 92093)

During the PhilSea09 experiment the US Office of Naval Research Five-Octave Research Array (FORA) was towed in various geometries making recordings of acoustic transmissions from an array of fixed sources, as well as a ship-suspended and ship-towed source. Physics issues include the structure of the convergence zone (CZ) and its dependence upon mesoscale sound speed and bathymetry, the structure of ambient noise, bottom interacting propagation and out-of-plane reverberation. In this paper, a survey of the recordings will be presented. Broadband Adaptive Beamforming techniques are applied to the array data. Direct comparison of measurements with high-fidelity broadband Parabolic Equation modelling will be presented. Much of the observed phenomenon, such as Transmission Loss (TL), Doppler spread, temporal coherence length and bottom bounce levels are well reproduced by the model. Physics issues that are not included in the modelling will be discussed – internal wave scattering, bottom roughness and out-of-plane propagation.

3:20

2pUWa5. Observed sound speed structure during the Phil Sea 2009-2011 field years. John Colosi (Naval Postgraduate School, Monterey, CA 93943, jacolosi@nps.edu), Brian Dushaw (Applied Physics Laboratory, University of Washington, Seattle, WA 98105), Lora Van Uffelen (University of Hawaii, Honolulu, HI 968822), Bruce Cornuelle (Scripps Institution of Oceanography, La Jolla, CA 92037), Matthew Dzieciuch, Peter Worcester (Scripps Institution of Oceanography, La Jolla, CA 92037), Steve Ramp (Soliton Ocean Services, Carmel, CA 93921), and Fred Bahr (Monterey Bay Aquarium Research Institute, Moss Landing, CA 95039)

From April 2010 to April 2011, 6 moorings equipped with temperature (T), conductivity (C), and pressure (P) sensors along with ADCP's observed oceanic variability in support of concurrent acoustic measurements between the moorings. In addition, for the month of April in 2009, two moorings monitored ocean conditions for a pilot study. During these periods energetic internal waves and tides, as well as eddies were observed thus creating an inhomogeneous, anisotropic, and rapidly changing acoustical environment. Some moorings possessed high precision T, C, and P records capable of resolving intrusive structures sometimes termed spice. In this talk statistical and deterministic metrics will be used to characterize the various dynamical sources of sound speed variability that were observed.

3:40

2pUWa6. Acoustic seagliders in the Philippine Sea: first steps towards moving-receiver tomography. Lora Van Uffelen (University of Hawaii, 1000 Pope Road MSB 205, Honolulu, HI 96822, loravu@hawaii.edu), Eva-Marie Nosal, Bruce Howe, Glenn Carter (University of Hawaii), Peter Worcester, and Matthew Dzieciuch (Scripps Institution of Oceanography, University of California San Diego)

Four Seagliders equipped with Acoustic Recorder Systems (ARS) received transmissions from moored, swept frequency (~200-300 Hz) acoustic sources as part of the ONR-sponsored North Pacific Acoustic Laboratory PhilSea10 Experiment. The gliders transited between the mooring sites from November 2010 until March 2011, diving between the surface and 1000-m depth and providing acoustic receptions at many ranges and depths with respect to the moored sources. The Seagliders utilized GPS positioning at the surface, but were underwater for up to 8 hours at a time, sometimes traveling several kilometers during a single dive. The precision to which the gliders can be located while underwater will be explored, based on acoustic arrivals from five moored sources, to resolve the fundamental ambiguity between position and sound speed. The ultimate goal is to use the Seagliders as additional acoustic tomographic receivers, thereby multiplying the number of acoustic paths in the tomography network.

4:00–4:20 Break

4:20

2pUWa7. Quantifying the mesoscale variability in the western subtropical countercurrent (STCC) and its impact on acoustic propagation. Steven Ramp (Soliton Ocean Services, Inc. 691 Country Club Drive, Carmel Valley, CA 93924, sramp@solitonocean.com), John Colosi (Dept. of Oceanography, Naval Postgraduate School, 1 University Circle, Monterey, CA 93943), Peter Worcester (Scripps Institution of Oceanography, La Jolla, CA 92093), and Fred Bahr (Monterey Bay Aquarium Research Institute, 7700 Sandholdt Road, Moss Landing, CA 95039)

During spring 2010 to spring 2011, the acoustics community deployed an impressive star-shaped moored array in the Philippine Sea to study acoustic propagation in deep water and its relationship to the oceanographic variability at a wide range of space and time scales. In addition to the acoustics instrumentation, six of the moorings spaced from 200-650 km apart were densely instrumented with velocity (u, v), Temperature (T), Conductivity (C), and Pressure (P) sensors making them ideally suited to study the mesoscale ocean circulation as well. Previous work has shown that the preferred baroclinic eddy length scale in the western STCC is order 350 km with most eddies propagating westward. These new observations will allow estimation of the vertical structure of the eddies, their propagation speed, their kinetic and available potential energy, and their dynamic stability. This improved view of the STCC eddies will be shared with the acoustics team for use in quantifying the impacts of the eddy variability on the acoustic propagation.

4:40

2pUWa8. Wavefront statistics from measurements made in the Philippine Sea and comparisons to path integral theory. Tarun K. Chandrayadula, John A. Colosi (Department of Oceanography, Naval Postgraduate School, Monterey, CA 93943, tkchandr@nps.edu), Peter F. Worcester, and Matthew Dzieciuch (Scripps Institution of Oceanography, UCSD, La Jolla, CA 92093)

Between April 2010 and April 2011, acoustic transmissions were carried out by six sources moored near the sound channel axis and, which were deployed across a 200-300 km radius in the Philippine Sea. The acoustic sources transmitted broadband chirp signals that spanned frequency bands ranging from 140-200~Hz to 200-300~Hz. The transmissions were recorded by a water column spanning Distributed Vertical Line Array (DVLA) that was placed roughly at the center of the area covered by the sources. The transmission ranges from the different sources to the DVLA varied from 125~km to 450~km. The Philippine Sea is an oceanographically diverse environment, which apart from internal waves also contains energetic eddies and internal tides. The acoustic data recorded by the spatially diverse array is hence an opportunity to quantify the degree of anisotropy in the acoustic propagation. This presentation first discusses the wavefront resolution capabilities of the DVLA. The receptions are then used to estimate narrowband ray statistics such as, mean intensity, scintillation index and, depth coherence. Apart from the narrowband statistics, broadband wavefront statistics such as pulse time spread, frequency coherence and pulse time wander are also estimated. The wavefront statistics for the different propagation paths at the array are contrasted with each other and then compared with predictions from path integral theory.

5:00

2pUWa9. Measured low frequency intensity fluctuations over a 107 km path in the 2009-2010 Philippine Sea experiment. Andrew W. White, Rex K. Andrew, James A. Mercer (Applied Physics Laboratory, University of Washington, Seattle, Washington 98105, andrew8@snark.apl.washington.edu), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238)

A broadband low-frequency acoustics pilot study was conducted in the Philippine Sea in April/May of 2009. 19,071 M-sequences were transmitted over a period of 60 hours at a range of 107 km to a vertical array of hydrophones. Timeseries of acoustic intensity for arrivals corresponding to known paths are formed. Results from a simulation of acoustic propagation through a time-dependent internal wave perturbed sound speed field agree qualitatively with arrivals for two of the acoustic paths but not with the arrivals for the path which had a shallow upper turning point of ~60 m. Intensity fades of 5 to 10 dB which last for approximately 18 or 20 hours are observed in this shallow-turning path. Estimates from data of spectra and correlation times for acoustic intensity will be compared to Monte Carlo Parabolic Equation based predictions. [Funding provided by ONR]

5:20

2pUWa10. Theoretical fluctuation predictions for low-frequency acoustical propagation ranges of 25 to 107 km in the 2009-2010 Philippine Sea experiment. Rex Andrew, Andrew White, James Mercer (Applied Physics Laboratory, 1013 Ne40th St, Seattle, WA 98105, randrew@apl.washington.edu), Peter Worcester, Matthew Dzieciuch (Scripps Inst. of Oceanogr. Univ. of California at San Diego, La Jolla, CA 92093), and John Colosi (Dept. of Oceanogr., Naval Postgraduate School, 833 Dyer Rd, Rm 328, Monterey, CA 93943)

Short range propagation experiments in deep water provide volume-only weak scattering paths that can be used to test the limits of validity of fluctuation theories for low-frequency acoustical signals. Colosi et al [J. Acoust. Soc. Am., 126, 1069-1083, 2009] suggested that Munk-Zachariasen theory (which uses first-order Rytov theory modified for the ocean environment) may be applicable in these cases; they used it to predict statistics for a 75-Hz signal propagated over a range of 87 km in the eastern North Pacific. Several scenarios used in the Philippine Sea experiments involving wideband signals may fall into this category: in 2009, ranges of 45 and 107 km were used (transmitter and receivers in the main sound channel) with carrier frequencies 82 and 284 Hz, and in 2010, continuous ranges from 25 to 43 km using a 61 Hz carrier (transmitter at 150 m and receivers near full ocean depth.) Predictions for all scenarios are presented, and comparisons are made against statistics observed for the 2009 107 km path. [Work supported by ONR.]

5:40

2pUWa11. Modeling uncertainty in transmission loss due to spatio-temporal variation in environmental parameters. Brett E. Bissinger (Graduate Program in Acoustics, The Pennsylvania State University, P.O. Box 30, State College, PA 16804, beb194@psu.edu), and Kyle M. Becker (NATO Undersea Research Centre, La Spezia, Italy)

Tactical prediction and decision aid tools require acoustic propagation modeling. The effectiveness of these tools relies on knowledge of the transfer function between model inputs - either measured or predicted - and model outputs. Of particular interest is this transfer when inputs are uncertain and characterized statistically. The Recognized Environmental Picture experiment 2011 (REP11) was designed to provide observations of spatio-temporal variability in oceanographic and acoustic quantities. REP11 was comprised of multiple runs of co-located and contemporaneous oceanographic and acoustic measurements repeated over twenty-four hours. Acoustic measurements were made using a broad-band source towed along radials from a fixed receiver array. The sound speed field in the water column was sampled independently during each run using gliders, towed instruments, and moorings, each having a different spatio-temporal resolution. Due to the nature of the ocean environment, the sound speed field varied both in time and space for each traversal of the track, yielding a suite of realizations representing the sensitivity of acoustic pressure to spatio-temporal variations in the environment. Based on these data, the statistics of transmission loss from propagation simulations using the observed sound speed fields are compared to the statistics of the measured acoustic pressure fields.

6:00

2pUWa12. Correlating measured oceanographic and acoustic variability toward understanding uncertainty transfer. Kyle M. Becker, John C. Osler, Yong-Min Jiang, Brett E. Bissinger (NATO Undersea Research Centre, La Spezia, Italy, becker@nurc.nato.int), and Sean Pecknold (Defence Research and Development Canada - Atlantic, Dartmouth, NS, Canada)

The Recognized Environmental Picture experiment 2011 (REP11) was conducted to support research in the areas of battlespace characterization, quantifying uncertainty, and decision support. By integrating results from these three areas, the goal is to incorporate physics based uncertainty transfer in models driving decision support tools. Generically, this is accomplished by parameterizing the environment according to need and providing the best knowledge available for each parameter including uncertainty. Experimentally, the research requires information on both the input and output sides of acoustic propagation models used for tactical prediction or decision aids. This talk describes an experimental effort to contemporaneously measure both oceanographic and acoustic quantities over a wide range of spatio-temporal scales using a combination of mobile, autonomous, and fixed assets. By measuring these quantities over the course of several days and repeating set geometries, many realizations of the oceanographic and acoustic fields were obtained. These data will be examined with respect to correlations between observed variability in the environment and variability in the measured acoustic fields.

6:20

2pUWa13. Influence of mesoscale eddies and frontal zones on sound propagation at the Northwest Pacific Ocean. V. A. Akulichev, L. K. Bugaeva, Yu. N. Morgunov, and A. A. Solovjev (V.I.Il'ichev Pacific Oceanological Institute, akulich@poi.dvo.ru)

The results of sound propagation through the warm mesoscale eddy in the region of Kuroshio at the Northwest Pacific are presented. The eddy core lies at point about 39° N, 149° E. Horizontal size of eddy was about 350 km, vertical size was more than 600 m. Continuous acoustic signals at the frequencies 232 Hz and 696 Hz were emitted by the sources towed at the depth of 100 m. Signals were received using a drifting system fitted with hydrophones. The results of experimental researches of sound propagation along the traces crossing subarctic frontal zones separating the cold subarctic and warm subtropical waters are submitted too. Reception system was located at a shelf zone near a peninsula Kamchatka at the depth about of 100 m. These traces extended to the distance 2100 km. It is shown that eddies and frontal zones result to the changes of acoustic signal levels. The experimental results were compared with the calculations of acoustic field.

6:40

2pUWa14. The effects of uncertain environment parameters on sonar operation range in shallow water. Wenbo Wang, Shuqiu Li, Guiqing Sun, Haining Huang, Chunhua Zhang, Jiyuan Liu, and Li Yin (Institute of Acoustics, Chinese Academy of Sciences, P.O. Box 2712, Beijing 100190, P.R. China, wangwenbo4@gmail.com)

Traditional sonar detection performance prediction is based deterministic prediction method, which does not take the impact of uncertain environmental parameters on transmission loss into consideration. Taking passive sonar detection as example, the farthest operation range is certainly unique for fixed frequency, of course, SL, NL, DI and DT are all well known. In this paper, a probabilistic prediction method is proposed in which transmission loss is subject to a probability distribution even for fixed frequency and range. An acoustic field modeling and numeric calculation framework is raised to fill the gap between underwater physics and signal analysis. For simple physical model depicting an infinite half-space consisting of two fluid layers (water and sediment), calculation results indicate that transmission loss is subject to gamma distribution due to roughness at the boundaries and sound speed fluctuation. In the end, some potential applicable prospects of this new acoustic field prediction method are discussed.

Session 2pUWb

Underwater Acoustics, Acoustical Oceanography, and Physical Acoustics: Rough Interface Scattering from Ocean Boundaries

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Invited Papers

2:00

2pUWb1. Frequency and angle spreading due to rough interface scattering. Cathy Ann Clark (Naval Undersea Warfare Center, Newport, RI 02841, cathy.clark@navy.mil)

A narrow band passive sonar system sensing propagation of acoustic energy reflected from the ocean bottom in shallow water incurs detection losses due to frequency and angle limitations of the processor and receiving array. These losses are expressible as ratios of integrals of a suitably defined bottom scattering function. This talk presents a modeling approach which uses propagation calculations and a tangent plane approximation to the Helmholtz scattering integral at planes within a layered bottom sediment to analyze sensitivities of the spreading problem and to predict system losses. Model results for a few hypothetical but representative sonar systems will be presented.

2:20

2pUWb2. The sea surface directional wave spectrum and forward scattering from the sea surface. Peter H. Dahl (University of Washington, dahl@apl.washington.edu), David R. Dall'Osto, and William J. Plant

An influence of the directional wave spectrum on acoustic forward scattering from the sea surface is difficult to measure. Here we present results of an experiment to measure vertical spatial coherence from an acoustic path interacting once with the sea surface at two different angles with respect to the wave direction. The measurements were part of the Shallow-Water 2006 program that took place off the coast of New Jersey in August 2006. An acoustic source was deployed at depth 40 m, and signals were recorded on a moored receiving system consisting of two, 1.4 m long vertical line arrays centered at depths 25 and 50 m. Measurements were made over four source-receiver bearing angles separated by 90°, during which sea surface conditions remained stable and characterized by an rms waveheight of 0.17 m and a mixed swell, and wind-wave field originating from different directions. The measurements show a statistically significant difference depending on source-receiver bearing when the acoustic frequency is less than about 10 kHz; a result not observed at higher frequencies. This paper will present field observations along with modeling based on a rough surface parabolic wave equation utilizing synthetic sea surfaces. [Research supported by ONR Ocean Acoustics].

2:40

2pUWb3. Reverberation modeling with transport theory. Eric I. Thorsos (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, eit@apl.washington.edu), Jie Yang, W. T. Elam, and Frank S. Henyey

Transport theory has been developed for modeling shallow water propagation at mid frequencies (1-10 kHz) where forward scattering from a rough sea surface is taken into account in a computationally efficient manner. The method is based on a decomposition of the field in terms of unperturbed modes, and forward scattering at the sea surface leads to mode coupling that is treated with perturbation theory. Transport theory has recently been extended to model shallow water reverberation, including the effect of forward scattering from the sea surface. Transport theory results will be compared with other solutions for reverberation examples taken from ONR Reverberation Modeling Workshop problems. These comparisons show the importance of properly accounting for multiple forward scattering in shallow water reverberation modeling. [Work supported by ONR Ocean Acoustics.]

Contributed Papers

3:00

2pUWb4. The effects of roughness on transmission loss in regions with elastic bottoms. Marcia Isakson (Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758, misakson@arlut.utexas.edu), and Nicholas Chotiros (Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758)

On rough elastic interfaces, incident acoustic waves couple more readily into the shear mode due to the steeper local angle relative to the nominal incident angle. This effect has a profound influence on transmission loss in shallow water areas with elastic bottoms. Although rock interfaces are often covered with sand or other sediment, the effect is still significant even for overlying sediment layers several acoustic wavelengths thick. In this study, the effects of roughness on transmission loss in an area with a hard elastic bottom with and without an overlying sediment layer are studied using a finite element transmission loss model. Results are compared with transmission loss data measured in an area with sediment layers over hard rock off the Western coast of Australia. [Work supported by ONR, Ocean Acoustics]

3:20

2pUWb5. Acoustic scattering from compact deformations of shallow-water waveguide's surfaces. Junying An (Qingdao Acoustics Laboratory, Institute of Acoustics, Chinese Academy of Sciences, annijy@yahoo.com.cn), and Haiting Xu (Qingdao Acoustics Laboratory, Institute of Acoustics, Chinese Academy of Sciences)

In shallow-water waveguide, compact deformations of sea surface or seafloor will affect the sound propagation. The deformations include local deformation of bottom topography, rock, trench and objects sank to the bottom, floater and surface ship on the sea surface, etc, which are all the components of an ocean waveguide. In this paper, the moments method and the boundary integral equation method are presented to compute the acoustic scattering from compact surface deformations in shallow-water waveguide. The former method combines the normal-mode theory with the numerical method for solving the boundary value problem of differential equation in local area, while the latter only covers the finite integral area of the deformed surfaces. The perturbation feature of near and far field and the deformation identification feature are analyzed.

3:40

2pUWb6. Attenuating underwater pile driving noise at a remote receiving location using an encapsulated bubble curtain. Mark S. Wochner, Kevin M. Lee (Applied Res. Labs., The University of Texas at Austin, Austin, TX 78713-8029, mwochner@mail.utexas.edu), and Preston S. Wilson (Mech. Eng. Dept. and Applied Res. Labs., The University of Texas at Austin, Austin, TX 78712-0292)

Noise generated underwater by pile driving can reach dangerous levels at significant distances from the source, and thereby potentially disturb both marine life and human activities. In this presentation, we describe a method of shielding a specific underwater region remote from the source using a curtain of encapsulated bubbles. The sizes of these encapsulated bubbles are chosen so that their resonance frequencies are near the peak frequency of the noise generated by the pile driving, thus maximizing each encapsulated bubble's ability to mitigate the incoming noise through resonance phenomena. The method of using encapsulated bubbles has been previously described by the authors [J. Acoust. Soc. Am. **127**:2015 (2010); J. Acoust. Soc. Am. **128**:2279 (2010); J. Acoust. Soc. Am. **129**:2462 (2011)], but in this work a region around the receiver, not the source, was treated with encapsulated bubbles, and the noise was due to pile driving activity rather than tonal. Results show that significant noise reduction can be attained using this encapsulated bubble curtain, in which a single layer yielded about 10 dB of SPL reduction. [Work supported by Applied Research Laboratories Internal Research and Development]

4:00–4:20 Break

4:20

2pUWb7. Effect of wind-generated bubble plumes on shallow water acoustic channels. Xiaopeng Huang (Stevens Institute of Technology, Castle Point on Hudson, Hoboken, NJ 07030, xhuang3@stevens.edu)

Wind blowing over the sea surface generates many dramatic effects. One of the most important effects is bubble plumes generated by breaking waves, which have major effects on high frequency acoustic propagation, sound speed profile and ambient noise due to the sound scattering and absorption properties of bubbles. On one hand, we will introduce several theoretical models of bubble plumes; on the other hand, we will propose how to calculate the acoustic signal attenuation and sound speed anomaly due to wind-generated bubble plumes. Simulation results will show the shallow water acoustic channel impulse response and sound speed anomaly with the effect of bubble plumes.

4:40

2pUWb8. Near bottom acoustic and video measurements of the rise rate of methane bubbles in the Gulf of Mexico. Christian de Moustier (HLS Research, Inc., 3366 North Torrey Pines Court, Suite 310, La Jolla, CA 92037, cpm@hlsresearch.com), Jan Boelmann (Hochschule Bremerhaven, University of Applied Sciences An der Karlstadt 8 D - 27568 Bremerhaven, Germany), and Barbara Kraft (Barrington, NH 03825)

Simultaneous acoustical and optical measurements of the rise rate of methane bubbles were made at several natural seafloor seeps in the Gulf of Mexico. The measurements were taken between 2 m and 50 m from the bottom, using a remotely operated vehicle equipped with a 500 kHz multibeam sonar system capable of imaging and Doppler measurements, and a high-definition color video camera. The bubble source on the seabed at all the seeps encountered in this work was an orifice a few centimeters across that released individual bubbles at various rates. Video measurements made 3 m above the bottom showed that these bubbles formed a column 5 to 30 cm in diameter. The acoustic measurements showed that the column transitioned from laminar to turbulent flow behavior about 6 m above the bottom. Rise rates in the laminar portion are between 20 and 30 cm/s. Work funded by BP