

Keynote Lecture

8:20

Language learning and the developing brain: Cross-cultural studies unravel the effects of biology and culture. Patricia K. Kuhl (Co-Director, Institute for Learning and Brain Sciences, Co-Director, NSF Science of Learning Center (LIFE), University of Washington, Seattle, Washington 98195)

Cross-cultural studies show that infants are born with innate abilities that make them “citizens of the world.” By the end of the first year of life, however, culture produces a dramatic transition. Infants’ abilities to discern differences in native-language sounds increase, and their abilities to discriminate sounds from other languages decreases. This perceptual narrowing of infants’ language skills is caused by two interacting factors: the child’s computational skills and their social brains. Computational skills allow rapid and automatic “statistical learning” and social interaction is necessary for this computational learning process to occur. This combination produces the neuroplasticity of the child’s mind, and contrasts with the more expert (but less open) mind of the adult. Neuroimaging of infants using Magnetoencephalography (MEG) is helping explain the extraordinary learning of young children. The work is leading to a new theoretical account for the “critical period” for language. Understanding the interaction between biology and culture in human learning in the domain of language may unlock some of the mysteries and mechanisms of the human mind.

Session 1aAA**Architectural Acoustics and Signal Processing in Acoustics: Multiple-Microphone Measurements and Analysis in Room Acoustics I**

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Chair’s Introduction—9:15

Invited Papers

9:20

1aAA1. Spherical microphone array processing of room impulse response data using frequency smoothing and singular-value decomposition. Nejem Huleihel and Boaz Rafaely (BGU, Beer Seva, 84105, nejem@ee.bgu.ac.il)

Room impulse responses (RIRs) play an important role in acoustical signal processing and room acoustics analysis. The problem of estimating the directions-of-arrival (DOA) of a source in a room and its reflections using RIR data and microphone arrays, is considered. Optimal array processing methods proposed for sound field analysis using spherical microphone array are utilized. Because of the possible coherence between the signals, these methods cannot be used directly, and a preprocessing technique is typically needed. Recently, frequency smoothing (FS) as a preprocessing technique has been developed for spherical microphone arrays. Although FS has already been developed for the general case, the study of its performance in a comprehensive manner, for spherical microphone arrays with RIR data has not been previously presented. Therefore, theoretical analysis of the signal matrix structure using RIR data is performed. The conclusions from this analysis may lead to an optimization of the smoothing process. A method for an optimal selection of frequencies in the smoothing process for the case of one reflection is presented, followed by formulations for smoothing in the more general case. Finally, FS and its relation to SVD of the array data matrix are also presented and discussed.

9:40

1aAA2. Joint spherical beam forming for directional analysis of reflections in rooms. Hai Morgenstern (Ben-Gurion University of the Negev, Beer-Sheva, hai.morgenstern@gmail.com), Franz Zotter (University of Music and Performing Arts, Graz), and Boaz Rafaely (Ben-Gurion University of the Negev, Beer-Sheva)

This contribution presents a new approach for analyzing spatial directions in room impulse responses captured with source and receiver of adjustable directivity. A distinct peak in a room impulse response is usually associated with an acoustic path length of direct or reflected sound. Given the ability to modify the directivity of source and receiver by spherical beamforming, beam coefficients can be

adjusted as to emphasize the peak at a preselected time instant. We present a new approach to jointly optimize the coefficients for both source and receiver under the constraint of a unit peak amplitude while minimizing the energy of the response. The beam pattern described by these coefficients highlights the dominant acoustic path directions of the corresponding path length at the source and the receiver.

10:00

1aAA3. Exploring spherical microphone arrays for room acoustic analysis. Jens Meyer and Gary W. Elko (mh acoustics, 25A Summit Ave, Summit NJ 07901, jmm@mhacoustics.com)

Spherical microphone arrays offer several advantages over linear microphone arrays and single sensor microphones for room acoustic analysis. Some advantages are the ability to: a) steer the directional response in 3D space, b) change the beam pattern shape (independent of the look direction) and c) spatial decomposition of the sound field into spherical harmonic orthonormal components. All of these features are available online and offline meaning that the analysis can be performed after the measurement has been done. We will present standard measurements such as spatially dependent reverberation time, diffuseness, etc. that take advantage of the spherical array decomposition of the soundfield. We will also revisit the spatial correlation function, a measure very suitable for spherical array based room analysis. Results for various setups will be presented.

10:20

1aAA4. On the Influence of sampling errors on the perception of spatial sound fields using spherical microphone arrays for auralization. Johannes Nowak (TU Ilmenau, Helmholtzplatz 2, 98693 Ilmenau, Germany, johannes.nowak@tu-ilmenau.de)

Spherical microphone distributions allow a three dimensional sampling of the sound field in a room. These microphone array data can be used for auralization on various playback systems. The aim of auralization is the reproduction of the sampled spatial sound field in order to give the listener the impression of being in the measured room. Due to the discrete spatial sampling process spatial aliasing corrupts the measured data. Therefore the resulting auralization quality is affected in terms of its spatial characteristics. Subjective quality measures for the spaciousness of sound fields can be represented by source localization accuracy, the apparent source width (ASW) and the listener envelopment (LEV). These subjective features are strongly related to objective measures like interaural level and time differences (ILD and ITD) or the interaural cross correlation (IACC). In subjective listening tests the influence of sampling errors on the binaural reproduction of a sampled sound field is investigated. The results are correlated with ITD, ILD and with IACC in order to gain an objective quality measure for sound fields recorded with spherical microphone arrays. The investigations are based on real measurement data taking various directions of arrival and different rooms into account.

10:40–11:00 Break

11:00

1aAA5. Interfacing spherical harmonics and room simulation algorithms. Michael Vorlaender, Martin Pollow, and Soenke Pelzer (RWTH Aachen University, D-52056 Aachen, Germany, mvo@akustik.rwth-aachen.de)

Room acoustic simulation by using geometrical acoustics is usually implemented with binaural receivers. Wave models such as FEM are easily applicable with binaural interfaces as well. This way, however, the signals are restricted to a specific set of HRTF, and a tedious task is to adapt the results to a proper reproduction system with very limited possibilities of listener individualization. With a more general interface such as spherical harmonics, room acoustic spatial data could be created in intermediate solutions. In post-processing this can lead to various binaural representations or to reproduction with Ambisonics (Dalenbäck, ICA 1995). In this paper it is discussed how standard routines in geometrical acoustics must be changed in order to implement multi-channel spherical microphone arrays. Furthermore, the corresponding output data can be multi-channel time signals or temporal SH coefficients or any other suitable spectral format. The amount of data and signal processing affects CPU time and memory. The discussion therefore is focused on feasibility and on consequences on the real-time performance on the one hand, and on the spatial quality of the room response, on the other.

11:20

1aAA6. The use of multi-channel microphone and loudspeaker arrays to evaluate room acoustics. Samuel Clapp (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180, clapps@rpi.edu), Anne Guthrie (Arup, 77 Water Street, New York, NY 10005), Jonas Braasch, and Ning Xiang (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180)

Most room acoustic parameters are calculated with data from omni-directional or figure-of-eight microphones. Using a spherical microphone array to record room impulse responses can yield more information about the spatial characteristics of the sound field, including spatial uniformity and the directions of individual reflections. In this research, a spherical array was used to measure room impulse responses on stage and in the audience in a wide variety of concert halls throughout New York State, with both the microphone array and an artificial head. The results were analyzed using beamforming techniques to determine spatial information about the sound field and compared to the results of geometrical acoustics and binaural localization models. Of particular interest was how the spatial data can help to differentiate between different spaces or listener positions that exhibit similar values for conventional metrics. Auralizations were created using both headphone playback and second-order ambisonic playback via a loudspeaker array. These systems were evaluated objectively to compare the reproduction systems with the measured data. Listeners were recruited for listening tests using each reproduction method. They were asked to evaluate the halls on both objective measures and subjective preference, and the results of binaural and ambisonic playback were compared.

11:40

1aAA7. Analysis and synthesis of room transfer function over a region of space using distributed spherical microphone arrays. Thushara Abhayapala (Australian National University, Canberra, ACT 0200 Australia, Thushara.Abhayapala@anu.edu.au), and Prasangha Samarasinghe (Australian National University)

Spatial sound field recording and reproduction in reverberant rooms requires measurement of room transfer functions (RTF) and corresponding compensation such as room equalization to avoid unintended effects. Typically, RTF rapidly varies over the room and hence requires a large number of point to point measurements to characterize the room. This paper uses (i) an efficient parameterization of the acoustic transfer function over a region of space, first introduced by Betlehem et al. ["Theory and design of soundfield reproduction in reverberant rooms," *Journal of the Acoustic Society of America*, Vol. 117, Issue 4, 2005] and (ii) a method to merge spatial soundfield recorded by distributed higher order microphones (such as spherical arrays) to analyze and synthesize the room transfer function over a region of space. This method provides a practical way to measure room transfer function over large areas with a minimum number of measurements.

12:00

1aAA8. On the importance of room acoustics in multi-microphone speech enhancement. Sharon Gannot (Bar-Ilan University, gannotsh@gmail.com)

Speech quality might significantly deteriorate in presence of interference. Multi-microphone measurements can be utilized to enhance speech quality and intelligibility only if *room acoustics* is taken into consideration. The vital role of the acoustic transfer function (ATF) between the sources and the microphones is demonstrated in two important cases: the minimum variance distortionless response (MVDR) and the linearly constrained minimum variance (LCMV) beamformers. The LCMV deals with the more general case of multiple desired speakers. It is argued that the MVDR beamformer exhibits a tradeoff between the amount of speech dereverberation and noise reduction. The level of noise reduction, sacrificed when complete dereverberation is required, is shown to depend on the direct-to-reverberation ratio. When the reverberation level is tolerable, practical beamformers can be designed by substituting the ATFs with their corresponding relative transfer functions (RTFs). As no dereverberation is performed by these beamformers, a higher level of noise reduction can be achieved. In comparison with the ATFs, the RTFs exhibit shorter impulse responses. Moreover, since non-blind procedures can be adopted, accurate RTF estimates might be obtained. Three such RTF estimation methods are discussed. Finally, a comprehensive experimental study in real acoustical environments demonstrates the benefits of using the proposed beamformers.

12:20

1aAA9. Representation of the spatial impulse response of a room. Filippo M. Fazi (University of Southampton, University Road, SO171BJ, Southampton, UK, ff1@isvr.soton.ac.uk), Markus Noisternig, and Olivier Warusfel (IRCAM - UMR CNRS, 1 place Igor-Stravinsky, 75004 Paris, France)

Microphone arrays allow for the measurement of the so-called spatial impulse response (SIR) of a room or of a concert hall. The SIR provides a local description of the reverberant field of that environment as a function of both time and space. It is shown that, under given assumptions, the SIR can be described by means of an integral operator, the so-called Herglotz wave function, which represents an infinite superposition of plane waves arriving from all possible directions. The kernel of this operator (the Herglotz kernel) contains all the information on the SIR. In practical cases only a limited amount of information is available to compute the Herglotz kernel, typically because a finite number of sensors is used for the measurement. In that respect, several alternatives are discussed to represent the Herglotz density as a sum of a finite number of basis functions. Some results for numerical simulations are then presented, which show the Herglotz kernel for simple examples. Finally, some limitations of this representation are discussed, especially those imposed by the use of real microphone arrays.

Session 1aBA

Biomedical Acoustics: Therapeutic Ultrasound

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Contributed Papers

9:20

1aBA1. Acousto-optic monitoring of high-intensity focused ultrasound lesion formation with fibre-coupled autocorrelation detection. Samuel Powell and Terence S. Leung (Department of Medical Physics and Bioengineering, Malet Place Engineering Building, University College London, London, WC1E 6BT, UK., spowell@medphys.ucl.ac.uk)

A focused acoustic source insonifies an optically turbid medium. Under coherent illumination the optical field in the focal region of the acoustic source is phase modulated by the acousto-optic interaction. The degree of this modulation can be determined using a fibre-coupled optical autocorrelation technique. Exploiting both the contrast of biological tissues at near-infrared wavelengths, and the non-linearity of the phase modulation process, it may be possible to determine the pertinent optical properties of biological tissues with a spatial resolution comparable to the dimensions of the acoustic focus. The same acoustic source may be employed therapeutically at higher power levels to instigate thermal necrosis and associated optical changes in e.g., tumours of the prostate. Whilst the proposed detection regime has significant technical and practical advantages over alternative approaches currently under investigation, it is incompatible with such treatment power levels. We present the theory of an interleaved treatment and sensing technique which could allow the use of our inherently compact and robust detection mechanism during HIFU therapy, simulated results obtained using a novel highly-parallel Monte-Carlo simulation code, and initial experimental results from the formation of lesions within ex vivo chicken breast samples.

9:40

1aBA2. Cavitation bubble in alcohol aqueous solutions. Weizhong Chen, Weicheng Cui, and Suibao Qi (Key Laboratory of Modern Acoustics, Ministry of Education, and Institute of Acoustics, Nanjing University, Nanjing, 210093, China, wzchen@nju.edu.cn)

The alcohol, as a surface active agent, plays an important role in sonoluminescence. The violent pulsation of the cavitation bubble makes the sonoluminescence possible. In this talking we report the experimental measurement for the bubble pulsations in alcohol aqueous solutions at different concentration subjected to the excitation of the ultrasound. The results shows that the maximum radius and the bearable intensity of the ultrasound of the bubble decrease with the concentration increasing. At the same time, the compression ratio of the volume goes also into decline as the concentration increases. These results are consistent with the observations of sonoluminescence in alcohol aqueous solutions. And we conclude that the weakened bubble pulsation causes mainly the sonoluminescence darkened in alcohol aqueous solutions. A question about decreasing in the bearable ultrasound intensity of the cavitation bubble in alcohol aqueous solution is still open and worthy of further investigation.

10:00

1aBA3. Real-time phase correction for transcranial focused ultrasound surgery. Yun Jing (North Carolina State University, 911 Oval Dr., EBIII, Campus Box 7910, Raleigh, 27695 NC, yjing2@ncsu.edu)

The skull has been a barrier to transcranial focused ultrasound therapy, because of its strong phase aberration. Previous methods for phase correction are based on numerically solving the wave equation, which outputs the desired phase delay for each transducer element. These methods are typically quite time-consuming. The present method aims to achieve real-time phase correction. This method is based on the Eikonal equation, which is a high frequency approximation to the wave equation. It fully accounts for the refraction in the skull, which is the main contribution to the phase aberration in the skull. Fast marching method (FMM) is used to solve the Eikonal equation. Preliminary results show that, solving the Eikonal equation is over 100 times faster than solving the wave equation by the finite-difference time-domain method. More importantly, a relatively sharp and accurate focus can be achieved in the brain using the present method.

10:20

1aBA4. The effects of acoustic power and exposure time on the hyperecho in ultrasound images at 55 °C using MRI and US guided HIFU in a bovine liver specimen. Faqi Li, Huarong Yi, Mingsong Zhong, Huijian Ai, Jie Chen, and Zhibiao Wang (State Key Laboratory of Medical Ultrasound Engineering Co-founded by Chongqing and Ministry of Science and Technology, Department of Biomedical Engineering, Chongqing Medical University, Chongqing 400016, P.R. China, ermei0810@163.com)

Ex vivo bovine liver specimens were exposed to the MRI-guided HIFU with the focusing depth of 15 mm in the specimens and various acoustic power (50 W, 100 W, 150 W, 200 W, 250 W and 300 W). Our interest was focused on a case of 55 °C in situ temperature. The temperature in situ was monitored via the T-map of MRI. The exposure time needed to reach 55 °C in the focus for a acoustic power was recorded. The same procedure was repeated to new but similar bovine liver exposed to the US-guided HIFU with the same sonication parameters. The procedure was also monitored by a passive cavitation detection system. The results showed to reach 55 °C in situ the exposure time decreased with the increase of acoustic power. The coagulative necrosis occurred when the acoustic power was 50 W, but no hyperecho in US images and half harmonic emission were found. The coagulative necrosis, hyperechoic US images and half harmonic emissions were observed when the acoustic power was 100 W or greater. At 55 °C, since no boiling bubbles occurred, therefore we concluded that the hyperecho in US images were caused by acoustic cavitation whose occurrence is determined by the applied acoustic power. Keywords: MRI-guided HIFU, US-guided HIFU, Coagulative necrosis, hyperecho, Acoustic cavitation This work was supported by National Nature and Science Foundation of China (No. 30830040, 30970827)

10:40–11:00 Break

11:00

1aBA5. Generating uniform lesions in high intensity focused ultrasound ablation. Yufeng Zhou (Nanyang Technological University, 50 Nanyang Ave., Singapore, 639798, yfzhou@ntu.edu.sg)

High intensity focused ultrasound (HIFU) is emerging as an effective oncology treatment modality. Because of thermal diffusion from nearby spots, the lesion size will gradually become larger as HIFU progresses. However, uniform lesions with the least energy exposure are preferred by the physician in tumor ablation. In this study, an algorithm was developed to determine the number of pulses delivered to each spot in order to generate uniform lesion pattern that fills the region-of-interest completely using different scanning pathways (raster scanning, spiral scanning from the center to the outside and from the outside to the center), spot spacing, and motion time. It is found that spiral scanning from the outside to the center with spot spacing of 2 mm and motion time less than 10 s would need the least number of pulses in uniform lesion production with the minimal temperature elevation. In addition, the effects of thermal properties of tissue (i.e., specific heat capacity, convective heat transfer coefficient, and thermal conductivity) on HIFU ablation were investigated. Altogether, dynamically adjusting ultrasound exposure energy can improve the efficacy and safety of HIFU ablation, and the treatment planning depends on the scanning protocol and thermal properties of the target.

11:20

1aBA6. Efficient generation of cavitation bubbles by dual-frequency exposure. Jun Yasuda (Tohoku University, 6-6-05 Aramaki Aoba Aoba-ku Sendai-shi 980-8579, Japan, j_yasuda@ecei.tohoku.ac.jp), Ryo Takagi, Shin Yoshizawa, and Shin-ichiro Umemura (Tohoku University, 6-6-05 Aramaki Aoba Aoba-ku Sendai-shi 980-8579, Japan)

Microbubbles are known to enhance high intensity focused ultrasound (HIFU) treatment, which is a new cancer treatment method. Highly negative acoustic pressure can efficiently generate cavitation microbubbles, but it is difficult to obtain at the focus of HIFU because of nonlinear propagation. In our previous study, a "Dual-Frequency Excitation" method was suggested to synthesize waveforms emphasizing either the positive-peak-pressure or the negative-peak-pressure by superimposing the second harmonic onto the fundamental. In this study, four different type of dual-frequency exposure sequence at the fundamental frequency of 0.8 MHz were used, and the behavior of cavitation bubbles captured by a high-speed camera was compared. In the first and second sequences, the positive-peak-pressure emphasized (P) and negative-peak-pressure emphasized (N) waves were employed for 125 μ s, respectively. In the third sequence, the N and P waves were employed in the earlier and later 62.5 μ s, respectively, and they were exchanged in the fourth sequence. In the results, the amount of cavitation bubbles generated by the third sequence was significantly more than the other three sequences. The cavitation bubbles, generated by the N waves, are thought to have provided a pressure-release surface converting the P to N waves, which further generated cavitation bubbles.

11:40

1aBA7. Detection of high intensity focused ultrasound induced cavitation activity in liver tissue. Tingbo Fan, Zhenbo Liu, Xiasheng Guo, and Dong Zhang (Key Laboratory of Modern Acoustics (MOE), Institute of Acoustics, Nanjing University, tingbof@gmail.com)

Microbubbles are known to be able to enhance the thermal effect of ultrasound. In HIFU procedure, microbubbles can be generated when the peak negative pressure is large enough or the temperature exceeds the boiling point. In this work, cavitation activities in various exposure protocols with equal total acoustic energy but variable focus pressure and variable duty cycle were monitored in vitro. A 10 MHz focused passive cavitation detector transducer was used to capture acoustic emissions emanated from

liver tissue exposed to 1.12 MHz HIFU pulses, while the focus temperature was recorded. The inertial cavitation dose (ICD) was calculated to analyze the cavitation activity qualitatively. The correlations of cavitation activity, temperature and focus pressure were discussed. [This work is supported by the National Basic Research Program 973 (Grant No. 2011CB707900) from Ministry of Science and Technology, China, National Natural Science Foundation of China (11174141), and the Fundamental Research Funds for the Central Universities (Grant Nos. 1103020402, 1116020410 and 1112020401)]

12:00

1aBA8. Infrared and hydrophone system for estimating the output power of high intensity focused ultrasound transducer. Ying Yu, Guofeng Shen, Jingfeng Bai, and Yazhu Chen (Biomedical Instrument Institute, School of Biomedical Engineering, Shanghai Jiao Tong University, Shanghai 200030, China, simonyu2008@gmail.com)

Output power of high intensity focused ultrasound (HIFU) transducer is not only important for the safety and efficiency of clinical treatment, but also for therapy planning in medical applications. In the current paper, a method was proposed to estimate output power of HIFU using a hydrophone and infrared system. The proposed method is independent of the thermal and acoustic parameters of the acoustic absorber and the type of transducer that has been measured. This method consisted of five steps. The amplitude absorption coefficient of the medium was measured through the first two steps. Through the second and third steps, we estimated the ratio of the heat capacity per unit volume to the ultrasonic amplitude absorption coefficient of the absorber. In fourth step, the temperature change at the absorber/air was captured by an IR camera, and the temperature change rate (TCR) was used to estimate the intensity based on the parameters measured by the first three steps. In last step, the sound power of HIFU transducer at high driving voltage can be obtained following the relationship between the sound intensity and sound power. The method was proposed and simulated in three 2-D 1.36 MHz-phased arrays and two kinds of absorbers. In last step, the sound power of HIFU transducer at high driving voltage can be obtained following the relationship between the sound intensity and sound power.

12:20

1aBA9. An Acoustic backscatter-based method for estimating attenuation towards monitoring lesion formation in high intensity focused ultrasound. Siavash Rahimian and Jahan Tavakkoli (Department of Physics, Ryerson University, Toronto, ON, Canada, M5B 2K3, siavash.rahimian@ryerson.ca)

This work investigated the transient characteristics of tissue attenuation coefficient before, during and after HIFU treatment at different total acoustic powers (TAP) in ex-vivo porcine muscle tissues. Dynamic changes of attenuation coefficient parameters were correlated with conventional B-mode ultrasound images over the whole HIFU treatment process. Two-dimensional pulse-echo radiofrequency (RF) data were acquired to estimate the changes of least squares attenuation coefficient slope ($\Delta\beta$) and attenuation coefficient intercept ($\Delta\alpha_0$) averaged in the region of interest, and to construct $\Delta\beta$, $\Delta\alpha_0$, and B-mode images simultaneously. During HIFU treatment, bubble activities were visible as strong hyperechoic regions in the B-mode images, causing fluctuations in $\Delta\beta$ and $\Delta\alpha_0$ during treatment. $\Delta\beta$ and $\Delta\alpha_0$ increased with the appearance of bubble clouds in the B-mode images to values in the range of 1.5-2.5 [dB/(MHz.cm)] and 4-5 [dB/cm], respectively. After the treatment, $\Delta\beta$ and $\Delta\alpha_0$ gradually decreased, accompanied by fadeout of hyperechoic spot in the B-mode images, until they were stable at 0.75-1 [dB/(MHz.cm)] and 1-1.5 [dB/cm], respectively. After treatment, $\Delta\beta$ and $\Delta\alpha_0$ images outperformed B-mode images by having significantly higher contrast to speckle ratios at all investigated TAP values.

Session 1aEA

Engineering Acoustics and Noise: Mufflers and Silencers

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Contributed Papers

9:20

1aEA1. Systematic design of reversal flow mufflers by topology optimization. Jin Woo Lee (Division of Mechanical Engineering, Ajou University, San 5 Woncheon-Dong, Yeongtong-Gu, Suwon 443-749, Republic of Korea, jinwoolee@ajou.ac.kr)

A new muffler design method is suggested for systematic design of reversal flow mufflers. In the new method, a muffler design problem is reformulated as an acoustical topology optimization problem, where the transmission loss at the frequency of interest is maximized. A finite element model is employed for acoustical analysis, and one design variable is assigned to each finite element and changes continuously from zero to one. When the design variable becomes one, the associated finite element is filled with rigid body and an incident acoustic wave is fully reflected. The rigid bodies build up partitions, which improve the acoustical characteristics of flow-reversing chambers. When the design variable becomes zero, an incident acoustic wave is freely transmitted to the other side. Since the optimal location and length of the partitions are determined automatically by the suggested muffler design method, the internal configuration of the reversal flow mufflers does not depend on the designers' intuition and experiences. Several numerical results prove the feasibility of the suggested muffler design method.

9:40

1aEA2. Transversal modes and acoustic attenuation characteristics of rectangular and oval silencers with perforated tube. Zhi Fang and Zhenlin Ji (School of Power and Energy Engineering, Harbin Engineering University, Harbin, Heilongjiang, P.R. China, zihuiying@163.com)

As the analytical method is not suitable for the silencers with arbitrary cross-sectional shape, the finite element method is developed to calculate the transversal modes of rectangular and oval silencers with circular perforated tube, the corresponding finite element formulation is derived and the computational code is written. In order to validate the present finite element formulation and computational code, the transversal modal frequencies of circular concentric straight-through perforated tube silencer are evaluated analytically and compared with the finite element results, and good agreements between them are observed. Then, the finite element method is used to investigate the effects of hole diameter, porosity and tube offset on the transversal modes and acoustic attenuation characteristics of rectangular and oval silencers with circular perforated tube. The numerical results demonstrate that, smaller hole diameter or higher porosity leads to higher plane wave cut-off frequencies and better acoustic attenuation in the middle frequency range, and the hole diameter and porosity have negligible effect on the plane wave cut-off frequencies when the porosity is higher than 40%. The plane wave cut-off frequencies of the non-coaxial silencers are lower than the concentric configurations in general.

10:00

1aEA3. Narrow sidebranches for duct silencing. S.K. Tang (Department of Building Services Engineering, The Hong Kong Polytechnic University, Hung Hum, Hong Kong, China, besktang@polyu.edu.hk)

A narrow sidebranch attached to the rigid wall of a duct will result in high sound transmission loss across it at its resonance frequencies. Coupling narrow sidebranches together will therefore produce a broadband silencing device for duct noise control. However, the sidebranch length variation will affect the broadband performance. Numerical investigation was carried out in this study to understand the effects of the sidebranch length variation and the sidebranch width on the overall sound attenuation spectrum of the coupled sidebranches. It is found that broadband sound attenuation below the first higher mode cut-off frequency of the main duct of over 20dB across the working bandwidth can be achieved if the length variation and widths of the sidebranches are appropriately chosen.

10:20

1aEA4. Effect of geometric uncertainties and variations on the one-dimensional sound transmission in a duct with periodic resonator array. Jeong-Guon Ih and Eun-Ok Yim (KAIST, J.G.Ih@kaist.ac.kr)

Sound transmission in a one-dimensional duct with periodic resonator array is characterized by Bragg stopband due to periodicity and resonance stopband due to resonator. Involved geometric parameters affecting the acoustic characteristics are resonator spacing, resonator length, widths or areas of main duct and resonator. Distortions of such geometric parameters are due to uncertainties in manufacturing and due to intentional design variations for focusing on a target frequency range. A side-branch array was taken as the test example. Stopband information was obtained by four-pole matrix and Bloch wave theory. Area and length ratios between side-branch and main duct periodicity properties were varied from zero to unity. Randomized distortions were generated from either Gaussian or uniform random distribution. As a deterministic distortion, sine function was employed. Simulation results showed that bandwidths and frequencies of stopbands were highly affected by the length ratio. Along with the increase of random distortion rate or function period of deterministic distortions, sound transmission at stopbands decreases, while passband transmission increases. It was also shown that one can change the bandwidth and/or frequency of stopbands as desired for sound reduction. (Work partially supported by BK21 project and NCRC (NRF 2011-0018242))

10:40–11:00 Break

11:00

1aEA5. New semi-active muffler system based on the H-Q tube concept. Xueguang Liu, Changchun Yin, Ye Wang, Shiming Cui, and Chunxia Li (School of Energy and Power Engineering; Harbin Engineering University, Harbin, Heilongjiang, Xueguang_liu@hotmail.com)

For a fixed bandwidth noise, the appropriate control device is used to change the internal structure of the semi-active muffler to get the large amount of noise reduction. This paper analyzes the principle of Herschel-Quincke tube, then according to the principle of the Herschel-Quincke tube, a semi-active silencing device is presented here, which can effectively control the noise. Then a test bench basing on the design is built. The control system which includes the control of the valves and the stepping motor is studied here. In the conditions without flow, the acoustic characteristic testing has been done using the control systems. It shows that the valves and the stepping motor have a rapid response, meanwhile, the testing results are identical with the theoretical control state, which achieves the control of the semi-active muffler. According to the analysis of the testing results, the muffler has a good noise reduction effect to low frequency noise and the harmonic frequency noise corresponding to the low frequency. It shows an average noise reduction of 10dB as well as the maximum noise reduction approaching to 35dB, which reveals the excellent noise reduction characteristic of the muffler.

11:20

1aEA6. Design of compartmental silencer for HVAC system. Y. H. Chan (Department of Mechanical Engineering, The Hong Kong Polytechnic University, Hong Kong, China, mmyschoy@polyu.edu.hk), Y. S. Choy, and R. C. K. Leung

Air conditioning and ventilation system is the major noise sources in the commercial building. Noise will be propagated from fan and through the associated ductwork into working area. In order to reduce the noise transmitted, various type of silencers can be placed in the ductwork to absorb noise or reflect them back to the source. Usually the dominant noise is low-

to-middle frequencies, which active noise control has the potential to control the low-frequency noise, issue related to reliability and cost remains. Concerning the real practical situation, passive control is the most preferable choice. The traditional in-duct silencers are in splitter type, with a bulk of fibrous material as duct lining. The existing passive silencers are usually bulky and long and can give a desirable performance at mid-to-high frequencies. Most ideally the silencer in concerned should be able to handle a broad frequency band and compact in size. In this paper, the performance of a new silencer design was examined and optimized using computation approach with experimental verification.

11:40

1aEA7. Determination of sound reflection coefficient of circular duct using time-domain computational fluid dynamics method. Chen Liu and Zhenlin Ji (College of Power and Energy Engineering, Harbin Engineering University, Harbin, Heilongjiang 150001, P.R. China, liuchenlqq@163.com)

In this paper, the software FLUENT is used as simulation tool, and the two-dimensional time-domain Computational Fluid Dynamics (CFD) approach is employed to compute the sound reflection coefficient of circular duct without and with gas flow. In the absence of mean flow, the pressure far-field boundary condition could be used as non-reflecting boundary condition in Fluent, and good agreement between the CFD prediction and experiment measurement available in the literature is observed. For the case with gas flow, the general non-reflecting boundary condition is available only with the density-based solver (high-speed compressible flow or Strong coupling flow) in FLUENT, and it is difficult to acquire the convergent solution for the calculation that the density-based solver is used to compute the reflection coefficient of circular pipe. Therefore, the non-reflecting boundary condition is not applied in the model. The computational results from time-domain CFD approach basically agree with experimental results available in the literature with gas flow, but there are some discrepancies at low frequencies. Finally, the effect of oblique termination on the the sound reflection coefficient of circular duct is studied numerically and discussed.

MONDAY MORNING, 14 MAY 2012

S226, 11:00 A.M. TO 12:40 P.M.

Session 1aED

Education in Acoustics: Teaching Acoustics on Both Sides of the Pacific I

Siu Kit Lau, Cochair
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Xiaojun Qiu, Cochair
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Invited Papers

11:00

1aED1. Comparing two acoustics degree programmes in China and UK. Y.W. Lam and F. F. Li (Acoustics research Centre, University of Salford, Salford, Greater Manchester M5 4WT, UK, y.w.lam@salford.ac.uk)

With the current trend of education globalisation, the past decade has seen a tide of student migration and exchanges across the Pacific Ocean, evidenced by a large number of students from Asia-Pacific regions pursuing their university degrees overseas. Comparatives pedagogical studies become timely, especially in niche science and engineering disciplines that tend to have rigorously specified

prerequisites that are likely to impose challenges on student exchange and the design of joint programmes. Taking a case study approach, this paper compares syllabi and pedagogical practices in acoustics degree programmes between two representative and reputable institutes in China and the UK, with the aims to promote good practice, suggest necessary harmonisation of syllabi in order to facilitate student exchange and possible exchange and joint programme schemes. This should be of interest to those who teach acoustics and related subjects in higher education or students who intend to participate in an exchange programme to study abroad. The result from this study shows that the current acoustics degree programmes in China and in UK are generally compatible. However discrepancies in pedagogical approaches and the command of foreign language(s) mean that students will need to be prepared to quickly adapt to a different environment.

11:20

1aED2. Incorporating real-world measurement and analysis experiences in the teaching of advanced acoustics. Scott D. Sommerfeldt, Kent L. Gee, and Tracianna B. Neilsen (Brigham Young University, Provo, UT 84602, scott_sommerfeldt@byu.edu)

In the teaching of advanced undergraduate and graduate-level acoustics, rigorous mathematical presentation and extensive homework sets are the norm. However, students often fail to see the connection between theoretical models and appropriate application to “real-world” situations. Consequently, efforts have been made in courses at Brigham Young University to find collaborative measurement and analysis opportunities that help bridge this gap. Although this effort is still in its infancy, three examples are discussed in this paper. The first was measurements of skateboarding park noise levels in a nearby neighborhood. The second involved analysis of the sound system and crowd noise levels inside and outside the Brigham Young University football stadium. The third example discussed was a graduate course project to assess feasibility of creating active zones of silence in a data center. Lessons learned by students (and faculty!) are described.

11:40

1aED3. The course of Theoretical Acoustics in Graduate University of Chinese Academy of Sciences. Hailan Zhang (Institute of Acoustics, Chinese Academy of Sciences, State Key Laboratory of Acoustics, Beijing 100190, China, zhanghl@mail.ioa.ac.cn)

Theoretical Acoustics has been a course in Graduate University of Chinese Academy of Sciences since it was founded in 1978. The course covers basic theories of vibration and acoustics. The 120 hour course is given in 2 terms of the first year. Every year 50-60 students from different institutes attend the course with different mathematical and physical background. One feature of the course is the application of the functional analysis theory. The common ground of the vibration of the coupled multi freedom system, string, membrane and room is extracted and a uniform theory of vibration is presented in the form of the operator theory. Besides, many numerical results of acoustic fields, especially the transient fields, given in the course provide more intuitive understanding and help students learn the physics better.

12:00

1aED4. Telecom, Electroacoustics and Audio (TEA) education in two prestigious universities in Taiwan. Mingsian R. Bai (Power Mechanical Engineering, National Tsing Hua University, Taiwan, ufo740912@yahoo.com.tw)

This presentation gives an overview of the acoustics education by the author’s 21-year career in National Chiao-Tung University and National Tsing Hua University in Taiwan. Although it is generally recognized that acoustics is an “old” subject in classical physics, it finds many new applications in the modern world. The paradigm of acoustic education of the author is to gear the domain knowledge of acoustics to the needs of main-stream industries in Taiwan, including Computer, Community, Consumer electronics and Car, the so-called 4C industries, with emphasis placed upon telecom acoustics, electroacoustics and audio signal processing (TEA) involved in the 4C products. To meet the ever changing challenges, a multidisciplinary approach including signal processing and control system is exploited, in addition to acoustics, in the pedagogic methodology. It is hoped that, with these new perspectives, classical acoustics can be rejuvenated within unified framework. In the author’s career in education, more than 100 (including 30 in JASA) journal papers have been published, an institute of Sound and Music Innovative Technology (SMIT) and the Telecom acoustics, Electroacoustics and Audio signal processing (TEA) laboratory have been launched in NCTU and NTHU, respectively, and a monograph on acoustic array systems is currently in preparation.

12:20

1aED5. Acoustics at the Georgia Institute of Technology. Erica E Ryherd (Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405, erica.ryherd@me.gatech.edu), Mardi C Hastings, and John Doane (Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405)

Acoustics at Georgia Tech spans multiple schools, including Mechanical Engineering, Electrical Engineering, Aerospace Engineering, Biomedical Engineering, Psychology, Music, Physics, Mathematics, and Architecture. The program began over 50 years ago and strengthened considerably in the 1960s and 1970s after Eugene Patronis, Ben Zinn, and Allan Pierce joined the faculty in the Schools of Physics, Aerospace Engineering, and Mechanical Engineering, respectively. Since then hundreds of students in acoustics have graduated and hold positions in academia and industry around the world. Currently the School of Mechanical Engineering has twelve academic and eight research faculty with primary interest in Acoustics and Dynamics. Areas of research include architectural acoustics, psychoacoustics, noise control, environmental acoustics, sustainable systems design, underwater acoustics, bioacoustics, ultrasonics, active/passive control, fluid-structure interaction, nonlinear acoustics, acousto-optics, micromachined sensors and actuators, vibration of nonlinear and frictional systems, shock and vibration isolation, structural acoustics, wave propagation, and structural health monitoring. Masters and Ph.D. level programs are offered in addition to various undergraduate courses. The depth of knowledge at Tech facilitates a variety of collaborations, allowing students a multi-disciplinary education in the science and application of acoustics. Student interactions are further facilitated by a number of organizations on campus, including a student chapter of the ASA.

Session 1aHT

Hot Topics: 3-D Sound I (Lecture/Poster Session)

Yang Hann Kim, Cochair
 yanghannkim@kaist.edu

Jung-Woo Choi, Cochair
 khepera@kaist.ac.kr

Invited Papers

9:20

1aHT1. Analysis of Korean head-related transfer function. Yongwon Ju, Youngjin Park, Daehyuk Son, and Seokpil Lee (Structural Dynamics and Applied Control Lab. Dept. of Mechanical Engineering, KAIST, infinitude@kaist.ac.kr)

It is necessary to construct head-related transfer function database for rendering and studying three dimensional audio. For this reason, many research groups have tried to develop a HRTF measurement system and to construct a HRTF database for their research. Even though there are various HRTF databases, there is no database with anthropometry in public domain aimed at Koreans even if the HRTFs vary based on physical shapes of subjects. Because Koreans hear three dimensional sound rendered by HRTF database based on Caucasians, performance of three dimensional sound might be hindered. To verify this possibility and remedy the drawbacks of established HRTF database, construction of new HRTF database aimed at Korean is needed. For constructing HRTF database, new HRTF measuring system using sine sweep signal was developed and the HRTFs for 10 subjects at 49 different elevation and 36 different azimuths at 5 angular increments were measured. By using measured HRTFs, the HRTFs aimed at Koreans were compared with CIPIC HRTF database and analyzed.

9:40

1aHT2. Reproduction of immersive sound using directional and conventional loudspeakers. Ee Leng Tan and Woon Seng Gan (Nanyang Technological University, etanel@ntu.edu.sg)

Visual and audio cues play very important roles in 3D media. In such media, 3D sound effects allow game developer or a movie director to position sound effects potentially anywhere in a virtual space surrounding the viewer. Hence, accuracy of 3D sound is critical to prevent any degradation of the overall 3D experience. While there are many breakthroughs in the display technology, 3D visual content is still delivered with the current audio systems, which does not accurately deliver 3D sound. This limitation is directly linked to the dispersive nature of the conventional loudspeaker, and the reproduced 3D sound may be perceived to lack sharpness in the spatial imaging due to reverberant nature of the room acoustics. For a directional loudspeaker, the reproduced 3D sound may seem to lack spaciousness due to little influence by the room acoustics. Since most of the loudspeakers in existing sound system are dispersive in nature, 3D audio image tends to be degraded. To solve this problem, we propose a unique setup which comprises of conventional and directional loudspeakers. This setup exploits high directivity of directional loudspeakers to recreate a high quality 3D sound and to recreate the spaciousness of the audio using the conventional loudspeaker.

10:00

1aHT3. Perceptual control of convolution based room simulators. Markus Noisternig, Thibaut Carpentier, and Olivier Warusfel (IRCAM - UMR CNRS, 1 place Igor-Stravinsky, 75004 Paris, France, markus.noisternig@ircam.fr)

Reverberation processing has been intensively studied in audio and acoustics research for many years now. Early approaches used feedback delay networks to control the temporal distribution of reflections and to simulate the statistical properties of room reverberation. Thanks to the increase in processing power and the development of low-latency convolution algorithms a new generation of reverberation processors has been developed. They apply room impulse responses (RIR) measured in real concert halls and thus guarantee naturalness and authenticity of reverberation. Extending this approach to the use of higher-order spherical microphone arrays provides the means for analyzing the spatiotemporal distribution of acoustic energy. This space-time-frequency representation of the acoustic wave field is also referred to as directional room impulse responses (DRIR) in literature. The objective of the presented work is to develop a perceptually motivated signal-processing environment based on the analysis and re-synthesis of DRIRs. It first extracts perceptual features from measured DRIRs (e.g. source presence and listener envelopment) and thus provides a perceptual signature of the measured room. The room acoustic behavior can then be modified along the various perceptual dimensions, preserving the microstructure of the original RIRs, before being re-synthesized for the use with reverberation processors.

10:20

1aHT4. Perceived elevation of simultaneously presented sound sources depends upon the correlation between the source signals.

William L. Martens (Faculty of Architecture, Design and Planning, The University of Sydney, NSW 2006, william.martens@gmail.com), and Densil A. Cabrera (Faculty of Architecture, Design and Planning, University of Sydney 2006)

Speech stimuli were presented from pairs of loudspeakers placed at matched azimuth angles on either side of the listening position in an anechoic chamber. The elevation angles of the loudspeaker pairs was either 10 degrees below ear level, 10 degrees above ear level, or 30 degrees above ear level. An additional loudspeaker was placed directly above the listening position to serve as a reference for an elevation estimation task. As the correlation between simultaneously presented pairs of loudspeaker signals reproduced at a common elevation angle was decreased, the auditory source width increased to create a broad auditory image that spread out horizontally, but had a well defined apparent elevation angle. The elevation angles reported for these broad auditory images increased with an increase in the correlation between pairs of simultaneously presented speech sound sources. These results have implications for how to control most accurately the direction of multiple sources presented over 3-D arrays of loudspeakers, distributed vertically as well as horizontally.

10:40–11:00 Break

11:00

1aHT5. Investigating physical parameters associated with listeners' perceived auditory depth. Sungyoung Kim (Sound&IT Development Division Yamaha Corporation, sungyoung@beat.yamaha.co.jp), Hiraku Okumura, and Makoto Otani

Recent 3D technologies allow viewers to perceive disparities in the depths of visual objects and to thus experience more realistic visual information. As for 3D auditory display, however, conventional loudspeaker layouts have not managed to manipulate perceived auditory depth in a sufficiently convincing way. Previously, we proposed a new method that utilizes a prototype electrostatic loudspeaker that is located above the listening position and generates auditory images similar to those of headphones. Using this phenomenon and amplitude-based panning, we were able to move auditory images along the line connecting the front loudspeaker and the listening position. In this study, we investigated physical factors that were idiosyncratic in electrostatic loudspeaker reproduction and that caused listeners to perceive sounds as being nearby. We both measured and simulated the loudspeaker-to-ear transfer functions using various types of loudspeakers at multiple locations, and extracted several physical parameters, including the InterAural Phase Difference (IAPD) and the InterAural Level Difference (IALD). The result revealed a new physical quantity that was associated with loudspeaker-listener distance: variance in phase response differentials. We conclude that the electrostatic loudspeaker produced relatively less variance in phase response differentials and allowed listeners to perceive near auditory images as if listening to headphones and to enjoy better integrated 3D content.

11:20

1aHT6. Dual-layer loudspeaker array for multiple listening zones. Filippo M. Fazi, Fabio Hirono, and Philip A. Nelson (University of Southampton, University Road, SO171BJ, Southampton, UK, ff1@isvr.soton.ac.uk)

A dual-layer array consisting of sixteen small (1") loudspeakers has been built for simultaneous transmission of audio signals to multiple listeners occupying different regions of the space. The audio signals are filtered through a bank of FIR filters, computed using a Least Mean Squares (LMS) approach with regularization. The plant matrix of the array, representing the transfer functions between the loudspeakers and a set of control points, was measured in the anechoic chamber of the ISVR and was used in the filter matrix calculation. It is shown that the selection of both the number and location of the control points has direct impact on the condition number of the plant matrix, on the frequency response of the digital filters, on the frequency response of the reproduced signals, and on the acoustic radiation pattern of the array. Results are shown for several application cases, which demonstrate also the capability of controlling independently the sound radiation to the front and to the back of the dual-layer array.

11:40

1aHT7. Role of 4 - 8 kHz band component for wideband noise localization in median plane. Yukio Iwaya, Tetsu Magariyachi (Res.

Inst. of Elect. Comm., Tohoku University, 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi, Japan, iwaya@iec.tohoku.ac.jp), Makoto Otani (Shinshu Univ., 4-17-1 Wakazato, Nagano, Nagano), and Yōiti Suzuki (Res. Inst. of Elect. Comm., Tohoku University, 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi, Japan)

When we localize a sound image, interaural cues, such as interaural level differences and interaural time/phase differences, are used in horizontal plane. On the other hand in median plane localization, spectral cues are more important than that of interaural cues. However, concrete spectral cues involved in head-related transfer functions are not sufficiently investigated. To clarify spectral cues of sound localization in a median plane, we conducted a sound localization test with broadband noises with frequency spectrum manipulation. The noises were generated based on a pink noise and modified so that they had various 1 oct. band levels (-6, -3, 0, +3, and +6 dB) in 4-8 kHz band. The noises were radiated via two loudspeakers located at 30 and 60 degrees of elevations, respectively, in the median plane. Nevertheless, the perceived elevation was shifted according to the band levels. The changes of perceived elevation resembled those of relative power levels in head-related transfer functions. This suggests that the relative level of this band in the head-related transfer functions would be one of spectral cues for elevation perception.

1aHT8. New 3D audio for ultra high definition digital TV; loudspeaker configuration and method for virtual elevation effect rendering. Sunmin Kim, Young Woo Lee (Samsung Electronics, sunmin21.kim@samsung.com), Hyun Jo, Youngjin Park (KAIST), and Ville Pulkki (Aalto University)

This paper suggests the next-generation audio system for ultra high definition digital TV in terms of loudspeaker layout and corresponding rendering method. First part introduces the listening test results of perceived audio quality with several loudspeaker arrangements in order to find the optimal configuration of loudspeakers for a next-generation multichannel sound system. The subjective evaluations focused on the loudspeaker configurations at the top layer were carried out with test materials by mixing in studio and from B-format recordings. The results show that the perceptual difference in the overall quality achieved with the new 10.2-channel vertical surround system with 3 top loudspeakers and the reference system was imperceptible. Second part presents the virtual elevation effect rendering algorithm which can give a listener an impression of virtual 10.2 channel speakers using the conventional 7.1 channel speaker system (ITU-R BS.775-2) placed in horizontal plane. The proposed virtual height speaker rendering method consists of a generic head-related transfer function (HRTF) and a mixing algorithm based on four loudspeakers. For subjective evaluation three kinds of playbacks were compared; Original 10.2 channel signals, proposed 7.1 channel signals, and down-mixed 7.1 channel signals.

Contributed Papers

12:20

1aHT9. A hybrid approach for simulation of room reverberation. Junfeng Li, Risheng Xia, and Yonghong Yan (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Beisihuan Xilu, Haidian, Beijing, China, lijunfeng@hcccl.ioa.ac.cn)

Simulation of room reverberation plays an important role in room acoustics, virtual surround sound and 3D audio. Traditional reverberation simulation approaches, e.g., the geometric technique (e.g., image method) and digital signal processing-based technique, suffer from the inefficient and unnatural problems. In this paper, we propose a hybrid approach for simulation room reverberation in which the early reflections are generated using the image method with low reflection order and the late reverberation is simulated using the digital signal processing based technique. The main focus of this paper is given to realize the smooth transition from early reflection to late reverberation without any audible artifacts. Specifically, the energy decay curve (EDC) of the early reflections modeled by the image method is first formulated and subsequently exploited for late reverberation generation by the feedback delay network (FDN) approach. The subjective and objective experiments demonstrate the effectiveness of this proposed hybrid reverberation simulation approach.

The following abstracts will be presented in poster format. The posters will be on display and the authors will be at their posters from 9:20 a.m. to 5:20 p.m.

1aHT11. Horizontal and vertical sound image control using multiple parametric speakers. Kumi Maeda, Takanori Nishino, and Hiroshi Naruse (Graduate School of Engineering, Mie University, 1577 Kurimamachiya-cho, Tsu, Mie, Japan 514-8507, maeda@pa.info.mie-u.ac.jp)

Stereophonic sound systems, such as a 5.1-ch surround system, are becoming more popular because they control horizontal sound localization; however, their vertical localization remains unsatisfactory. In this paper, a method that controls a sound image with four parametric speakers are proposed and evaluated. These parametric speakers have 50 ultrasonic transducers with 10 mm diameters on the substrate and achieve super-directivity using frequency modulation. Proposed system uses sounds that are reflected on a wall and controls the sound images based on the sound level differences among parametric speakers. The sound localization performances were evaluated by subjective tests. From the results, horizontal sound localization was roughly achieved; however, vertical sound localization was difficult. [Work supported by the Hosono Bunka Foundation.]

12:40

1aHT10. Reproduction of the sound field from a virtual source inside of loudspeaker arrays. Jung-Woo Choi and Yang-Hann Kim (Korea Advanced Institute of Science and Technology(KAIST), 291 Daehak-ro, Yuseong-gu, Daejeon 305-701, Republic of Korea, khepera@kaist.ac.kr)

A sound field reproduction method for providing the auditory illusion of a virtual sound source in front of a loudspeaker array is proposed. The Kirchhoff-Helmholtz integral has been popularly used to reproduce the sound field using loudspeaker arrays, and related theories have shown that the internal sound field from virtual sources outside of the array can be reconstructed. Unlike the virtual-source-outside case, however, perfect reproduction of the virtual source inside is physically not possible because of the wavefront converging towards the location of the virtual source. The converging wavefront is one of the artifacts that always arise with artificial rendering, and the reduction of such artifact is seen as a key to reproduce the virtual source inside. For example, in the field of Wave Field Synthesis(WFS), it has been addressed that a focused source inside can be reproduced by combining time-reversal operator with the 2.5D Rayleigh integral equation. In this work, we propose three kinds of integral equation for the virtual-source-inside problem. The first equation is a generalized three-dimensional formula, and the second one is an approximated form for the far-field monopole arrays. An equation having minimal radiation property to the external field is also derived to realize the room-independent reproduction.

1aHT12. Beamforming design for linear loudspeaker array with different feeding distribution. Baoying Zhang (Beijing Institute of Technology, Information and Electronics School, Information and Communication Engineering, Grade 2009, Master, Class 2, zhangbaoying2009@163.com), and Xiang Xie (Beijing Institute of Technology, Information and Electronics School)

In the recent years, loudspeaker array has been widely considered and used in household appliance products. For the flat-panel TV, how to use the ultra-thin loudspeaker array to generate a directional beam is becoming the current research focus. In this paper, the beamforming effects of linear loudspeaker array with different feeding distributions are compared. The simulation configures 7 loudspeakers in a line with a gap of 14cm. Its directional diagrams under 1KHz are examined with 5 types of feeding distributions, which include the uniform, binomial, triangular, inverted triangular and Dolph-Chebyshev distributions. The simulation shows that the linear loudspeaker array beamforming is significantly impacted by the feeding

distributions and Dolph-Chebyshev distribution has the ideal performance in beamforming. which is because it makes a good compromise between the main lobe width and the side lobe height.

1aHT13. A simplified crosstalk cancellation method for multichannel audio equalization. Qinghua Ye, Hefei Yang, and Xiaodong Li (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, yqh@mail.ioa.ac.cn)

In deviation from the ideal listening environment, multichannel loudspeaker equalization can improve the listening experience. In this paper a multichannel equalization method based on crosstalk cancellation is

presented. The basic idea is to estimate the real and desired spatial location or acoustic transfer function for each loudspeaker, and design the equalization filters by a simplified crosstalk cancellation algorithm. The process can be divided into three steps. Firstly, the loudspeakers emit uncorrelated signals simultaneously, while the spatial location and transfer function of each loudspeaker can be measured using a binaural microphone pair. Transfer functions of other desired directions can also be measured by head rotation. Secondly, set the expected loudspeaker configuration, and get the transfer functions between the expected speakers and the listening position utilizing physical model or measuring results from previous step. Finally, the equalization filters are calculated by means of a simplified and robust multichannel crosstalk cancellation algorithm. This method can achieve equalization quickly and easily for multi-loudspeaker systems, and its effectiveness is verified by comparison with other equalization methods.

MONDAY MORNING, 14 MAY 2012

S228, 9:40 A.M. TO 11:40 A.M.

Session 1aMU

Musical Acoustics: Asian String Instruments

Chris Waltham, Cochair
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Tianreng Hua, Cochair
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Invited Papers

9:40

1aMU1. Acoustic radiation from the pipa and yueqin. Chris Waltham, Evert Koster, Andrzej Kotlicki, James Simard, and Nathan Wolfe (Department of Physics & Astronomy, University of British Columbia, Vancouver BC, Canada V6T 1Z1, *cew@phas.ubc.ca*)

The pipa and yueqin are Chinese plucked string instruments. The examples studied here have soundboards made of wu-t'ung (Paulownia) wood and each has a small cm-sized tone hole. The acoustic radiation patterns of these instruments have been measured as a function of angle and frequency. The measurements were made in an anechoic chamber and were obtained using an automated impact hammer. The radiation data are compared with measured vibration modes of the soundboxes. The spectra obtained peak at higher frequencies than are typical for Western instruments (500 Hz for the yueqin and 700 Hz for the pipa, compared to 100 Hz for the guitar), whose construction tends to emphasize the fundamental frequencies of the strings. The Helmholtz resonances of the pipa and yueqin have also been observed, but the effect of the tone holes have not been detected in the radiation data and their function is not understood.

10:00

1aMU2. Measurement and analysis of sound radiation patterns of the chinese Ruan and the Yueqin. Florian Pfeifle (University of Hamburg, Neue Rabenstrasse 13, 20354 Hamburg, Germany, *Florian.Pfeifle@haw-hamburg.de*)

The chinese Yueqin and Ruan are among the oldest documented asian string instruments. Both have a long tradition in chinese music and are commonly played in orchestral music, smaller ensembles as well as solo instruments. Eventhough both instruments have similar geometrical features, like a cylindric resonance body made out of wood, they differ in several design aspects like the fixation of the strings or the presence of sound holes. In this work the effect of these differences on the radiated and percieved sound is researched. Both instruments are measured using a 11x11 microphone array. The resulting density plots of the sound radiation over the audible spectrum are compared and analyzed.

10:20

1aMU3. Tonal features of Chinese plucked string instruments extracted from constant-Q transform spectrum. Jing Liu and Lingyun Xie (Communication University of China, 100024, small_123@hotmail.com)

The tonal features can demonstrate some musical acoustical characteristics of a musical instrument. The Constant-Q Transform (CQT) transforms temporal signal into logarithmically spaced frequency domain, which suits musical content very well. To analyze the sound of the notes played on Chinese plucked string instruments, this paper proposed an algorithm to draw the chromagram used for extracting tonal features and recorded four Chinese plucked string instruments. The classic brute force method for CQT was employed to produce the spectrograms of notes which were tuned and mapped to obtain the chromagrams. Tonal features were then extracted from them and proved to be informative for analyzing the timbre of Chinese plucked string instruments.

10:40–11:00 Break

11:00

1aMU4. Tonal characterisation of a wooden resonance box. Xiaolin Wang, Anne Shen (Key Lab of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, 21 Beisihuan Xi Lu, Beijing, China 100190, wangxl@mail.ioa.ac.cn), and Jianbo Gao (Department of Mechanical and Materials Engineering, Wright State University, Dayton, OH, 45435)

In acoustic studies of musical string instruments, it is a common practice to identify vibration modes of the instrument resonance box. Usually, when a finite element method is employed, typical shell elements instead of solid elements are used for modelling the box, and its viscous properties are typically not taken into account. Although previous researches have indicated that this does not have much impact on the calculation of vibration modes, questions arise when the characteristics of a tonal signal are to be investigated. The problem with such modelling practice is that when shell elements

are used and viscous properties are excluded, can we still effectively distinguish the subtle differences of tonal characteristics among those fine instruments. In this work we examine the effects of using different types of finite elements as well as applying viscoelastic properties of wooden materials on the tonal characteristics of sound radiation. By conducting this research for a simply structured wooden resonance box, we are attempting to answer questions such as whether we can afford to exclude either viscous properties or the use of solid elements or both in the study of tonal characteristics. If this is not permissible, then in what way these properties will affect the tonal characteristics.

11:20

1aMU5. Sound power level measurement of Chinese bowed stringed instrument-Gaoyinbanhu. Nan Li, Yuezhe Zhao, Shuoxian Wu (State Key Laboratory of Subtropical building Science, South China Univ. of Tech., 381 Wushan Road, 510640 Guangzhou, China, arlinan@scut.edu.cn), Hong Huang, and Liling Wu (Dept. of Musicology, Xinghai Conservatoire of Music, 510500 Guangzhou, China)

Gaoyinbanhu is a kind of Chinese traditional musical instrument which is popularly used in the north of China. This instrument and other two-bowed stringed instruments are adapted to play Chinese traditional musical scales and melodies which are composed with 5 notes. In this paper the sound power level measurements of Gaoyinbanhu were performed in a semi-anechoic chamber. Two professional musicians were invited to perform on their own instrument. 10-channels acoustic measuring equipments were used to investigate the sound power level and the dynamic ranges when single notes, musical scale and melodies are performed under pp, mp, f and ff dynamics. It was found that both the sound power level and its spectrum were quite close when music scale was performed under f dynamic to that when melodies were performed under normal dynamic mark. Thus the typical sound power level of Gaoyinbanhu instruments can be represented by the radiated sound power levels when musical scale was performed under f dynamic marking.

Session 1aNSa**Noise: Noise Source Localization I**

David Woolworth, Cochair
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Jun Yang, Cochair
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S.K. Tang, Cochair
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Invited Papers**9:20**

1aNSa1. Constrained beamforming for coherence sources parameters estimation. Kai Chung Tam (Department of Building Services Engineering, The Hong Kong Polytechnic University, Hung Hom, Hong Kong, jeffrey.tam@connect.polyu.hk), Siu Kit Lau (Charles W. Durham School of Architectural Engineering and Construction University of Nebraska - Lincoln, 203C Peter Kiewit Institute, 1110 S. 67th Street Omaha, NE 68182-0816), and Shiu Keung Tang (Department of Building Services Engineering, The Hong Kong Polytechnic University, Hung Hom, Hong Kong)

The phased-microphone-array acquisition-system with beamforming signal processing provides sharp directivity in receiving sound source signal from desired direction. Conventional power beamforming algorithm makes use of spatial power spectrum to estimate the source locations and power by adjusting the steering direction of the focusing beam. Although such power beamforming method shows success, there is still a research gap of power distortions caused by coherent sound fields. Furthermore the coherences between sources are still unable to identify, which could give more detailed investigation of the source characteristics. We proposed a novel beamforming algorithm which is complex signal basis instead of signal power output basis in order to explore the phase information of the sources, moreover the coherence interference is eliminated by applying linear constraints which enhance the accuracy of the source-parameter estimation. The algorithm is further validated by numerical simulations with multiple coherence sources.

9:40

1aNSa2. Decomposition of moving vehicle noise with dynamic transfer model and acoustic holograph. Sifa Zheng, Peng Hao, and Xiaomin Lian (Tsinghua University, 100084, zsf@tsinghua.edu.cn)

Identifying the moving noise of vehicles is the important step to make the optimal countermeasures. A dynamic transfer path model was proposed to describe the relation between the sources on the vehicle and the test point on the ground. The noise signals both from the vehicle and the pass-by test were simultaneously recorded with a wireless device. The parameters in the dynamic model were estimated, and the results with singular value decomposition and Tikhonov regularization were compared using simulation and experiment. The contribution of the sources was decomposed with the dynamic transfer model and acoustic holograph at any test location. Finally, the proposed method was used to decompose the contributions of the sources in a bus. The results show: the dynamic transfer path analysis could be used to identify each of the noise sources and decompose their contribution in the pass-by test.

Contributed Papers**10:00**

1aNSa3. Noise source identification with increased spatial resolution used in automotive industry. Svend Gade and Jørgen Hald (Brüel & Kjær, Skodsborgvej 307, Nærum, Denmark, sgade@bksv.com)

Delay and sum Planar Beamforming has been a widely used Noise Source Identification Technique for the last decade. It is a quick one shot measurement technique being able to map sources that are larger than the array itself. The spatial resolution is proportional to distance between array and source and inversely proportional to wavelength, thus the resolution is only good a medium to high frequencies. Improved algorithms using iterative de-convolution techniques offers up to three times better resolution.

The principle behind these techniques is described in this paper, as well as measurement examples from the automotive industry are presented.

10:20

1aNSa4. Contribution analysis method for vehicle interior noise using independent component analysis. Hikaru Ishihara and Junji Yoshida (Osaka Institute of Technology, m1m11405@st.oit.ac.jp)

For reducing vehicle interior noise efficiently, it is necessary identifying sound sources with high contributions to the noise and countermeasuring them. However, measuring source signals are sometimes difficult depending on the type of sound source such as wind noise. In this case, obtaining the

contribution and performing effective countermeasure become difficult. In this study, we considered a contribution separation technique using only response signals by employing independent component analysis (ICA). In order to apply ICA to separate contributions of vehicle interior noise, we added two procedures to frequency domain ICA method. The first is a procedure which could calculate the amplitude of each contribution correctly in each frequency from the obtained source signal by ICA, which amplitude has arbitrary property. Second is a procedure which could solve a permutation problem of frequency domain. Next, we simulated two observation signals by mixing acceleration engine and wind noises measured in cabin to verify the method. We applied the ICA to the observation signals to calculate the contribution of each sound. As a result, the calculated contribution was almost same as the actual contribution. Consequently, the proposed ICA method was clarified to be applicable contribution separation method for vehicle interior noise.

10:40–11:00 Break

11:00

1aNSa5. An algorithm for artillery noise signal detection and classification in time-domain. Yinlong Zhou (The Third Research Institute of China Electronics Technology Group Corporation, zhouyinlong@ritvea.com.cn)

Artillery noise signal can propagate long distance. It's changed greatly in complex weather condition, and becomes difficult to detect. In addition to variety of environment noise and echo signal, the truth is that artillery noise signal becomes more difficult to detect and classify. The classic and modern spectral estimation are used to process these problems frequently, while these methods will lose detail of signal are not suitable, because of the signal is non-stationary. Wavelet analysis is likely the proper method to process non-stationary signal, while spending of computation is worth of considering for each algorithm. An algorithm for artillery noise signal detection and classification in time-domain is proposed in this paper. By processing in time-domain, we can save spending of computation. By use of short-term zero-crossing rate Z_s and short-term amplitude A_s , we won't loss detail of signal; By use of proposed "period-amplitude" P_s in paper, we can detect artillery noise signal that propagating from far away; By use of dynamic threshold and array signal processing, we can suppress effect of environment noise and echo signal. By use of sample libraries of definite physical definition in time-domain, we can classify clearly.

11:20

1aNSa6. Design-optimization of a broadband phased microphone array for aeroacoustic applications. Robert Reger, Nikolas Zawodny, Kyle Pascioni (University of Florida, 231 MAE-A, P.O. Box 116250 Gainesville, FL 32611, U.S.A., rregor@ufl.edu), Drew Wetzel (Boeing Commercial Airplanes, P.O. Box 3707, Seattle, WA 98124, U.S.A.), Fei Liu, and Lou Cattafesta (University of Florida, 231 MAE-A, P.O. Box 116250 Gainesville, FL 32611, U.S.A.)

Phased microphone arrays are commonly used in acoustic beamforming applications. While numerous beamforming algorithms have been proposed to alleviate deficiencies of the delay-and-sum approach, few studies have focused on the array design itself. In aeroacoustic applications, the most common designs are based on circularly symmetric spiral arrays devised by Underbrink (1995). The design of an array using such a method is complex and tedious due to the numerous design variables and corresponding trade-offs between resolution, sidelobe suppression, size, and cost. In this paper, a systematic design-optimization approach is described that offers several objective functions and constraints. Candidate arrays for use in the University of Florida Aeroacoustic Flow Facility (UFAFF) are designed for a broadband frequency range of 1 to 80 kHz. The results of these different cases will be compared to those of an existing array design currently used in

the UFAFF. An optimized design is selected and fabricated for characterization and testing in the UFAFF. These results and comparison are described.

11:40

1aNSa7. A new method of machinery diagnosis monitoring based on the acoustic imaging measurement. 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, yychun@mail.ioa.ac.cn)

Abstract A new method has been developed to perform a fantastic diagnosis monitoring for different kinds of machines, including small as computer, shaver, hand phone, or large as 4000hp engine, slow as a walking, or fast as flying in sky, based on acoustic imaging technology with a microphone array measurement system. With it noise field of machine can be analyzed from its sound image coupled in video in door and out door, even in some extent reverberated space. It is particularly useful to analysis large machine with multi noise source and low noise level machine. Comparing to traditional vibration measurement, this method can locate the diagnosis' center and separate every source with 15dB sub-lobe suppression.

12:00

1aNSa8. A fast and hierarchical source localization algorithm for planar spiral array implemented using GPU. Lizhi Yu (Department of Automatic Control, College of Mechatronics and Automation, Xiangtan University, Xiangtan, P.R. China, 411105, yulizhi81@126.com), Yichun Yang, and Rilin Chen (Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190)

Accurate and fast localization of multiple sources is an important issue in many applications such as fault diagnosis. The well-known steered response power (SRP) method is widely used in the source localization but exhibits high computational expense. Therefore, this paper investigates a coarse-to-fine steered response power source localization algorithm to speed up the localization process. The detailed comparisons with previous algorithms are made to demonstrate that the proposed scheme is much faster, robust, and accurate. In addition, the algorithm is implemented in real time based on CUDA frame (Compute Unified Device Architecture) using GPU with high-parallel execution.

12:20

1aNSa9. Source localization using a double three-dimensional intensity array. Sung-Kyu Cho and Jeong-Guon Ih (Dept. of Mechanical Eng., KAIST, chosk03@kaist.ac.kr)

The precision of source localization methods using an array of multiple microphones depends on the number of microphones and spacing, i.e., it requires many microphones, small spacing and large aperture. To overcome the demerit in size, cost and data processing time, a double-module array system was suggested, of which a three-dimensional intensity array consists of a module. A three-dimensional intensity vector indicating the bearing angle was estimated using a set of four microphones arranged in a tetrahedral shape. Because a microphone in the apex was used in common for two modules along with the compactness of tetrahedron, number of microphones and size could be reduced. To cover a wide frequency range, two modules had different microphone spacing to minimize the low frequency phase error and high frequency finite difference error. Three-dimensional intensity was calculated by using the Taylor series expansion. For a double-module array having 16 and 80 mm in array spacing, simulations, assuming an anechoic condition, were conducted to test performances of angle detection varying bearing angle of source location, which was 1.3 m apart from the detection module. Average error of all bearing angles was less than 2σ for 270-7800 Hz. (Partially supported by BK 21 project)

Session 1aNSb**Noise and ASA Committee on Standards: Annoyance and Health Effects I**

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Invited Papers**9:20**

1aNSb1. A large scale study of the health effects of transportation noise in Hong Kong. Kin-che Lam (The Chinese University of Hong Kong, Shatin, N.T., Hong Kong SAR, *kinchelam@cuhk.edu.hk*), A. Lex Brown, I van Kamp, TW Wong, YK Chan, MKL Yeung, A Lui, CW Law, and YT Chung

Transportation noise is a problem in many large cities with possible annoyance and health related consequences. To provide the necessary data for making informed decisions on noise control strategies, a large scale study was commissioned by the Environmental Protection Department of the Hong Kong SAR Government and undertaken by an international team coordinated by the Chinese University of Hong Kong in 2009-2010. The study was based on the interview of a total of 10,077 randomly selected households and a city-wide assessment of the exposure of the selected households to road traffic noise using the state-of-the-art noise mapping technique. Noise response was measured by an internationally standardised question on an 11-point numeric scale (ISO/TS 15666, 2003). It was very much the first ever comprehensive study of such a scale, following strict international standards, carried out in Asian countries. This paper describes rationale of the study, key research questions, sampling and questionnaire design, data validation and quality control and the overall study methodology. The key study findings are given for possible comparison with similar large scale studies. Implications for noise control based on such findings will also be discussed.

9:40

1aNSb2. International comparison of Hong Kong response to road traffic noise. A.L. Brown (Environmental Planning, Griffith University, Nathan 4111 Brisbane Australia, *Lex.Brown@griffith.edu.au*), KC Lam, I van Kamp, YK Chan, and A Lui

The association between transport related noise and community response to that of exposure has been well documented. In order to develop a baseline data set for Hong Kong and enable international comparison, a household survey was conducted territory-wide in 2009-2010. The response rate was high of 75% and a total of 10,077 households were interviewed. Noise response was measured by an internationally standardised question on an 11-point numeric scale (ISO/TS 15666, 2003). Transformations were made to the data by means of the "Miedema approach" to allow for comparisons. Estimates of the percentages of highly annoyed (%HA) at the population level were plotted against Lden and compared with both Miedema's generalized curve derived from international data sets and those produced by Phan (2008) based on Vietnamese data. The Hong Kong curve lay considerably below that of Miedema, but was comparable with Phan's curves. Personal and contextual factors related to response were: noise sensitivity, window closing and home ownership. Factors reducing annoyance were: access to a quiet room in the dwelling, satisfaction with environmental circumstances in the immediate residential area and the number of households in the living quarters. These findings are well in line with those elsewhere.

10:00

1aNSb3. Sleep-disturbance and quality of sleep in Hong Kong in relation to night time noise exposure. Irene van Kamp (MGO, RIVM, PoBox 1, 3720 BA, Netherlands, *irene.van.kamp@rivm.nl*), K.C. Lam, A.L. Brown, T.W. Wong, and C.W. Law

Sleep disturbance is a main aversive effect of night time noise exposure; there is ample evidence that night time transport noise leads to acute effects such as physiological response, arousal, awakening, sleep stage changes, and amount of total sleep. Indirect effects as sleep disturbance, reduced performance and concentration have also been established. However, the long term effect of these changes is still unclear and highly hypothetical. As part of the Hong Kong transportation noise study, sleep quality was measured by means of two widely used instruments: a one question 11-point sleep disturbance scale and the Groninger Sleep Quality Scale (GSKS). Results show that 30% scores above 3 on the GSKS, indicating this be a matter of concern in Hong Kong, especially among residents of more exposed housing estates. However, this effect is not reflected in the percentage of highly sleep disturbed by road traffic noise. International comparison actually shows a lower curve in Hong Kong compared to elsewhere. Other noises were identified as sources of sleep disturbance in the survey. The influence of personal and contextual factors is highly comparable to those found elsewhere for annoyance, which includes noise sensitivity, access to a quiet side, density and overall residential satisfaction.

10:20

1aNSb4. A new approach to investigate annoyance responses to sound elements. Ken Hume (Metropolitan University, Mujthaba Ahtamad WMG, University of Warwick, UK, K.I.Hume@mmu.ac.uk)

Soundscape research indicates that sound perception is a complex auditory experience with emotional content and the potential for annoyance should not be measured simply in terms of loudness. However, there are limited objective tools available to investigate annoyance and the relative health implications of negative soundscape elements. As part of a Positive Soundscapes (UK) project, the physiological responses [heart rate (HR), respiratory rate (RR) and electromyography (EMG)] to soundscape elements were compared with the subjective assessment of pleasantness and arousal (assessed on 9 point scales) evoked in 80 subjects who listened to 18 x 8 second sound-clips. The data were analyzed via a linear mixed-model ANOVA. Listening to sound-clips lowered HR slightly but significantly. More unpleasant sound-clips caused larger falls in HR. Listening to a sound-clip raised RR slightly but significantly. The more pleasant the sound-clip was judged the greater was the rise in RR. The EMG tended to be raised by unpleasant sound-clips. Therefore, distinctive significant relationships were found between physiological measurements and the subjective estimates of pleasantness for the sound-clips presented. Therefore, an objective technique could be developed for sound engineering which allows for the potential investigation and assessment of annoyance levels to various sound elements.

10:40–11:00 Break

11:00

1aNSb5. The application of a notice-event model to improve classical exposure-annoyance estimation. Peter Lercher (Division of Social Medicine, Medical University of Innsbruck, Sonnenburgstrasse 16, A-6010 Innsbruck, Peter.Lercher@i-med.ac.at), Annelies Bockstael, Bert De Coensel, Luc Dekoninck, and Dick Botteldooren (Acoustics Research Group, Ghent University, Belgium)

Sound perception of humans is determined by a variety of factors such as intensity, frequency, temporal structure, masking and localization. Furthermore, a wide range of non-acoustical factors determine whether certain sounds are perceived as annoying. However, classical exposure-response determination for the assessment of annoyance and health effects is based on average sound levels - sometimes with applied penalties for evening and night noise (Lden). A research collaboration between Ghent University and the Medical University Innsbruck focuses on the improvement of exposure-annoyance modeling by including characteristics of the temporal structure and the attention of the involved human subjects. The basis for this work is the developed "notice-event-model" (De Coensel B et al. 2009). Intensive traffic modeling as input for extended individual noise mapping per dwelling allows to test the additional impact by the inclusion of derived acoustical indicators of the temporal pattern (Fluctuation, emergence) of the main sources (highway, main road, railway) and the human activity pattern to accommodate for masking and habituation (e.g. Notice Sound Exposure Level, notice time). This improved exposure assessment is compared with the existing classical exposure-response information from two large-scale surveys in Austrian alpine valleys. The results show that this approach is promising - but further development is needed.

11:20

1aNSb6. Development of long-term data acquisition system of noise exposure and personal behavior for analysis of health risk: Research background. Hiroyuki Imaizumi (National Institute of Advanced Industrial Science and Technology (AIST), 16-1 Onogawa, Tsukuba, Ibaraki 305-8569 Japan, hiroyuki.imaizumi@aist.go.jp), Kazutoshi Fujimoto (Kyushu University, 6-10-1 Hakozaki, Higashi-ku, Fukuoka, 811-8581 Japan), Ken Anai (Kyushu Institute of Technology, 1-1 Sensui-cho, Tobata, Kitakyushu, 804-8550 Japan), and Yasuhiro Hiraguri (Kyushu University, 6-10-1 Hakozaki, Higashi-ku, Fukuoka, 811-8581 Japan)

Since people living in urban areas are continuously exposed to loud environmental noises for a long duration, the noise has to be treated not only as nuisance in our daily lives and adverse psychological effect but also as possible risk on health. WHO has presented an environmental noise guideline and has suggested dangers or risks on health by long-term high noise exposure, and has recently published nighttime noise guideline to prevent adverse health effect to sleep disturbance. Some research projects in EU have revealed that detailed measurement in time of individual noise exposure is needed to improve the current assessment method, instead of those based on energy-averaged value over the exposed duration to noise. It suggests necessity of short time-interval measurement of individual noise exposure as well as information when and where people are exposed to the noise. It is also necessary to measure environmental condition in nighttime, since the condition very likely disturbs our sleep and therefore gives some effects to our health. From these circumstances and relating issues in Japanese, we have established a new research project which aims to investigate the effect of individual noise exposure on health. This report presents the research background and objectives.

11:40

1aNSb7. Development of long-term data acquisition system of noise exposure and personal behavior for analysis of health risk: Preliminary studies. Kazutoshi Fujimoto (Kyushu University, 6-10-1 Hakozaki, Higashi-ku, Fukuoka, 812-8581 Japan, fujimoto@arch.kyushu-u.ac.jp), Hiroyuki Imaizumi (National Institute of Advanced Industrial Science and Technology (AIST), 16-1 Onogawa, Tsukuba, Ibaraki, 305-8569 Japan), Ken Anai (Kyushu Institute of Technology, 1-1 Sensui-cho, Tobata-ku, Kitakyushu-city, Fukuoka, 804-8550 Japan), and Yasuhiro Hiraguri (Kyushu University, 6-10-1 Hakozaki, Higashi-ku, Fukuoka, 812-8581 Japan)

To investigate relationship between individual noise exposure and the effect on health, firstly we have designed a measuring equipment because commercially-available noise exposure meter adopts the averaging times of a few minutes that are longer for our purpose. Requirements of the equipment we focus on are to measure (1) intermittent characteristic of noise that include the maximum level, the number of event, and level difference between background and target noises for sleep disturbance, and (2) equivalent continuous A-weighted sound pressure levels during 24h for physiological effect. Environmental condition especially in nighttime and position where people are exposed to noise are also important parameters to be taken into consideration. In addition to the noise exposure meter, the measuring equipment developed includes thermohygrometer, illuminator and smartphone. We pursue portability and simplicity throughout the equipment. We suppose that subjects put on the noise exposure meter and the smartphone on their waist in daytime. Accuracy of the noise exposure levels at the waist is examined by simultaneous measurement near the ear at various scenes in our daily life. Long-term measurement of individual noise exposure that several subjects participate in is performed to verify the equipment. This report presents the results of preliminary measurements.

12:00

1aNSb8. Development of long-term data acquisition system of noise exposure and personal behavior for analysis of health risk: Measuring equipments. Yuichi Yonemoto, Masaharu Ohya (RION Co., LTD., y-yonemoto@rion.co.jp), Hiroyuki Imaizumi (National Institute of Advanced Industrial Science and Technology), Kazutoshi Fujimoto (Kyusyu University), Ken Anai (Kyusyu Institute of Technology), and Yasuhiro Hiraguri (Kyusyu University)

We have built a trial prototype of a long-term data acquisition system of individual noise exposure (hereafter referred to as HIKE) and have carried out technical verification of the system. HIKE consists of a noise exposure meter, a thermohygrometer, an illuminometer, and a smartphone. HIKE continuously measure $L_{Aeq,1s}$ and global positions of subjects in daytime, and in addition environmental parameters such as atmospheric temperature, relative humidity and illumination in nighttime. The noise exposure meter should be as small and light as possible for the portability and equip longer battery-life to realize the long-term data acquisition. Global positioning system on the smartphone is utilized to detect the position of subject, and we have newly developed original software for integrating the functions of collecting, storing, and displaying all data measured on the smartphone. Wireless network is applied to connect the smartphone with other measuring equipments for convenience of long-term measurement, and to accumulate all data on a database server successively and prepare for health risk analyses. This report presents the system specification and the technical considerations for setting up HIKE.

Contributed Paper

12:20

1aNSb9. Vibration and noise induced sleep disturbance from freight trains – an experimental study. Michael Smith (Occupational and Environmental Medicine, The Sahlgrenska Academy, Gothenburg University SE 405 30 Gothenburg, michael.smith@amm.gu.se), Mikael Ögren (The Swedish National Road and Transport Research Institute; SE 40278 Gothenburg), and Kerstin Persson Waye (Occupational and Environmental Medicine, The Sahlgrenska Academy, Gothenburg University SE 405 30 Gothenburg)

A substantial increase in transportation of goods on railway networks may be hindered by public fear of annoyance and sleep disturbance due to a corresponding increase in vibration and noise. As the majority of freight

trains run during night time, sleep disturbance is expected to be the most serious adverse health effect arising from resulting vibration and noise. However, very little data exists that may be used to investigate the potential impact. As part of the European project Cargovibes, we are experimentally investigating sleep disturbance. An initial pilot study explored the relative perception of horizontal versus vertical vibration for subjects in a supine position and a following study investigated the relationship between various levels of horizontal vibration and sleep disturbance. Sleep was assessed using polysomnography and questionnaires. In total 12 subjects slept for six nights in the sleep laboratory, with one adaptation night, one control night and four nights with a variation of vibration exposures maintaining the same noise exposure. The results will be discussed at the conference.

MONDAY MORNING, 14 MAY 2012

THEATRE 2, 9:20 A.M. TO 12:40 P.M.

Session 1aNSc

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

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Invited Papers

9:20

1aNSc1. Construction and performance of Japanese low noise pavements. Hitoshi Fujita (The Nippon Road Co., LTD 1-6-5 Shim-bashi Minato-ku Tokyo, hotoshi.fujita@nipponroad.co.jp)

In Japan, there have been growing demands for the protection and improvement of roadside living environment. One of the environmental problems is traffic noises. Porous asphalt has such functions as permeability and tire noise reduction, therefore has rapidly spread due to contribution to safe driving and improvement of environment around roads. Its annual achievement area in Japan estimated at more than 30 million square meters. Furthermore some types of low noise pavements such as porous cement concrete, thin layer overlay, SMA and porous elastic road surface are developed in Japan. This paper provides an overview of Japanese developments and experiences about construction and performance of low noise pavements.

9:40

1aNsc2. Noise-reducing asphalt rubber surfaces in China. George Way, Jorge Sousa (Consulpav USA, wayouta@cox.net), Rongji Cao (Jiangsu DOT China), and Krishna Biligiri (Arizona State University)

Beginning in about 2004 the state-of-the-art and practice of asphalt rubber (AR) surfaces as used and applied in the United States (US) to reduce traffic noise was presented in China. AR surfaces have been used in US for many years to reduce the traffic noise. The AR surfaces can be applied as the final wearing course on concrete or asphalt pavements. Both the US states of California and Arizona and others have successfully employed thin wearing courses (12.5 to 40 mm) of AR to reduce highway noise. Following this early exposure to the AR technology, China began to experiment and later use AR to reduce noise, as well as to provide a durable and good quality skid resistant wearing course. This paper reports on the progress of use of AR in China and the noise data from various surfaces in China. These surfaces include Stone Mastic Asphalt, Polymer Asphalt, Asphalt Rubber Asphalt Concrete and Asphalt Rubber Open Graded Friction Course.

10:00

1aNsc3. Innovative low noise road pavement materials studied in Portugal. Elisabete Freitas, Joel Oliveira, and Pedro Machado (Universidade do Minho, Campus de Azurém 4800 058 Guimarães Portugal, efreitas@civil.uminho.pt)

This paper deals with materials not conventionally used in road layers but widely used in the building construction to reduce noise. These materials are expanded clay aggregates and cork granulates. The former is characterized by a high porosity and therefore used in a surface course to partially absorb the noise and the latter is characterized by a resilient behaviour and thus used in the binder course, to cushion the vibrations originated at the top of the pavement on the vehicles movements. Their mechanical and acoustic behaviour must be proved in laboratory before construction in real scale and surface characteristics such as skid resistance must keep a high level along time. This paper addresses particularly these issues. The first results are very encouraging. When compared to equivalent conventional mixtures, their mechanical properties obtained from laboratory tests have improved. Acoustic properties, such as noise absorption, and acoustic related properties, such as those extracted from mechanical impedance tests, have also indicated a superior performance.

10:20

1aNsc4. The importance of the bottom layer in double-layer porous asphalt for noise reduction. Ulf Sandberg (Swedish National Road and Transport Research Institute (VTI), SE-58195 Linköping, Sweden, ulf.sandberg@vti.se), and Piotr Mioduszewski (Technical University of Gdansk, ul. G. Narutowicza 11/12, PL-80952 Gdansk, Poland)

Double-layer porous asphalt concrete (DPAC) surfaces are generally considered to be the acoustically most effective low noise road surfaces ready for implementation. While DPAC used on highways in warm climates may have an average life of around 8 years, in Scandinavia with severe winter climate DPAC usually survive only about 3 years; partly due to wear of studded tyres. An ongoing project in Sweden, applying DPAC and single-layer porous asphalt (PAC), the latter consisting of the top layer of the DPAC, on motorway E4 in Jönköping-Huskvarna, has revealed interesting performance. Initial noise reduction was 7.5 dB(A) compared to a set of reference surfaces (conventional SMA 0/16). Amazingly, after one year of operation this noise reduction is unchanged. Most interesting is that the noise reduction difference between the single-layer and double-layer PAC is approx. 5 dB(A). Since the single-layer PAC is identical to the 30 mm thick top layer in the DPAC (although 5-8 mm thicker), it follows that 2/3 of the noise reduction is due to the bottom layer of the DPAC; i.e. what lies approx. 35-40 mm below the top surface. The paper will discuss the effect of the bottom layer on the overall acoustical efficiency of the DPAC.

10:40–11:00 Break

11:00

1aNsc5. Ultra long life low noise porous asphalt. D. Alabaster (New Zealand Transport Agency, David.Alabaster@nzta.govt.nz), P.R. Herrington (Opus International Consultants), and J. Waters (Fulton Hogan Ltd)

This New Zealand laboratory study and field trial forms part of a larger collaborative research programme conducted under the auspices of the OECD/ECMT (European Conference of Ministers of Transport) Joint Transport Research Centre, focused on the economic evaluation of long-life pavements. The aim of the research was to investigate the potential of epoxy-modified asphalt as a low-maintenance long-life (>30 years) low noise surfacing material. The New Zealand Transport Agency's contribution to the research focused on the potential benefits of epoxy-modified open-graded porous asphalt (OGPA). Investigations into the cohesive properties and oxidation resistance of an acid-cured, epoxy-modified OGPA were undertaken, and an associated field trial constructed on State Highway 1 in Christchurch in December 2007. Results from the Cantabro Test at 10° C indicated that the cohesive properties of the oxidised epoxy OGPA were markedly superior to those of conventional OGPA. On the basis of the Cantabro test results, lifetimes of up to 144 years were estimated for an increase in cost of up to 2.3 times that of conventional OGPA. Similarly, the fatigue life of oxidised epoxy OGPA was found to be more than 25 times that of the control. Experiments were also conducted with epoxy modified bitumen diluted with up to 75% standard 80–100 penetration grade bitumen, as a possible means of reducing costs. OGPA made with the 75% diluted material had an estimated life of up to 93 years for 1.3 times the cost of conventional OGPA. The fatigue lives of the oxidised diluted OGPA mixes were similar to that of the control. An initial CAPTIF trial and a later field trial demonstrated that full-scale manufacture and surfacing construction with epoxy OGPA, could be easily undertaken without any significant modification to plant or operating procedures. A road trial to evaluate (undiluted) epoxy OGPA sections with 20% and 30% air voids was constructed and initial noise monitoring using the statistical pass by method has produced good results. The trial has been in place for almost 4 years and is performing well.

1aNSc6. Which is a better metric - road or air temperature - in assessing temperature effects on tyre/road noise? Wing-tat Hung (The Hong Kong Polytechnic University, CSE, Hung Hom, Kln, HK, cewthung@polyu.edu.hk), Yat-ken Lam (Department of Civil and Structural, The Hong Kong Polytechnic University), Randolph Chi-kin Leung (Department of Mechanical Engineering, The Hong Kong Polytechnic University), and Chung-fai Ng (Department of Civil and Structural Engineering, The Hong Kong Polytechnic University)

The temperature effect on tyre/road noise level has been vigorously studied recently. Recommendation of temperature correction is being made in the draft ISO 11819-2 for tyre/road noise measurement. Air temperature is used as the basic metric for temperature. However, road surface temperature appears to be a more direct metric and now is equally easy to measure as air temperature. To assess which is a better temperature metric, CPX tyre/road measurements with a pair of Standard Reference Test Tyre (Uniroyal Tigerpaw 225/60-R16) running on a number of stone mastic asphalt and polymer modified porous asphalt road surfaces at reference speed 50 km/h were made in day and night time. Instantaneous air and road surface temperatures were also recorded during the measurements. Results show that tyre/road noise decreases as temperatures increase. The correlations between the noise level and air or road surface temperatures varied between road sections. The temperature coefficients derived using the road temperature have smaller variation than that estimated using the air temperature.

Contributed Papers

11:40

1aNSc7. Long term noise performance of road surfaces in urban environment. YK Lam (Department of Mechanical Engineering, The Hong Kong Polytechnic University, lamyatken@yahoo.com.hk), IWK Ng (Environmental Protection Department, Hong Kong SAR), and WT Hung (Department of Civil and Structural, The Hong Kong Polytechnic University)

Noise reduction performance of road surfaces is of great concern as it has direct impact on the cost-effectiveness of this measure for noise abatement purpose. Over 70 low noise road surfaces, mainly polymer modified porous asphalts, were laid on low speed streets (speed limit < 50 km/h) in the urban areas of Hong Kong. A single-wheeled CPX trailer fitted with a local commonly fitted tyre, the yokohama tyre, was used to measure the tyre/road noise on eight sections of stone mastic asphalt surfaces for over one year and over twenty sections of polymer modified porous asphalt surfaces from one to three years. While the monitoring work is going on, initial results show that the tyre/road noise on stone mastic asphalt surfaces and that on polymer modified porous asphalt were different, and had different aging effect in noise terms.

12:00

1aNSc8. Temperature effects on tyre/road noise on wearing course and stone mastic asphalt surfaces in Hong Kong. W.T. Hung (CSE, HKPolyU, Hung Hom, Kln., cewthung@polyu.edu.hk), Y.K. Lam (ME, HKPolyU, Hung Hom, Kln), and E.K.Y. Kam

To assess the impact of temperature on tyre/road noise, two sets of tyre/road noise survey were conducted; one in the day time and the other in the night time. A twin-wheeled CPX trailer fitted Standard Road Testing Tyre (SRTT) was employed to measure the tyre/road noise. Four stone mastic asphalt and four wearing course surfaces which are common in Hong Kong were chosen for this study. The surveys were conducted in the late summer

of Hong Kong in 2011. At least four runs were made on each road section in each set of the survey. It was found that the tyre/road noise is sensitive to both air and road surface temperatures on the four wearing course and four stone mastic asphalt surfaces. The SMA surfaces are more sensitive than WC surfaces. The air temperature coefficient ranges from -0.122 to -0.462 for the four WC surfaces and from -0.265 to -0.945 for the four SMA surfaces. The road temperature coefficient ranges from -0.030 to 0.086 for four WC surfaces and from -0.056 to -0.139 for SMA surfaces.

12:20

1aNSc9. A Study on the acoustical properties of road surfaces of recycled CFB materials. Ha Ngo, Zhuang Li (Department of Engineering, McNeese State University, Lake Charles, LA 70609, msu-hngo@student.mcneese.edu), and Alan Davis (Industrial Executives and Academic Partnership (IEAP) Group, Sulphur, LA 70663)

Traffic noise and noise control are major concerns of transportation, as noise and vibration will cause both psychological and physiological consequences. Great efforts have been made to use more sound absorbent road surfaces in order to reduce traffic noise. The raw materials under study are recycled byproducts from circulating fluidized bed boiler (CFB). The recycled CFB materials have been approved for use by the Environmental and Transportation Departments in various regions throughout the United States for road stabilization and base/surface installations. These (CFB) materials have shown good ecological, civil and mechanical properties, and are more environmentally friendly than asphalt and concrete. However, the acoustical properties of the pavements are not known. Two types of measurements have been conducted. First, the traffic noise was measured using the statistical pass-by method on various road surfaces and a comprehensive comparison was conducted. Second, the sound absorption coefficients of the CFB materials were measured using impedance tubes.

Session 1aPA

Physical Acoustics: Sonoluminescence (Lecture/Poster Session)

Lawrence A. Crum, Cochair
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Invited Papers

9:20

1aPA1. Sonofragmentation and sonocrystallization. Kenneth S. Suslick (University of Illinois, 600 S. Mathews Av., Urbana, IL 61801, *ksuslick@illinois.edu*), and Brad W. Zeiger (University of Illinois, 600 S. Mathews Av., Urbana, IL 61801)

Developing processes for the production of active pharmaceutical ingredients (APIs) with a specific crystal size or polymorph distribution is critical for improved drug delivery by aerosolization, injection or ingestion, for control of bioavailability, and for economy of preparation. The use of ultrasound for the crystallization of APIs has attracted substantial recent attention due to (1) its influence on particle size and size distribution, (2) reduction of metastable zone-width, induction time, and supersaturation levels required for nucleation, (3) improved reproducibility of crystallization, (4) control of polymorphism, and (5) reduction or elimination of the need for seed crystals or other foreign materials. Possible mechanisms for the breakage of molecular crystals under high-intensity ultrasound were investigated using acetylsalicylic acid (aspirin) crystals as a model compound for active pharmaceutical ingredients. Surprisingly, kinetics experiments ruled out particle-particle collisions as a viable mechanism for sonofragmentation. Two other possible mechanisms (particle-horn and particle-wall collisions) were dismissed on the basis of decoupling experiments. Direct particle-shockwave interactions are therefore indicated as the primary mechanism of sonofragmentation of molecular crystals.

9:40

1aPA2. Numerical simulations of oriented aggregation of sonochemically synthesized BaTiO₃ nanocrystals. Kyuichi Yasui and Kazumi Kato (National Institute of Advanced Industrial Science and Technology (AIST), 2266-98 Anagahora, Shimoshidami, Moriyama-ku, Nagoya 463-8560, Japan, *k.yasui@aist.go.jp*)

Numerical simulations of sonochemical production and aggregation of BaTiO₃ nanocrystals have been performed under the experimental condition of Dang et al. [Jpn.J.Appl.Phys. 48, 09KC02 (2009)]. The theoretical model used in the simulations consists of three processes: chemical reactions, nucleation, and aggregation. The experimental data of the particle (aggregates) size distribution have been reproduced only when aggregation occur only for primary particles (nuclei). In the experiment of Dang et al., aggregates of sonochemically synthesized BaTiO₃ nanocrystals were mesocrystals. For mesocrystals, the crystal axes of nanocrystals in an aggregate are aligned. In order to study the mechanism of mesocrystal formation, electric dipole-dipole interaction model has been studied in the present numerical simulations of collisions between two particles. It has been shown that primary particles aggregate with other particles and that the crystal axes are aligned by the dipole-dipole interaction. On the other hand, large aggregates do not aggregate due to the repulsive double-layer interaction which is stronger for larger particles. The results are consistent with the above simulations on the particle size distribution. It suggests that sonochemically synthesized 5 nm BaTiO₃ nanocrystal may have spontaneous polarization.

10:00

1aPA3. Nonlinear bubble dynamics of cavitation. Yu An (Department of Physics, Tsinghua University, Beijing 100084, China, *anyuw@mail.tsinghua.edu.cn*)

A theoretical framework for studying cavitation dynamics is revived. It consists of a nonlinear sound wave equation in an acoustic cavitation environment together with the bubble motion equation. The nonlinear sound wave equation considers time delayed bubble-bubble interaction. For cavitation clouds generated in a standing sound wave driven by an ultrasonic horn, the equations are numerically solved under an approximation. It is found that the number density of bubble is a key parameter in describing the bubble dynamics of cavitation. Adjusting this parameter, our calculation may produce the chaotic acoustic pressure and various different forms of bubble motion in cavitation cloud, and can qualitatively reproduce experimentally observed phenomena.

10:40–11:00 Break

10:20

1aPA4. Self-nucleated nuclear effect of acoustic cavitation in focusing acoustic field. Qian Cheng (Institute of Acoustics, Tongji University, Shanghai 200092, China; q.cheng@tongji.edu.cn), Xin-Nian Li (Shanghai Applied Radiation Institute, Shanghai University, Shanghai 201800, China), Meng-Lu Qian, and Yin-Guan Wang (Institute of Acoustics, Tongji University, Shanghai 200092, China)

Bubble fusion have been discussed for a decade since Taleyarkhan¹ reported that fusion could take place under the condition of acoustic cavitation at SCIENCE in 2002, and a lot of theoretical and experimental investigations have been engaged to probe the nuclear effect of acoustic cavitation (NEAC). The self-nucleated NEAC in focusing acoustic field is preliminary investigated in this paper. The 5.5 MeV alpha particles from radioactive Am241 sheets placed in heavy water for bubble nucleation are used to avoid the possible influence of external incident neutrons on the counting of emitted neutrons. In consideration of the penetration depth of alpha particles in heavy water is very small ($\sim 40 \mu\text{m}$) due to its heavily ionizing radiation, different focusing acoustic transducers are designed in order to focus the acoustic energy around Am241 sheets. The experimental datas show that the cavitation counts $C_{\text{cav.on}}$, is higher than the cavitation-free counts $C_{\text{cav.off}}$, and the neutron increment ΔC , i.e. the difference between $C_{\text{cav.on}}$ and $C_{\text{cav.off}}$, is statistically significant. Besides, the tritium content of the test liquids also increases. The experiment results verify that self-nucleated acoustic cavitation in focusing acoustic field can help to intensify nuclear effect. One of the reasons may be that the cavitation bubbles increase the collision cross section of deuterium. This work is supported by the National Natural Science Foundation of China (No. 10974145 and 10804085)

The following abstracts will be presented in poster format. The posters will be on display and the authors will be at their posters from 11:20 a.m. to 12:40 p.m.

1aPA6. Trapping of microorganism using cylindrical standing ultrasound waves and its application to water purification. Hae-Rang Hwang, Yonggang Cao (Pukyong National University, 599-1, Daeyon3-Dong, Nam-Gu, Busan, Korea, espoirang@gmail.com), Jungsoo Kim (Tongmyong University, 428, Sinsun-Ro, Nam-Gu, Busan), Moojoon Kim, and Kang-Lyeol Ha (Pukyong National University, 599-1, Daeyon3-Dong, Nam-Gu, Busan, Korea)

In biological fields, it is known that the ultrasound is useful for trapping of biological cells or microorganism. Recently, several experimental results of micro-particle trapping by acoustic standing wave fields which are formed by plane waves have been reported. In this study, we confirm that the standing waves by a cylindrical ultrasonic transducer can trap and aggregate the microorganism such as green algae, chlorella, etc. Those microorganism were trapped at specific positions determined by acoustic pressure distribution, and the density of aggregation is increased according to the lapse time after the transducer operating. The dense colony of microorganism is easily filtered out by a sieve. A water purification system using this phenomenon was designed and the efficiency was verified by considering the density change of the microorganism after the ultrasonic treatment. Consequently, it is shown that the standing wave in a cylindrical transducer can be applied to water purification.

11:00

1aPA5. Acoustic, thermal and sonoluminescence investigation of enhanced cavitation of flowing polymer- and lipid-shelled microbubbles during focused ultrasound exposures. Siyuan Zhang, Yangzi Qiao, and Mingxi Wan (Xi'an Jiaotong University, No. 28, Xianning West Road, Xi'an, Shaanxi, 710049, P.R. China, xjtusyzhang@mail.xjtu.edu.cn)

Our previous work has investigated spatial-temporal dynamics of cavitation during focused ultrasound (FU) exposures using acoustic cavitation detection and high-speed photography. In this paper, acoustic, thermal and sonoluminescence investigation of enhanced cavitation of flowing polymer- and lipid-shelled microbubbles (MBs) during FU exposures were exposed as the two types of shelled MBs and pure controls flowing through a vessel in the phantom with varying flow velocities at different acoustic power levels. Vibration characteristics of two shelled MBs and the effects of acoustic pressure threshold for destruction of the two shelled MBs on the intensity and spatial distribution of sonoluminescence and sonochemiluminescence were investigated using an intensified charge coupled device camera. The inertial cavitation dose (ICD), sonoluminescence intensity and temperature for the lipid-shelled MBs were higher than those for the polymer-shelled MBs, which were both higher than pure controls. Temperature around the vessel initially increased with increasing flow velocities of MBs, followed by a decrease of the peak temperatures with increasing flow velocities when the velocity was much higher. Meanwhile, ICD showed a trend of increases with increasing flow velocity. Thermal lesion appeared around the vessel as MBs flowing through the vessel, at which lesion was not observed originally without MBs.

1aPA7. A study of the ultrasonic preparation of neodymium doped zinc oxide nanoparticles in room temperature ionic liquid. Yuetao Yang, Hao Yang, Shuyi Zhang, and Xiaojun Liu (Institute of Acoustics, Nanjing University, yyang@nju.edu.cn)

The sonochemical process has been proved to be a useful technique for generating various nanostructured materials. The physical and chemical effects of cavitation are highly dependent on the contents of the collapsing bubble and hence on the choice of solvent. Recently, room-temperature ionic liquid (RTIL) have been developed to a central point of interest in both academia and industry. RTIL are nonvolatile, non-flammable, and thermally stable solvents. These properties make ionic liquids potentially attractive for use in sonochemical reactions. The application of ultrasound to synthesis nanomaterials in conjunction with ionic liquids, however, has been rarely exploited. In this work, a facile approach has been developed for the preparation of neodymium doped zinc oxide nanoparticles in RTIL via an ultrasonic irradiation. The morphology and properties of the products have been characterized by X-ray diffraction and transmission electron microscopy. A possible mechanism is proposed to explain the formation neodymium doped zinc oxide nanoparticles via ultrasonic irradiation.

Session 1aPP

Psychological and Physiological Acoustics and Animal Bioacoustics: Open Challenges in Auditory Scene Analysis I

Mounya Elhilali, Cochair
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Invited Papers

9:20

1aPP1. Differentiating the roles of parietal cortex, auditory cortex and the thalamus in auditory stream segregation. Rhodri Cusack (Brain and Mind Institute, University of Western Ontario, London, Canada N6A 5B7, rhodri@cusacklab.org)

In the last decade, great progress has been made in identifying neural structures that underlie auditory streaming. Regions of the auditory cortex have been implicated in macaque electrophysiology (Fishman et al. 2001, 2004; Micheyl et al, 2005, 2007), human MEG (Gutschalk et al, 2005, 2007) and fMRI (Kondo & Kashino, 2009; Deike et al, 2010; Hill et al, 2011). The parietal cortex has been implicated using human fMRI (Hill et al, 2011; Cusack et al, 2005) and MEG (Teki et al, 2011). Finally, there is intriguing data from fMRI on the involvement of the thalamus (Kondo & Kashino, 2009), and from electrophysiology on subcortical regions in the auditory periphery (Pressnitzer et al, 2008). However, the roles of these different regions are far from clear. I will report results from multi-voxel pattern analyses of fMRI data, which probe what kind of information is encoded within each brain region. This revealed markers of stream segregation in the thalamus, auditory cortex and the parietal cortex, but representation of the basic stimulus features only in auditory cortex. I will discuss the roles of the different regions in automatic and voluntary scene analysis, selective attention, and multimodal object representation.

9:40

1aPP2. Concurrent sound perception interferes with signal detection. Claude Alain (Rotman Research Institute, Baycrest Centre, 3560 Bathurst Street, Toronto, ON, Canada M6A 2E1, calain@rotman-baycrest.on.ca), and Ada Leung

The object-based account of auditory scene analysis posits that attention operates on perceptual auditory objects. An important implication of such a theory is that perception of two simultaneous auditory objects may interfere with signal detection. In a series of experiments, we show that perception of concurrent sound objects, induced by varying frequency of one tonal component in an otherwise periodic sound complex, impaired gap detection. This effect was observed for a wide range of gap duration, and was greater when the mistuned harmonic was perceived as a separate object. These results suggest that one auditory object is processed at a time, which is consistent with the object-based theory. The impaired gap detection in the mistuned harmonic could be interpreted in terms of competition for attention between the gap and the mistuned harmonic: The perception of the mistuned harmonic as a separate object "wins" the competition for attentional resources.

10:00

1aPP3. Stream segregation of simultaneous harmonic sounds in normal and impaired hearing. Andrew Oxenham, Christophe Micheyl, and Heather Kreft (University of Minnesota, 75 E. River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Many everyday situations involve hearing out harmonic sounds, such as voiced speech or musical notes, and following them over time in the presence of other harmonic sounds. Despite decades of research on pitch perception, it remains unclear whether the ability to hear out the pitch of one harmonic sound in the presence of others is limited by peripheral frequency selectivity or by other factors, such as phase-locking to the temporal waveform of sounds. In this study a direct test of the role of frequency selectivity was undertaken by examining the relationship between measures of frequency selectivity and measures of performance in pitch- and melody-discrimination tasks in normal-hearing and hearing-impaired listeners. Preliminary data suggest a relationship between auditory-filter bandwidths and the amount of interference produced by a harmonic-complex masker. However, some aspects of the results indicate that factors other than frequency selectivity also play an important role, particularly in the complex tasks involving melody discrimination. [Work was supported by NIH grant R01DC05216; subject recruitment was facilitated by Starkey Laboratories, Inc.]

10:20

1aPP4. The influence of perceptual organisation on an auditory context effect. Claire Chambers (Equipe Audition, Département des Etudes Cognitives, Ecole Normale Supérieure, 29 rue d'Ulm, Paris., claire.chambers@ens.fr), Sahar Akram, Shihab Shamma (Institute for Systems Research, Electrical and Computer Engineering Department, University of Maryland), and Daniel Pressnitzer (Equipe Audition, Département des Etudes Cognitives, Ecole Normale Supérieure, 29 rue d'Ulm, Paris)

Perceptual organization of auditory scenes has a well-documented effect on subjective reports of listeners. Here, we investigated whether it also influenced an auditory context effect based on pitch. Stimuli were complexes of sinusoidal components arranged to produce ambiguous pitch transitions when presented successively. The context effect was established by preceding an ambiguous test pair with tone complexes comprising frequencies between the

components of the ambiguous pair, which was found to produce a strong bias on the test [Chambers & Pressnitzer, MidWinter Meeting of the Association for Research in Otolaryngology, 2011]. We first tested whether spatial attention modulated the context effect. Two sequences inducing opposing biases were presented dichotically, followed by an ambiguous monaural test. Listeners were more likely to be biased by the attended context. Then we tested whether figure-ground segregation was required for the context effect. We embedded the context tones in random clouds of pure tones, and varied the temporal coherence between the components of the context stimuli. High coherence produced more detectability of the context and, generally, stronger context effects. Both experiments show that stream segregation of the context sequences strongly influences the resulting bias, for identical physical stimuli. This may provide an additional objective measure of streaming.

10:40–11:00 Break

Invited Papers

11:00

1aPP5. Auditory scene analysis: It's all about expectations! Mounya Elhilali (Johns Hopkins University, 3400 N Charles Street, Barton Hall, Rm 105, Baltimore, MD 21218, mounya@jhu.edu)

Cocktail parties and other complex acoustic scenes present organisms with intricate sound mixtures and configurations. Perception in these complex settings relies on tracking regularities over time of sound patterns that arise from a statistical parsing of the scene as well as priors and expectations that bias how we organize the scene into its putative sound objects. Predictions arising from these expectations and sound regularities operate differently along different acoustic and cognitive domains. Here, we discuss the role of the interplay of expectations along these different domains in mediating the organization of complex acoustic scenes.

11:20

1aPP6. Toward an integrated neurocomputational model of auditory scene analysis. Charles Delbé and Nicolas Grimault (CNRS - Univ Lyon 1 50 av T Garnier 69366 Lyon cedex 07, charles.delbe@olfac.univ-lyon1.fr)

The functional models of auditory scene analysis (ASA) available in the literature have several limitations. First, they independently implement various principles and theories that are specific to the auditory modality. Second, they rarely account for top-down, high level, cognitive effects on ASA. The present paper aims to propose a new integrated model of ASA and reports results within a connectionist modeling framework to account for a wide range of effects on auditory scene analysis. The used connectionist framework is conformed to the known functional and anatomical constraints regarding the biological principles underlying auditory processing. This new neurocomputational model is specifically dedicated to account for top-down effects on ASA, such as attentional control, long-term memory knowledge effects and cross-modal interactions.

11:40

1aPP7. A computational model for the dynamic aspects of primitive auditory scene analysis. Makio Kashino, Eisuke Adachi, and Haruto Hirose (NTT Communication Science Laboratories, 3-1, Morinosato Wakamiya, Atsugi, Kanagawa, 2430198, Japan, kashino.makio@lab.ntt.co.jp)

Recent psychophysical and neuroscientific studies suggest that auditory scene analysis is not fully determined by the spectrotemporal properties of acoustic signals, but also dependent critically on the various forms of predictions generated in the listener's brain. The predictions could be based on prior knowledge about the statistical properties of acoustic events in the real world, and regularity found in a given acoustic signal. Here, a computational model of primitive auditory scene analysis is proposed, with an emphasis on the dynamic interaction between the analysis of acoustic features and the generation of predictions. The model consists of several functional components, including: (1) the decomposition of spectrotemporal patterns into basic elements and their temporal changes, based on repetitive co-occurrence of spectral components, (2) the Bayesian inference incorporating prior knowledge and signal regularity, and (3) temporal gating using internally-generated signals. It will be examined whether the proposed model can explain the dynamic aspects of primitive auditory scene analysis, including the temporal buildup of stream segregation, multistable perception for prolonged stimulation, and the detection of repeated patterns embedded in random patterns.

Contributed Paper

12:00

1aPP8. A physiologically inspired model of auditory stream segregation based on a temporal coherence analysis. Simon Krogholt Christiansen, Morten Løve Jepsen, and Torsten Dau (Centre for Applied Hearing Research, Technical University of Denmark, DK-2800 Kgs. Lyngby, Denmark, skch@elektro.dtu.dk)

The ability to perceptually separate acoustic sources and focus one's attention on a single source at a time is essential for our ability to use acoustic information. In this study, a physiologically inspired model of human auditory processing *Jepsenet al., 2008* was used as a front end of a model for auditory stream segregation. A temporal coherence analysis *Elhilali et al.,*

2009 was applied at the output of the preprocessing, using the coherence across tonotopic channels to group activity across frequency. Using this approach, the described model is able to quantitatively account for classical streaming phenomena relying on frequency separation and tone presentation rate, such as the temporal coherence boundary and the fission boundary *van-Noorden, 1975*. The same model also accounts for the perceptual grouping of distant spectral components in the case of synchronous presentation. The most essential components of the front-end and back-end processing in the framework of the presented model are analyzed and future perspectives discussed.

Invited Paper

12:20

1aPP9. Role of coherence and rapid-plasticity in active perception of complex auditory scenes. Shihab Shamma (University of Maryland, A. V. Williams Building, College Park, MD 20742, sas@umd.edu)

Humans and other animals can attend to one of multiple sounds, and follow it selectively over time. The neural underpinnings of this perceptual feat remain mysterious. Some studies have concluded that sounds are heard as separate streams when they activate well-separated populations of central auditory neurons, and that this process is largely pre-attentive. Here it is argued that stream formation depends primarily on temporal coherence between responses that encode various features of sound source. Furthermore, we postulate that only when attention is directed toward a particular feature (e.g., pitch) do all other temporally coherent features of that source (e.g., timbre and location) become bound together as a stream that is segregated from the incoherent features of other sources.

MONDAY MORNING, 14 MAY 2012

S222, 9:20 A.M. TO 12:40 P.M.

Session 1aSA

Structural Acoustics and Vibration and Noise: Energy Based Methods in Structural Acoustics I

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Contributed Papers

9:20

1aSA1. On the measurement of angular-dependent, airborne sound transmission through finite supercritical bars: Further results. Matthew D. Shaw (Penn State Acoustics, 201 Applied Science Building, University Park, PA, 16802, mdshaw16@gmail.com), and Brian E. Anderson (Acoustics Research Group, Dept. of Physics and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT, 84602)

A method of measuring the angular dependence of sound transmission through supercritical bars in air is discussed. The coincidence effect occurs in a supercritical bar when the component of the acoustic wave number parallel to the bar matches the bending wave number in the bar. The

transmission of sound is at a maximum at the angle where this trace wave number matching occurs. The theory of the coincidence effect is well-defined for unbounded thin plates using plane-wave excitation. An experimental setup has been developed in order to observe the coincidence effect using continuous-wave excitation and phased-array methods through finite bars. Experimental results through a 0.5 mm thick aluminum bar exhibit strong maxima at the predicted coincidence angles, showing that coincidence is observable using continuous waves. Measurements of the coincidence angle at frequencies spanning from the critical frequency up to nearly three times the critical frequency have been made. A curve fit to the frequency dependent measurement of coincidence angles allows one to determine the bending stiffness of a bar of unknown material properties.

9:40

1aSA2. Dynamic behaviour and active sound radiation control of a two-stage vibration isolation system equipped on a flexible plate. Yao Sun, Tiejun Yang, Lu Dai, and Jingtao Du (College of Power and Energy Engineering, Harbin Engineering University, Harbin, Heilongjiang, 150001, China, sunyaoheu@gmail.com)

This paper presents a general model of a two-stage vibration isolation system involving a flexible base plate with arbitrary boundary conditions. The dynamic behavior of the coupling system was obtained by using an impedance method in which the impedance matrix of the base plate was derived from a combined use of an improved Fourier series expansion and the Rayleigh-Ritz method. Subsequently, with the purpose of attenuation of sound radiation from the base plate structure, the optimal force information is formulated under various active control strategies including: 1) minimizing the total vibratory power of the coupled structure, 2) minimizing the sound power radiation from the base plate, 3) minimizing the vibratory power transmitted to the base plate, and 4) minimizing the mean square velocities at the isolator locations on the flexible plate. Numerical results were presented and discussed in detail. Finally, some concluding remarks are made.

10:00

1aSA3. Sound radiation of elastically restrained stiffened orthotropic plates. Tai Yan Kam and B. Y. Lee (Mechanical Engineering Department, National Chiao Tung University, Hsin Chu 300, Taiwan, tykam@mail.nctu.edu.tw)

In this paper, a method is proposed for studying the sound radiation behaviors of elastically restrained stiffened orthotropic plates. The plate stiffened by a number of relatively thick beams on the bottom surface of the plate is elastically restrained around the periphery of the plate. In the vibro-acoustic analysis of the plate, the deformation of the stiffened plate is formulated based on the first-order shear deformation theory and the peripheral elastic restraint modeled as a set of distributed springs. The Ritz method is used to construct the equations of motion for the elastically restrained stiffened plate. The response (amplitudes and phases) at any point on the top surface of the plate subjected to harmonic excitation is used in the Rayleigh first integral to calculate the sound pressure generated by the plate. The sound pressure level (SPL) curve of the plate in the audible frequency range (20-20kHz) is constructed for the plate. The proposed method is then used to study the effects of material properties and stiffened pattern on the SPL curve of the elastically restrained stiffened orthotropic plate. The experimental SPL curves of two elastically restrained stiffened orthotropic plates are determined to verify the accuracy of the theoretical results and demonstrate the applications of the proposed method in the audio industry.

10:20

1aSA4. Power flows between strongly coupled structural components. Shiliang Jiang, Wen L Li (Wayne State University, Department of Mechanical Engineering, 5050 Anthony Wayne Dr., Detroit, MI 48202, shiliang.jiang@wayne.edu), Tiejun Yang, and Jingtao Du (Harbin Engineering University, College of Power and Energy Engineering College, Harbin, Heilongjiang province, China)

In this study, the energy distributions and power flows between some basic structural components such as beams, plates, and shells are studied using a so-called Fourier Spectral Element Method (FSEM). Similar to the SEA modeling, a complex system is here also considered as an assembly of subsystems or components. The FSEM, however, is deterministic in nature in that the solution is obtained by directly and faithfully solving the governing equations for each component under the actual boundary and coupling conditions. What make this model powerful and unique lie in its capability and flexibility of effectively dealing with model uncertainties (due to the probabilistic/stochastic natures of some model parameters) and engineering and manufacturing errors which tend to become critically important at higher frequencies. Since this method does not involve any artificial assumptions or simplifications, it potentially offers a whole frequency solution with adaptive spatial and frequency resolutions.

10:40–11:00 Break

11:00

1aSA5. Energy flux streamlines versus acoustic rays for modeling interaction with rigid boundaries: a Lloyd's mirror experiment. Cleon E. Dean (Physics Department, P.O.B. 8031, Georgia Southern University, Statesboro, GA 30460-8031, cdean@georgiasouthern.edu), and James P. Braselton (Department of Mathematical Sciences, Georgia Southern University, P.O.B. 8093, Statesboro, GA 30460-8093)

An energy flux streamline model was developed in support of a simple Lloyd's mirror experiment originally intended for use by high school students wherein 10 000 Hz harmonic sound, emitted from a roughly 10 cm diameter baffled loudspeaker, was reflected off a floor, treated as a rigid boundary. The model is used to draw out similarities and differences between energy flux streamlines and acoustic rays. Particular attention is paid to conditions and angles of reflection that hold for acoustic rays reflected from a rigid boundary versus the conditions that hold for the equivalent reflection and reflection angles of energy flux streamlines.

11:20

1aSA6. Vibration analysis of moderately thick rectangular plates with elastically restrained edges. Jingtao Du (College of Power and Energy Engineering, Harbin Engineering University, Harbin, 150001, P.R. China, jingtdu@yahoo.com), Wen L. Li (Department of Mechanical Engineering, Wayne State University, 5050 Anthony Wayne Drive, Detroit, Michigan 48202-3902), Tiejun Yang, and Zhigang Liu (College of Power and Energy Engineering, Harbin Engineering University, Harbin, 150001, P.R. China)

In this paper, an improved Fourier series method is proposed for the free vibration analysis of moderately thick rectangular plates with uniform elastic restraints along each edge. The effect of shear deformation is considered by using Mindlin plate theory (namely, the first order shear deformation theory). The transverse deflection and rotation displacement functions are invariantly expressed as the superposition of a double Fourier cosine series and four supplementary functions in the form of the product of a polynomial function and a single cosine series expansion introduced to ensure (accelerate) the uniform and absolute convergence (rate) of the series representation on the plate including four edges. The unknown expansion coefficients are determined using the Rayleigh-Ritz procedure in conjunction with the energy formulation of Mindlin plate system. Several numerical examples are presented to demonstrate the effectiveness and reliability of the proposed method for predicting the modal parameters of rectangular Mindlin plates with various thickness-length ratios under different boundary conditions. Although the constraint is considered uniformly distributed over each edge, the current method can be readily extended to the general cases when the spatial variation of the restraining stiffness is of interest.

11:40

1aSA7. The underwater vibration characteristics of double plate with periodic connectors. Liu Xiaobin and Yu Mengsa (5 department, the 702 institute, WuXi city, JiangSu province, China, liuxbin@yahoo.cn)

Double plate is widely used to isolate the vibration and noise in the air, it needs more research on the underwater application. The vibration model of double plate with periodic connectors was built up. The transmission path of power was analyzed carefully and this analysis revealed the specific frequency and wavenumber selective characteristics of double plate in comparison with the single plate. The incident sound wave and the turbulence have very different frequency and wavenumber's spectrum, this may introduce very interesting usage in the sonar dome which needs consideration of both sound wave and turbulence.

12:00

1aSA8. Transient dynamics of three-dimensional beam trusses using higher order kinematics. Yves Le Guennec and Eric Savin (ONERA – The French Aerospace Lab BP 72, F-92322 Châtillon cedex, France, yves.le_guennec@onera.fr)

Spatial structures are often subjected to impulse loads which induce high-frequency (HF) wave propagations. Despite some recent researches, the characterization of the transient response to such loads remains an open

problem. The objective of this research is to develop a reliable model of the HF energy evolution within three-dimensional beam trusses in order to predict, for example, their potential steady state behavior at late times or the energy paths. The theory of micro-local analysis of wave systems shows that the energy density associated with their HF solutions satisfies a Liouville-type transport equation. A suitable HF transport model for beams is derived from the spectrum relations for Lamb waves in the HF range. At the interfaces between substructures, the energy flow is partly reflected and partly transmitted. The corresponding reflection/transmission coefficients are also derived in this study. Numerical simulations are performed by a spectral discontinuous Galerkin (DG) method for spatial resolution and a strong stability-preserving Runge-Kutta (RK) method for time integration. Numerical results using the RK-DG method are presented for the example of a three-dimensional beam truss that exhibit diffusive behavior at late times.

12:20

1aSA9. Prediction of cavity noise with multiple layer composite plate as the back panel induced by the turbulence. Lv Shijin, Liu Xiaobin, and Yu Mengsa (5 department, the 702 institute, WuXi city, JiangSu province, China, lsj534@sohu.com)

Cavity noise in the cube with multiple layer composite plate as the back panel was predicted and analyzed in this paper. The mode solution was used to solve this acoustical problem, the mode strain and stress transmission matrix in the composite plate was built up, this paper chose the Corcos model as the turbulent boundary layer pressures's wavenumber-frequency spectrum. The cavity noise was calculated, the sound pressure level's difference between the theory and experiment was less than 3dB, the method in this paper can be used to predict the cavity noise underwater.

1a MON. AM

MONDAY MORNING, 14 MAY 2012

S425, 9:40 A.M. TO 12:40 P.M.

Session 1aSCa

Speech Communication: Speech Perception and Early Language Development: Cross-Linguistic Studies of English, Cantonese, and Mandarin

Estella Ma, Cochair
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Benjamin Munson, Cochair
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Chair's Introduction—9:40

Contributed Papers

10:00

1aSCa1. Three- to five-year-old children in Taiwan show little development in their production of monosyllabic Mandarin lexical tones. Pusan Wong (Department of Otolaryngology, College of Medicine, The Ohio State University, 915 Olentangy River Road, Columbus, OH 43212, pswResearch@gmail.com)

While a couple of studies reported that 2-year-old children in Beijing have mastered the production of Mandarin tones in various contexts, several studies found that three-year-old children learning Mandarin as a first language in the U.S. have not produced adult-like tones in monosyllabic words. This study collected monosyllabic Mandarin tone productions from 33 three- to five-year-old children growing up in Taiwan. Five judges categorized the tones of the 734 child productions and 92 productions by 4 adults via low-pass filtered words in which the segmental information was degraded while F0 information was retained. Adult tones were categorized with 93%, 96%, 82%, and 94% accuracy. Children's tones were identified with significantly lower accuracy ($p < .05$) at 63%, 50%, 50% and 77%, respectively. Age accounted for 0.2%, 2.4%, 4.0% and 8.2% of the variance in children's accuracy of the four tones, respectively, suggesting little developmental change. Children produced T4 more accurately than T1. T2 and T3 were significantly more difficult. These results are in line with findings in previous studies with children growing up in the U.S. using the same methodology and seem to support that tone development is related to maturation of speech motor control. [Work supported by NSF EAPSI]

10:20

1aSCa2. Comparing language experience and task demands in Mandarin tone processing: Neurophysiological evidence. Yan H. Yu, Valerie L. Shafer (The Graduate Center, City University of New York, 365 5th Avenue, New York, NY 10016, yanhyu@gmail.com), Elyse Sussman (Albert Einstein College of Medicine 1300 Morris Park Avenue Bronx, NY 10461), and D. H. Whalen (The Graduate Center, City University of New York, 365 5th Avenue, New York, NY 10016)

Behavioral studies have suggested that speech discrimination can operate at the acoustic/phonetic level at relatively short interstimulus intervals (ISIs < 500 ms) because the auditory trace is robust. However, with longer delays (> 1500 ms) the short-term memory trace decays, and thus, discrimination must rely on the phonemic information stored in long-term memory (Werker & Logan, 1985). To study the neurophysiology of tone perception, native Mandarin and monolingual English speakers participated in a passive oddball paradigm designed to elicit mismatch negativity (MMN). Event-related potentials were recorded from 65 electrode sites. Two tone-contrast pairs ("easy": tone 3-tone 1; "hard": tone 3-tone 2) were presented in bisyllabic nonword contexts in short and long ISI conditions. It is found that Mandarin listeners have similar amplitude MMN evoked by the easy-tone and hard-tone contrasts at both ISIs. English listeners, in contrast, have larger amplitude MMNs to the hard-tone contrast only in the short ISI condition. Further, the English-speaking group also showed a larger N1 peak amplitude in the long ISI condition compared to the short ISI or to Mandarin listeners. The results suggest that language

experience and task demands influence speech processing at both the lower sensory (indexed by N1) and higher cognitive (indexed by MMN) levels.

10:40–11:00 Break

11:00

1aSCa3. Prosodic realization of focus in Mandarin by advanced American learners of Chinese. Ying Chen and Susan Guion-Anderson (Department of Linguistics, 1290 University of Oregon, Eugene, OR 97403, ychen12@uoregon.edu)

Prosodic focus in Beijing Mandarin and American English involves language-specific patterns of expansion in duration, F0 and intensity on the focused item as well as post-focus compression (PFC) of F0 and intensity (Xu, 1999; Xu & Xu, 2005). The current study examined whether advanced American learners of Mandarin realize prosodic focus and PFC in the same way as native speakers. Ten native Beijing Mandarin speakers and ten non-Chinese American learners of Mandarin produced stimuli with four Mandarin tone types on focused constituents, and Tone 1 in pre-focus and post-focus constituents. Preliminary results indicated that the learners produced focus-related duration changes in a manner similar to native Mandarin speakers. However, learners did not show native-like patterns of in-focus changes in intensity on Tone 2, mean F0 on Tone 1, and F0 excursion on Tone 4. Furthermore, learners showed no PFC of F0 or intensity, consistent with the idea that PFC is not easily transferred from L1 to L2 (Wu & Chung, 2011). Future work will investigate prosodic focus in the Mandarin of Chinese-heritage American learners. The goal is to investigate whether earlier exposure to the language (via heritage) affects learners' ability to realize prosodic focus in a native-like manner.

11:20

1aSCa4. Phonetic characteristics cuing continuation of talking beyond possible completion in Chinese conversation. Wei Zhang, Bin Li, and Angela Chan (Dept of Chinese, Translation and Linguistics, City University of Hong Kong, Kowloon Tong, Hong Kong, weizhang@cityu.edu.hk)

One of the grossly apparent facts about conversation is that speakers take turns to talk (Sacks, Schegloff & Jefferson 1974). Both syntactic and prosodic cues contribute to the smooth transition between conversational turns (Couper-Kuhlen & Ford 2004, Ford & Thompson 1996). Two prominent and similar turn-holding devices have been identified, namely, rush-through (Schegloff 1982, 1998) and latching (Liddicoat 2007), which enable speakers to bid for turn continuation beyond possible completion of a turn. However, systematic and detailed examination of their exact phonetic design has been reported only recently for the English data (Walker 2003, 2010). In this study, data from naturally-occurring Mandarin Chinese conversations have been examined for prosodic correlates which are associated with turn continuation. These correlates include pitch variation, intensity, and vowel duration. It is found that prosodic cues vary between the two turn-holding devices. The findings have also been compared with those reported for English conversations. This research contributes to cross-linguistic investigation of the prosody that constitutes turn-holding functions in conversation. Acknowledgement: This study is supported by the General Research Fund [CityU 151408] awarded by the Hong Kong Research Grants Council.

11:40

1aSCa5. Text-independent pronunciation quality automatic assessment system for English retelling test. Yaohui Qi, Bin Dong, Fengpei Ge, and Yonghong Yan (Key Laboratory of Speech Acoustics and Content Understanding at Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei Si Huan West Road, Haidian District, Beijing, China, qiyaohui@hcccl.ioa.ac.cn)

An automatic grading system for spoken English retelling test is presented in this paper. Speech recognition technology is used in the system to

evaluate the quality of retelling according to the pre-defined scoring rubric which includes speech fluency, pronunciation accuracy and content integrity. Scoring features for these quality aspects are firstly extracted by applying LVCSR, keyword spotting, forced alignment and confidence measurements. And then, these features are mapped to a score by using SVM model which is pre-trained on human rated test items. According to the experimental results the correlation coefficient between machine scores and expert scores is 0.729, which means that the system can be used in real examination to replace human scores. This work is partially supported by the National Natural Science Foundation of China (No. 10925419, 90920302, 10874203, 60875014, 61072124, 11074275, 11161140319).

12:00

1aSCa6. Speech perception development in monolingual and bilingual infants. Adrian Garcia-Sierra, Nairan Ramirez-Esparza, and Patricia K. Kuhl (I-LABS at The University of Washington, gasa@uw.edu)

We investigated the relation between language exposure and neural commitment to the phonetic units of language in 11-14 month-old English monolingual (N=22) and English-Spanish bilingual infants (N=22). Our previous work suggested that bilingual infants develop phonetic neural commitment at a different pace than their monolingual peers (Garcia-Sierra et al., 2011). However, interpretation of the bilingual data requires testing a speech contrast that is non-native for both bilinguals and monolinguals. We assessed language exposure using LENA digital recorders. Neural speech discrimination (English, Spanish, Mandarin) was tested using event-related potentials (ERPs) to determine the Mismatch Response (MMR). Both groups showed significant correlations between MMRs and language exposure. However, monolinguals showed negative MMRs and negative correlations between MMR and exposure; bilinguals showed positive MMRs and positive correlations with exposure. Negative MMRs are interpreted as an established commitment to native speech sounds. Positive MMRs are interpreted as an initial ability to discriminate sounds. No correlations were found between Mandarin-MMRs and language exposure. Another phonetic contrast (Hindi), nonnative for both groups, is now being tested in the monolingual and bilingual children. Our results support the view that bilingual and monolingual infants show a different pattern of speech perception development.

12:20

1aSCa7. Phonetic category formation in Korean-English bilingual children. Sue Ann Lee (Texas Tech Univ Health Sci Ctr, sueann.lee@ttuhsc.edu), and Gregory Iverson (Univ of Wisconsin-Milwaukee)

This is an NICHD (RHD061527A) funded study examining vowels and stops produced by Korean-English bilingual (KEB) children at 3, 5, and 7 years of age in order to determine whether bilingual children develop single or separate linguistic systems in the learning of their two languages. Though a long-standing theoretical issue in bilingualism, the question of whether bilingual children develop one or two distinct PHONETIC systems has not been fully explored. In the present study, 55 KEB children participated who first learned Korean, then English, in the US. Word-initial VOT and f0 values in the following vowel were measured for stops in both languages, as well as F1 and F2 values for vowels. We found developmental patterns and multi-dimensional representation of phonetic categories between vowels and stops. Specifically, 3 year-old KEB children did not distinguish between English and Korean vowels or stops, whereas 5 year-olds distinguished vowels but not the stop categories of Korean and English, and 7 year-olds distinguished both vowels and stops. Results suggest that the phonetic systems of bilingual children continue to evolve during the developmental process, and that bilingual children require different durations of exposure per speech category in order to establish detailed phonetic categories across languages.

Session 1aSCb

Speech Communication: Speech Processing Potpourri (Poster Session)

Jeffrey Berry, Cochair
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Contributed Papers

All posters will be on display from 9:20 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 9:20 a.m. to 11:00 a.m. and contributors of even-numbered papers will be at their posters from 11:00 a.m. to 12:20 p.m.

1aSCb1. Entropy coding for training deep belief networks with imbalanced and unlabeled data. Jeffrey Berry (University of Arizona, Department of Linguistics, Tucson, AZ 85721, *jjberry@email.arizona.edu*), Ian Fasel (University of Arizona, School of Information: Science, Technology and Arts, Tucson, AZ 85721), Luciano Fadiga (Italian Institute of Technology, Department of Robotics, Brain and Cognitive Sciences, Genoa, Italy 16163), and Diana Archangeli (University of Arizona, Department of Linguistics, Tucson, AZ 85721)

Training deep belief networks (DBNs) is normally done with large data sets. In this work, the goal is to predict *traces* of the surface of the tongue in ultrasound images of the mouth during speech. Performance on this task can be dramatically enhanced by pre-training a DBN jointly on human-supplied traces and ultrasound images, then training a modified version of the network to predict traces from ultrasound only. However, hand-tracing the entire dataset of ultrasound images is extremely labor intensive. Moreover, the dataset is highly imbalanced since many images are extremely similar. This work presents a bootstrapping method which takes advantage of this imbalance, iteratively selecting a small subset of images to be hand-traced, then (re)training the DBN, making use of an entropy-based diversity measure for the initial selection. With this approach, a three-fold reduction in human time required to trace an entire dataset with human-level accuracy was achieved.

1aSCb2. Voice search optimization using weighted finite-state transducers. Yuhong Guo, Ta Li, Yujing Si, Jieli Pan, and Yonghong Yan (Key Laboratory of Speech Acoustics and Content Understanding, Institute of Acoustics, Chinese Academy of Sciences, *guoyuhong@hcl.ioa.ac.cn*)

Voice search system can provide users with information according to their spoken queries. However, as the most important module in this system, the high word error rate of the automatic speech recognition (ASR) part degrades the whole system's performance. Moreover, the runtime efficiency of the ASR also becomes the bottleneck in the large scale application of voice search. In this paper, an optimized weighted finite-state transducer (WFST) based voice search system is proposed. A weighed parallel silence short-pause model is introduced to reduce both the final transducer size and the word error rate. The WFST network is optimized as well. The experimental results show that, the recognition speed of proposed system outperforms the other recognition system at the equal word error rate and the miracle error rate is also significantly reduced. This work is partially supported by the National Natural Science Foundation of China (No's. 10925419, 90920302, 10874203, 60875014, 61072124, 11074275, 11161140319).

1aSCb3. Hybrid low delay frame loss concealment in an MDCT based audio codec. Zhibin Lin (Key Laboratory of Modern Acoustics, Institute of Acoustics, Nanjing University, Nanjing 210093, Jiangsu, China, *zblin@nju.edu.cn*), Ming Wu (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China), Jing Lu, and Xiaojun Qiu (Key Laboratory of Modern Acoustics, Institute of Acoustics, Nanjing University, Nanjing 210093, Jiangsu, China)

By combining tonal-dominant and noise-dominant signal frame loss concealment (FLC) approaches, a hybrid low delay FLC method is proposed for an modified discrete cosine transform (MDCT) based codec. Based on the observations that the phase of the MDCT-MDST (modified discrete sine transform) coefficients of tonal-dominant signals decreases linearly with the increase of the frame index and the amplitude keeps unchanged, the tonal-dominant signal FLC approach uses the frame interpolation to estimate the phase and magnitude of the MDCT-MDST coefficients of the lost frame while the noise-dominant signal FLC method implements a modified shaped-noise insertion. Both objective and subjective test results show that the proposed technique provides better performance than the existing methods for music signals and voiced speech signals.

1aSCb4. Multi-band speech recognition using band-dependent confidence measures of blind source separation. Atsushi Ando, Hiromasa Ohashi (Nagoya University, Furo-cho, Chikusa-ku, Nagoya 464-8603, Japan, *atsushi.ando@g.sp.m.is.nagoya-u.ac.jp*), Sunao Hara (Nara Institute of Science and Technology, 8916-5 Takayama, Ikoma, Nara 630-0101, Japan), Norihide Kitaoka, and Kazuya Takeda (Nagoya University, Furo-cho, Chikusa-ku, Nagoya 464-8603, Japan)

One of the main applications of Blind Source Separation (BSS) is to improve performance of Automatic Speech Recognition (ASR) systems. However, conventional BSS algorithm has been applied only to speech signals as a pre-processing approach. In this paper, a closely coupled framework between FDICA-based BSS algorithm and speech recognition system is proposed. In the source separation step, a confidence score of the separation accuracy for each frequency bin is first estimated. Subsequently, by employing multi-band speech recognition system, acoustic likelihood is calculated from the estimated BSS confidence scores and Mel-scale filter bank energy. Therefore, our proposed method can reduce ASR errors which caused by separation errors in BSS and permutation errors in ICA, as in the conventional approach. Experimental results showed that our proposed method improved word accuracy of ASR by approximately 10%.

1aSCb5. Analysis of sidelobe blanking technique for two-channel speech enhancement. Kaiyu Jiang, Qiang Fu, and Yonghong Yan (Institute of Acoustics, Chinese Academy of Science, 100190, jiangkaiyu@hcl.ioa.ac.cn)

This paper analyzes the application of Sidelobe Blanking Logic to Two-Channel speech enhancement. We show that several separately proposed Two-Channel post-filtering speech enhancement methods can be viewed as variants of Sidelobe Blanking Logic technique which first arose in the Radar community around the 1970s. We show that the core mechanism of this kind of technique lies in the two combined target detection measures, that is nonstationarity and Main to Auxiliary ratio. Consequently, the key role played by the detection thresholds is revealed. From this point of view, we show that a well-known two-channel post-filtering method can be improved by adapting the threshold to the main and auxiliary receiver characteristics, and simplified by using a single hard threshold and Wiener filtering instead of double thresholds and OM-LSA, without significant performance loss.

1aSCb6. Noise reduction for auditory prostheses based on harmonic detection. Ningyuan Wang and Andrew J Oxenham (Department of Psychology, University of Minnesota, wang2087@umn.edu)

Difficulty in understanding speech in background noise is one of the most common complaints of hearing-aid and cochlear-implant users. Various noise-reduction and spectral-enhancement algorithms have been designed and tested over the years, often with limited success in terms of improving speech intelligibility. A commonly used method involves spectral subtraction, which is based on the assumption that the noise spectrum can be estimated and the clean speech signal can be extracted by subtracting the noise spectrum from the noisy speech signal. However, recent studies have shown that such methods often result in poorer signal-to-noise ratios in the modulation-spectrum domain, which may explain why little benefit in speech intelligibility has been found. Also, identifying noise in terms of its stationarity runs the risk of misidentification in more stationary signals, such as music. Here a noise reduction algorithm based on harmonic detection and enhancement was explored. Simulation results showed that this algorithm could help suppress noise in both speech and music. Results from perceptual tests will be reported. [Supported in part by Advanced Bionics.]

1aSCb7. Analysis of discrepancy between subjective and objective evaluation of noise-reduced speech. Mitsunori Mizumachi (Kyushu Institute of Technology, 1-1 Sensui-cho, Tobata-ku, Kitakyushu, Fukuoka 804-8550, Japan, mizumach@ecs.kyutech.ac.jp)

Discrepancy between subjective and objective evaluation is one of the annoying issues in speech processing including speech enhancement. Subjective evaluation is ideal, although it is time-consuming with a lot of participants. Then, objective distortion measures have been designed as the substitutes for subjective listening tests. However, each distortion measure is optimized under very restricted condition for the specific application. Therefore, discrepancy between subjective and objective evaluation of noise-reduced speech is often caused in the real world. In this paper, the factor of the discrepancy is investigated in detail by comparing the subjective evaluation with the short-term objective evaluation. Almost all state-of-the-art distortion measures introduce the importance weight in the frequency region. On the other hand, this paper considers the temporal variation of speech distortion to understand the relationship between subjective and objective evaluation of noise-reduced speech. Distribution of short-term speech distortion was prepared using the temporal frames with various lengths. It is found that the skewness of the short-term speech distortion distribution could be a clue for explaining the discrepancy between subjective and objective evaluation. [Work supported by NEDO, Japan]

1aSCb8. Robust voice activity detection based on harmonic to noise ratio. Yanmeng Guo and Qiang Fu (Key Laboratory of Speech Acoustics and Content Understanding, Institute of Acoustics, Chinese Academy of Sciences, No. 21, Beisihuan Xilu, Beijing, China, 100190, guoyanmeng@hcl.ioa.ac.cn)

A robust voice activity detection algorithm based on harmonic to noise ratio (HNR) is proposed. Harmonic to noise ratio is high in voice segments, because most of the voice energy distributes on the harmonic structure. However, it is unreliable or complicated to estimate the harmonic

frequencies of noisy speech, and the HNR in full frequency band is not robust for environments with non-stationary band-limited noise. In this paper, several harmonic templates with fundamental frequency changing in log-scale step are used to match the wide-band voice harmonic structure, and the fundamental frequencies are not need to be estimated. To avoid the non-stationary band-limited noise, the contaminated frequencies are neglected automatically by frequency bin selection, which discards the harmonic and the noisy bins with the highest and lowest energy to keep the main clear harmonic structure. The final voice activity detection is based on the HNR of continuous frames, and it shows robust performance on several databases.

1aSCb9. Objective and subjective intelligibility evaluations of noise-reduction algorithms in Mandarin. Junfeng Li, Dongwen Ying, Qiang Fu, Yonghong Yan (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Beisihuan Xilu, Haidian, Beijing, China, lijunfeng@hcl.ioa.ac.cn), and Masato Akagi (School of Information Science, Japan Advanced Institute of Science and Technology, 1-1, Asahidai, Nomi, Ishikawa, Japan)

In this paper, we summarize the recent work that we have done on the objective and subjective evaluations of single-channel noise-reduction algorithms in Mandarin Chinese. In the evaluations, clean Mandarin speech signals were first corrupted by three types of noises at two signal-to-noise ratios and then processed by five typical single-channel noise-reduction algorithms. The processed signals were presented to normal-hearing listeners for recognition in subjective evaluations, and passed to eight intelligibility prediction measures in objective evaluations. Subjective evaluation results showed that the majority of noise-reduction algorithms did not improve Mandarin speech intelligibility, and the objective evaluation results indicated that of all tested objective measures, the short-time objective intelligibility (STOI) measure provided the highest abilities in predicting Mandarin speech intelligibility in all conditions and in predicting the effect on speech intelligibility due to non-linear noise-reduction processing. These evaluation results reported here do provide valuable hints for analyzing and optimizing noise-reduction algorithms for Mandarin.

1aSCb10. Improved acoustic models for spontaneous speech recognition. Qingqing Zhang, Shang Cai, Jieli Pan, and Yonghong Yan (Key Laboratory of Speech Acoustics and Content Understanding, Chinese Academy of Sciences, zhangqingqing@hcl.ioa.ac.cn)

This paper describes advances for acoustic models in Chinese spontaneous Conversational Telephone Speech (CTS) recognition task. A number of approaches were investigated in the acoustic modeling, including Heteroscedastic Linear Discriminant Analysis (HLDA), Vocal Tract Length Normalization (VTLN), Gaussianization, Minimum Phone Error (MPE), Feature space MPE (fMPE), and etc. Considering pronunciation variations in continuous speech, tones in recognition vocabulary were modified due to the Sandhi rule. The acoustic models were trained on over 200 hours of audio data from standard LDC corpora. The improved acoustic models reduce the relative Character Error Rate (CER) by about 25% over the baseline acoustic models on standard LDC test set and China 863 program evaluation data set. Acknowledgment: This work is partially supported by the National Natural Science Foundation of China (No's. 10925419, 90920302, 10874203, 60875014, 61072124, 11074275, 11161140319).

1aSCb11. A study of feature selection in phonotactic language recognition. Chunyan Liang, Lin Yang, Junjie Wang, and Yonghong Yan (Key Laboratory of Speech Acoustics and Content Understanding, Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei Si Huan West Road, Haidian District, Beijing, China, liangchunyan@hcl.ioa.ac.cn)

This paper is a comparative study of feature selection methods in phonotactic language recognition. The phonotactic feature is presented by n-gram statistics derived from one or more phone recognizers in the form of high dimensional feature vectors. Feature selection is necessary for its ability of reducing the dimension of feature vectors so that the higher order n-gram features can be adopted in language recognition. This paper investigates four feature selection strategies that are introduced from text categorization, including mutual information (MI), Chi-squared test (CHI), information gain (IG) and weighted log likelihood ratio (WLLR). These methods are compared on the NIST 2009 Language Recognition Evaluation (LRE) task. The experimental results show that CHI, IG and WLLR can effectively obtain much lower dimensional features without affecting the language

recognition performance. In contrast, MI has relatively poor performance due to its bias towards favoring rare terms. This work is partially supported by the National Natural Science Foundation of China (No. 10925419, 90920302, 10874203, 60875014, 61072124, 11074275, 11161140319).

1aSCb12. Large margin gaussian mixture models for discriminative training in language recognition. Jinchao Yang and Yonghong Yan (Institute of Acoustics, Chinese Academy of Sciences, yangjinchao@hcccl.ioa.ac.cn)

In this paper, we try to integrate the concept of large margin gaussian mixture models (large margin GMMs) into discriminative training for language recognition. We proposed a new language recognition system (SVM-LM-ModelPushing system) which combines model pushing by large margin GMMs (LM-ModelPushing) with original model pushing by SVM (ModelPushing). Our experiments show that LM-ModelPushing includes the language dependent information. What's more, our experiments show that LM-ModelPushing contains different language dependent information comparing to ModelPushing. Experiment results on 2007 National Institute of Standards and Technology (NIST) language Recognition Evaluation (LRE) databases show SVM-LM-ModelPushing system gains relative improvement in EER of 9.1% and in minDCF of 8.8% comparing to original ModelPushing system in 30-second tasks.

1aSCb13. Non-negative matrix factorization of mixed speech signals based on improved particle swarm optimization. Hua Li (Institute of Acoustics, CAS 100190, leehwa@mail.ioa.ac.cn)

NMF (non-negative matrix factorization) is a recently addressed speech signal processing method. In this paper, we proposed a new NMF algorithm

based on improved PSO (particle swarm optimization) techniques at aims to extract non-negative components with low cross-talking error and high SNR. Compared with standard PSO algorithm, the improved PSO can overcome lower velocity of convergence by updating dynamic inertia weight. Our discussion is supported by experimental results for separating speech signals, which show that the proposed approach exhibits good performance than traditional NMF methods.

1aSCb14. High payload audio watermarking using multiple marking spaces. Md. Rifat Shahriar and Uipil Chong (University of Ulsan, 680 - 749, rsbdce@yahoo.com)

Audio watermarking is the process that imperceptibly embeds desired message into an audio file for the purposes like content authentication, content identification, data monitoring and tracking, and copyright protection. High embedding capacity is one of the desired requirements of every watermarking algorithm that always struggles against other important requirements like robustness and imperceptibility. In this paper we propose a time domain audio watermarking scheme that performs embedding of more than one digital message into the same cover work thus ensuring higher data payload as well as higher capacity. In this proposed approach, different watermark messages are inserted into different marking spaces which are obtained through orthogonal decomposition of the original audio signal. The proposed algorithm exploits perception characteristics of Human Auditory System (HAS) while providing robustness and higher embedding capacity. Our proposed scheme appears to be computationally efficient and simulation results confirm its robustness against strong attacks like noise addition, filtering, compression, re-sampling, re-quantizing, geometric distortion.

MONDAY MORNING, 14 MAY 2012

S421, 9:20 A.M. TO 12:40 P.M.

Session 1aUWa

Underwater Acoustics and Signal Processing in Acoustics: Advances in Underwater Acoustic Communication and Networking I

Daniel Rouseff, Cochair
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Wen Xu, Cochair
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James Preisig, Cochair
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Invited Papers

9:20

1aUWa1. An overview of acoustic telemetry: 2001 - 2011. Arthur Baggeroer (MIT, 77 Massachusetts Avenue, 5-206, Cambridge, MA 02139, *abb@boreas.mit.edu*)

In 2000 Kilfoyle and Baggeroer authored a review of acoustic telemetry. Since then there has been rapid progress on all aspects of acoustic telemetry. There have been many experimental investigations which have highlighted the doubly spread features of the acomsms channel. There have been many advances in single and multichannel equalizers for both channel inversion and channel matched filtering, also known as time reversal, for coherent communications. Nevertheless, the incoherent comms such as MFSK remains a reliable standby by the especially difficult channels. The presentation will highlight the many advances made in acomsms of the last decade.

10:20

1aUWa2. Signal processing in underwater acoustic communication system for manned deep submersible “Jiaolong”. Weiqing Zhu, Min Zhu, Yanbo Wu, Bo Yang, Lijun Xu, Xiang Fu, and Feng Pan (Lab of Ocean Acoustic Technology, Institute of Acoustic, No. 21, Bei-Si-huan-Xi Road, Beijing, China, 100190, zhu_min@263.net)

In this report, signal processing in underwater acoustic communication system for manned deep submersible “Jiaolong” is introduced. 1. Four communication methods are integrated to meet different needs: (1) coherent underwater acoustic communication, with a variable transmission rate from 5kbps to 15kbps, to transmit images. (2) Non-coherent underwater acoustic communication, with a transmission rate 300bps, to transmit texts, instructions, and sensor data. (3) Spread spectrum underwater acoustic communication, with a transmission rate 16bps, to transmit instructions. (4) Underwater voice communication, using analogy modulation method to transmit human voice. 2. Signal processing method in coherent communication mainly consists of concatenation of decision feedback equalizer and Turbo decoder, and wavelet based image compression with fixed length coding. In the equalizer, Doppler compensation, multichannel combining and equalizer coefficients updating are all using fast self-optimized adaptive algorithm. 3. A linear hydrophone array is lowered from the mother ship to certain depth, and spatial diversity combining technology is adopted. From July to August 2011, diving trial of “Jiaolong” was carried out in the Pacific Ocean. The communication distance can cover nearly all ocean depth. The covering conical area is wider than 100 degree. An optical/acoustic image could be transmitted in 7 or 14 seconds.

10:40–11:00 Break

11:00

1aUWa3. Differential OFDM for acoustic communications. Yashar Aval and Milica Stojanovic (Northeastern Univ., aval.y@ece.neu.edu)

High-rate acoustic communication typically rely on coherent detection which requires sophisticated channel estimation, and may in turn suffer a penalty in performance when channel tracking is less than ideal (a situation that is often inevitable on time-varying channels). To improve the robustness of signal detection, orthogonal frequency division multiplexing (OFDM) is considered with differentially coherent detection. The resulting receiver has very low computational requirements, and a potential to outperform coherent OFDM detection when channel tracking becomes difficult. Differential encoding is applied in the frequency domain (across carriers) so that it does not require the channel to remain constant over consecutive blocks in time. Instead, it requires only that the channel transfer function change slowly between adjacent carriers, but this requirement coincides with the basic premises of OFDM system design. At the same time, closely spaced carriers promote bandwidth-efficiency. For extreme situations, when close carrier separation leads to insufficient temporal coherence within each OFDM block, a method of partial FFT demodulation can be used with differentially coherent detection. The ensuing receiver algorithm is cast into the multi-channel (spatial diversity) framework, and its performance is illustrated using synthetic, as well as experimental data.

11:20

1aUWa4. Orthogonal frequency-division multiplexing underwater acoustic communications with time reversal processing. Xinyang Nie and Wen Xu (Department of Information Science and Electronic Engineering, Zhejiang University, Hangzhou, 310027, China, starsun87@126.com)

In dispersive underwater channels where impulse responses commonly last tens of milliseconds, large symbol durations and guard intervals are needed for orthogonal frequency-division multiplexing (OFDM) acoustic communications, which could introduce severe inter-carrier interferences and reduces effective data rate. This paper presents a scheme of OFDM transmission combined with passive time reversal processing, which has been demonstrated as a promising self-adaptive technique to compensate for multipath distortion explicitly due to its spatial focusing and temporal compressing characteristics. Using time reversal as a preprocessing step prior to OFDM, the equivalent channel impulse response is greatly shortened; moderate symbol durations and guard intervals can thus be used the same way as in conventional OFDM schemes. Moreover, to improve the robustness in harsh-environment applications, Reed Solomon channel coding is exploited for its good performance against burst errors caused by channel fading and ambient burst noise. Tradeoffs between data rate and robustness are discussed along with the transmission scheme. Finally some field experimental results are presented, which demonstrate the effectiveness of the developed approach. [Work supported by Chinese 863 high-tech program under Grant 2009AA093601]

11:40

1aUWa5. Multi-band OFDM for underwater acoustic communications. Robert Griffin (Colorado State University, griffin.rt@gmail.com), Fengzhong Qu (Zhejiang University), and Liuqing Yang (Colorado State University)

For underwater acoustic communications (UAC), the bandwidth is wide compared with the carrier frequency. Because of this fact, the advantages of using multiband OFDM (MB-OFDM) for ultra-wideband communications in terrestrial environments may also apply to UAC scenarios. In this paper, a comparison is made between the use of single-band OFDM and MB-OFDM for UAC. The complexity of each method is shown and experimental results from the WHOI09 undersea trial are presented for both single-band and multiband schemes. From the analysis and experimental results, the validity of treating UAC as ultra-wideband can be determined and the comparative advantages and disadvantages of MB-OFDM versus single-band OFDM for UAC discovered.

12:00

1aUWa6. Ranging, localization and tracking as functions of underwater acoustic networks. Joseph Rice (Naval Postgraduate School, Monterey, CA 93943, United States, jarice@nps.edu)

Through-water acoustic communications are now enabling distributed underwater networks with fixed and mobile nodes. This paper presents implementations of node-to-node ranging as a by-product of link-layer RTS/CTS handshaking and as an explicit product of ping/echo bidirectional communications. Simultaneous ranging to multiple nodes is accomplished by use of a broadcast ping. Experimental deployments of acoustic networks have demonstrated the use of ranging for purposes of neighbor discovery, network routing optimization, and node localization. These functions are combined to enable the autonomous initialization of large networks deployed in an arbitrary distribution. Acoustic ranging is also shown to enable underwater navigation by a mobile node operating in the domain of a fixed distributed network.

12:20

1aUWa7. A dual-channel cross-layer architecture for underwater acoustic networks. Xiaomei Xu, Zheguang Zou, and Yi Tao (College of Ocean and Earth, Xiamen University, China. 361005, xmxu@xmu.edu.cn)

The performance of underwater acoustic networks (UAN) is affected by the node device constraints of memory, processing power, battery life time and network topology variation. To improve the performance and to utilize scarce resource can be obtained with a Cross-layer design. Cross-layer design, one of the key technique in underwater communication networks, overcomes the disadvantages of the strictly layered networks such as nonoptimality and inflexibility. It enables the system to utilize the limited resources more sufficiently, especially in no central control, rapid changes topology of underwater networks, and achieves better performance. This paper discusses the benefits of cross-layer underwater acoustic networks and related work, introduces a representative cross-layer architecture, named dual-channel cross-layer architecture, to promote the overall system performance for underwater networks. In addition, three cross-layer solutions, node adaptive modulation and channel coding, joint design of MAC and nodes ranging, and MAC networks information extraction, are presented perfectly based on this architecture. Finally, by using NI compactRIO and LabVIEW, an experimental networking was carried out, which demonstrated promising results.

MONDAY MORNING, 14 MAY 2012

S426 + S427, 9:20 A.M. TO 12:40 P.M.

Session 1aUWb

Underwater Acoustics, Acoustical Oceanography, and Physical Acoustics: Sediment Acoustics of Continental Shelves I

Nicholas Chotiros, Cochair
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Invited Papers

9:20

1aUWb1. Wave propagation prediction over a thin elastic sediment and rock basement. Cathy Ann Clark (Naval Undersea Warfare Center, Newport RI 02841, cathy.clark@navy.mil)

A bottom model which includes compressional and shear wave transmissions and reflections through a sediment layer is utilized to derive a single bottom reflection loss coefficient. When used in conjunction with a normal mode model, the single coefficient is shown to successfully reproduce resonance effects due to shear wave conversion in various sediments. The consolidation of an infinite number of reflections and transmissions is accomplished by formulating an infinite sum of matrices and expressing the result as a convergent series. Comparisons to measured data are presented for a number of underwater environments.

9:40

1aUWb2. Matched-field processing using time-reversal concept in a range-dependent environment. Kunde Yang, Tongwei Zhang, and Yuanliang Ma (Institute of Acoustic Engineering, Northwestern Polytechnical University, ykdzym@nwpu.edu.cn)

Time-reversal processing (TRP) is an implementation of matched-field processing (MFP) where the ocean itself is used to construct the replica field. This paper introduces virtual time-reversal processing (VTRP) that is implemented electronically at a receiver array and simulates the kind of processing that would be done by an actual TRP during the reciprocal propagation stage. MFP is a forward

propagation process, while VTRP is a back-propagation process, which exploits the properties of reciprocity and superposition and is realized by weighting the replica surface with the complex conjugate of the data received on the corresponding element, followed by summation of the processed received data. The number of parabolic equation computational grids of VTRP is much smaller than that of MFP in a range-dependent waveguide. As a result, the localization surface of VTRP can be formed faster than its MFP counterpart in a range-dependent waveguide. As the number of parabolic equation computational grids for VTRP is much smaller than that for MFP, VTRP proceeds about 100 times faster than MFP. The performance of VTRP for source localization is validated through numerical simulations and data from the Mediterranean Sea.

10:00

1aUWb3. Inversion of sediment geoacoustic parameters with echo envelope characteristics. Guofu Li, Dazhi Gao, and Ning Wang (Ocean University of China 238 Songling Road, Qingdao, China, zhenglimuyun@sina.com)

Bottom backscattered signals of different sites located in the Bohai Sea of China were acquired using a calibrated vertically oriented echosounder working at 20kHz. Envelopes of the received signals are extracted. The backscattering intensity envelope is also computed based on a time-dependent model described by Daniel D.Sternlicht and Christian P.de Moustier, of all input parameters the mean grain size is used only. Other geoacoustic parameters related to the mean grain size are adapted from the APL-UW High-Frequency Ocean Environment Acoustics Models Handbook. Both the envelopes of experimental and modeled are used to calculate characteristics such as the duration of echoes, statistical and spectral moments and finally give out the estimation of mean grain sizes. The estimated parameters are consistent with the ground truth.

10:20

1aUWb4. Assessment of geoacoustic inversion methods. Ross Chapman (University of Victoria, 3800 Finnerty Road, Victoria, BC V8P5C2, chapman@uvic.ca)

Sound transmission in shallow water is strongly affected by the physical and acoustic properties of the ocean bottom. Over the past decade, sophisticated methods have been developed for estimating parameters of geoacoustic models that account for the interaction of sound with the bottom. The performance of the methods has been compared in benchmarking exercises for range-independent and range-dependent shallow water environments using simulated data. This paper extends the comparison of geoacoustic inversion methods to assess performance using data from experiments at sites where the ocean bottom environment was well known from independent ground truth information. There are several aspects to performance assessment. The comparison presented here shows the accuracy of estimates from various inversion methods compared to the ground truth data about the ocean bottom sediments. The methods that are compared include matched field inversion; perturbation techniques based on modal waveumber estimation; bottom loss measurements; travel time tomography; and in situ physical measurement. The assessment shows overall consistency from all the methods.

10:40–11:00 Break

11:00

1aUWb5. On the acoustics of gas-bearing marine sediment. Klaus C. Leurer and Colin Brown (National University of Ireland, Galway, Earth and Ocean Sciences, Galway, Ireland, klaus.leurer@nuigalway.ie)

Gas forms in marine sediments because of the decay of organisms in anoxic conditions abundant in sediments of inhibited water mobility. Its mechanical and thermodynamic properties, e.g., density and compressibility, which are significantly different from those of the pore water and the grain material will lead to a dramatic decrease in the sediment's sound velocity and effective density, as well as the quality factor, whenever even only a few percent of free gas is present in the sediment. A variety of possible spatial distributions of a gaseous phase has been identified, ranging from free spheroidal gas bubbles in the pore space-filling fluid over various "patchy-saturation" scenarios to the displacement of parts of the total saturated sediment matrix, the respective scheme depending on such factors as grain size, sorting, wettability, among others. These different spatial distribution schemes require individually appropriate conceptions for the calculation of the acoustic properties from sediment physical characteristics. A recently proposed acoustic model [JASA 123, pp. 1941-1951, 2008] has been developed to account for the two cases of free gas bubbles in the pore space and for the local displacement of the saturated sediment.

Contributed Papers

11:20

1aUWb6. Study on single-parameter inversion for shallow oceans. Ke Qu, Changqing Hu, and Mei Zhao (Shanghai Acoustic Laboratory, Institute of Acoustics China, Shanghai, Xuhui district, No. 465, Xiao mu qiao road, quake09@mails.gucas.ac.cn)

A new rapid geoacoustic inversion technique has been developed, by reducing the number of inversion parameters to one instead of multi-parameters inversion. After fitting basic seabed parameters, a new quantity defined as the bottom loss gradient was proposed and single-parameter inversion

method was designed accordingly. Seabed properties were inverted directly using single-hydrophone without complex measurement, intensive signal processing and optimization algorithm which once multi-parameters inversion needed. In this study, Experiments at sea proved single-parameter inversion to be effective. Good agreement is also shown between the results of this method and the matched field inversion carried out in the same experiment. The reflective date inverted by the technique also can predict propagation loss accurately. As a convenient way, the single-parameter inversion method proposed a new choice for real-time seafloor properties determination.

11:40

1aUWb7. Shear wave speed inversions using scholte wave dispersion.

Gopu R. Potty, James H Miller, Jennifer Giard (Department of Ocean Engineering, University of Rhode Island, Narragansett, RI 02882, potty@egr.uri.edu), Andrew R. McNeese, Preston S. Wilson (Mechanical Engineering Department and The Applied Research Laboratories, The University of Texas at Austin, 1 University Station C2200, Austin, TX 78712), and Yong-Min Jiang (NATO Undersea Research Centre, 19126 La Spezia, Italy)

Shear speeds in semi-consolidated and consolidated shallow water sediments can significantly impact compressional wave attenuation and arrival times of acoustic normal modes. In addition, shear properties of sediments are directly related to the strength of the sediments in geotechnical applications. All of these factors emphasize the importance of estimating shear speeds in shallow water sediments. One of the most promising approaches to estimate shear speed is to invert the shear speed profile using the dispersion of interface waves (Scholte waves). The propagation speed and attenuation of the Scholte wave are closely related to shear-wave speed and attenuation over a depth of 1-2 wavelengths into the seabed. Data from the tests conducted in Narragansett Bay and off Block Island in water depths ranging from 10 m to 25 m using the shear measurement system, developed at the University of Rhode Island, will be presented. Combustive Sound Source (CSS) was used to generate interface waves during these tests. An inversion algorithm to estimate the shear wave speed profile in the sediment will be presented. Estimates of the shear speed will be compared with ground truth data. [Work supported by Office of Naval Research]

12:00

1aUWb8. Passive vs active geoacoustic inversion with a compact receiver array (MREA/BP'07 sea trials). Jean-Pierre Hermand (Environmental Hydroacoustics lab, Université libre de Bruxelles (U.L.B.) av. F. D. Roosevelt 50, CP 194/05, B-1050 Brussels, Belgium, jhermand@ulb.ac.be), Olivier Carrière (Marine Physical Laboratory-0238 University of California, San Diego Scripps Institution of Oceanography 9500 Gilman Drive Spiess Hall, Room 457A La Jolla, CA 92093-0238), and Qunyan Ren (Environmental Hydroacoustics lab, Université libre de Bruxelles (U.L.B.) av. F. D. Roosevelt 50, CP 194/05, B-1050 Brussels, Belgium and National Key Laboratory of Underwater Acoustic Technology, Harbin Engineering University, Heilongjiang, 150001, China)

MREA/BP'07 sea trials were an interdisciplinary experimental effort that aimed at addressing novel concepts of Maritime Rapid Environmental

Assessment in shallow waters. Southeast of Elba island in Mediterranean sea, several standard and advanced techniques of environmental characterization covering the fields of underwater acoustics, physical oceanography and geophysics were combined within a coherent scheme of data acquisition, processing and assimilation. Broadband (0.2-1.6 kHz) active and passive sounds propagated over ranges on the order of 1 km have been used to extract information about the ocean and subbottom environments. This paper compares the results of different inversion methods: 1) global optimization and sequential Bayesian filtering applied to matched-field (MFP) and model-based matched filter (MBMF) processed multitone and frequency-modulated data, respectively, and 2) local feature analysis of striations extracted from interference data due to ship noise. The approaches only require a compact and sparse hydrophone array which is easily deployable from small vessels giving similar estimates of the bottom geoacoustic properties for assimilation into hybrid MREA schemes.

12:20

1aUWb9. Robustness of acoustic interferometry for sediment geoacoustic characterization. Qunyan Ren (Environmental Hydroacoustics lab, Université libre de Bruxelles (U.L.B.) av. F. D. Roosevelt 50, CP 194/05, B-1050 Brussels, Belgium and National Key Laboratory of Underwater Acoustic Technology, Harbin Engineering University, Heilongjiang, 150001, China, qunyanren@ulb.ac.be), and Jean-Pierre Hermand (Environmental Hydroacoustics lab, Université libre de Bruxelles (U.L.B.) av. F. D. Roosevelt 50, CP 194/05, B-1050 Brussels, Belgium)

Spectrogram of broadband sound field radiated by a moving ship usually exhibits striations in the frequency-range plane, whose structure is characterized by the multilayered waveguide properties. An acoustic interferometry technique was proposed for sediment geoacoustic characterization using local interference structure features. Such technique has been proven to be robust to source depth and range uncertainties by theoretical analysis and numerical simulation. In this paper, its robustness to receiver depth is discussed through real data processing, which is usually critical for inversion techniques based on matched field processing that exploiting the spatial and temporal structure of waterborne sound fields. Ship noise data were collected on the four elements of a drifting shallow receiver array in a soft sediment area, south of Elba Island in the Mediterranean Sea. For all the receivers, their inversion results together with uncertainties are in good agreement with that of active inversion method. The study demonstrates the reliability of the acoustic interferometry technique on even single-hydrophone receiver system for sediment geoacoustic characterization.

Session 1pAA**Architectural Acoustics and Noise: Acoustics in Concert Halls I**

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Chair's Introduction—1:55

Invited Papers

2:00

1pAA1. Concert hall acoustics. Leo L Beranek (BBN (now, ACENTECH) Cambridge, MA 02199, beranekleo@ieee.org)

Three of the world's famous concert halls, Boston's Symphony Hall, Vienna's Grosser Musikvereinsaal and Amsterdam's Concertgebouw, were built before 1901, are rectangular in shape, and are known to have good acoustics. Then came the overriding postulate of Architect Hans Sharoun, "Music in the Center", which resulted in the Philharmonie Hall in Berlin, built in 1963. This type of hall, now called Surround Shape because the audience sits on all sides of the stage, has since been adopted elsewhere. For very large audiences, a fan-shaped hall, the Koussevitzky Tanglewood Music Shed seating 5000, was erected in 1940, underwent major acoustical changes in 1959, and has become the model for summer venues. An entirely different design is the Town Hall in Christchurch, New Zealand (opened in 1972), sometimes called the "Lateral-directed reflection sequence (LDRS) type," which emphasizes lateral reflections that arrive early after the direct sound and that result in reduced energy in the reverberant sound. These types are discussed along with pertinent physical data. In addition to the effects of acoustics on the orchestral sound, factors in selecting a shape are the number of seats, the distance of the farthest seats from the stage, and who are likely to purchase tickets.

2:20

1pAA2. Acoustic design of Grand Theatre projects in China. Eckhard Kahle, Thomas Wulfrank, Yann Jurkiewicz (Kahle Acoustics, Avenue Moliere 188, B1050 Brussels, Belgium, ekahle@kahle.be), Brian Katz (LIMSI-CNRS, BP 133, F91403 Orsay, France), and Henrik Möller (Akukon Consulting Engineers Ltd, Hiomotie 19, FI-00380 Helsinki, Finland)

In the People's Republic of China a large number of so-called Grand Theatre projects have recently been completed or are at present being constructed. In addition, other Grand Theatres are still on the drawing boards. These new cultural venues, typically housing multiple auditoria, are dedicated to Chinese and western opera, symphonic music and theatre. The present paper discusses the acoustic design of the Wuxi, Weifang and Jinan Grand Theatres. Due to the stringent fast-track design process, it was considered unpractical to carry out conventional computer modelling studies to inform the design. Novel acoustic design techniques were used for fast optimisation of early reflections in close collaboration with the architects during the early design stages. In the later design stages, full acoustic verification was carried out, often while construction was already underway.

2:40

1pAA3. Acoustics of vineyard concert hall concerning the audience and the performers. Weihwa Chiang, Yirun Chen, Ite Yeh, Chiachun Chen, Yenkun Hsu (National Taiwan University of Science and Technology #43, Keelung Rd. Section 4, Taipei 106, Taiwan, edchiang1224@gmail.com), and Wei Lin (Hwa-Hsia Institute of Technology 111 Gong Hjuan Rd., Chung Ho, Taipei, Taiwan)

Designing a vineyard hall is generally considered as more challenging and time consuming than designing a shoebox hall. Acoustics design of a moderately large vineyard hall was investigated by computer simulation regarding both the audience and the performers. Alternative schemes with orthogonal and splayed terraces were developed with design strategies featuring seating arrangement, wall inclining, ceiling pitch, recess of remote side seating, wall splaying, riser slope, and railing arrangement. Low correlations among most acoustical measures indicated the potential for setting diverse design goals. Optimized schemes with a frontal terrace increased high-frequency components of a voice source by nearly 2 dB for the seats surrounding the stage while confined spatial decay of strength and sustained early decay were achieved. Lateral energy fraction was mainly determined by inclination and splaying of the terraces and side walls. Without significantly affecting energy distribution for the audience, layouts of the walls near the stage could cause 4 dB differences in early reflective strength measured between performers.

3:00

1pAA4. A Note on practical aspect on diffusive reflection in concert halls. Takayuki Hidaka (Takenaka R&D Inst. 1-5-1, Otsuka, Inzai, Chiba, 270-1395 Japan, hidaka.takayuki@takenaka.co.jp)

A number of reports have been recently published about the effect of irregularities, that is, diffusive reflection, on the interior wall of the concert hall. The majority of those reports, however, deal with measurements of the scattering coefficient. And there is only a small number of reports reviewing what size irregularity is actually preferable when a diffusive surface should be taken into account for the interior wall. In this paper, we focus on the actual diffusive surfaces in halls and discuss the acoustic behaviour of those surfaces by numerical analysis. We also address the acoustically recommendable property of the diffusive surface, which is considered to prevent acoustic glare, (Beranek, Concert and Opera Halls, 1996, chap. 10) and discuss its citing actual measurement data.

3:20

1pAA5. Improving orchestra pits for the benefit of musicians. Stephen Dance, Alba Losasa (LSBU, FESBE, Borough Road, London SE1 0AA, UK, dances@lsbu.ac.uk), Sarah Large, and Sheldon Walters (LSBU, FESBE, Borough Road, London SE1 0AA, UK)

Increased publicity regarding hearing loss in those working in the music and entertainment sectors and the need to meet compliance with the UK Control of Noise at Work Regulations 2005 has heightened the importance and need to reduce noise exposure of professional classical musicians. Advice on hearing protection for musicians often concludes that ear plugs are the most effective, although often problematic, noise mitigation measure and hence the need for alternative solutions. With the full co-operation of the Royal Academy of Music the noise exposure of musicians has been investigated to establish their typical noise dose. It was found that the orchestra pit was the most challenging environment, primarily limited by available space. To mitigate the noise dose of the musicians' two approaches were taken – changing the design of the pit and developing new zero footprint acoustic screens. The later involved developing a hybrid absorbing screen that could be placed on music stands. The former involved detailed room acoustic measurements and acoustical computer simulation of the theatre. The solutions provided a measured and predicted reduction in noise level of between 5-8 dBA for all the musicians without affecting the musical perception of either the conductor or audience.

3:40

1pAA6. Artec concert halls - a new generation. Damian J. Doria, Edward P. Arenius, Tateo Nakajima, and Todd L. Brooks (Artec Consultants Inc, New York, dd@artecconsultants.com)

In 2011, Artec opened new concert halls in Carmel, IN (USA), Reykjavik (Iceland), and Montreal (Canada). These three halls, each quite different from each other, indicate the directions that Artec's leadership will pursue and develop in the future. As on any major concert hall project, each tells a complex story about the Client's ambitions, expectations, resources and unforeseen challenges, as well as the interpersonal dynamics of the design teams. The differences of these halls development and final design will be discussed in regards to the acoustic adjustability, plan and sectional conformation, and the connection of the programming of each room relationship to its final form.

4:00–4:20 Break

4:20

1pAA7. Visual aspect in concert hall design - recent trend. Yasuhisa Toyota (Nagata Acoustics America, Inc. 2130 Sawtelle Blvd., Suite 308, Los Angeles, CA 90025, U.S.A., toyota@nagata.co.jp)

The visual aspect has become much more important in concert hall design. The New World Symphony opened a new concert hall in Miami in early 2011, expressing a multimedia visual concept where five large walls of the auditorium are used as screens for a multitude of projectors. A flexible layout for both musicians and the audience is enabled by a movable stage and audience areas. Musicians located away from Miami Beach collaborate with resident musicians through the use of the high-speed Internet2 network. Concerts in the Performance Hall are also served simultaneously to an audience outside the building in the adjacent city park with high-definition audio and video. Experimental collaboration between music and visual images is a key function of the building. The acoustics and acoustical design of this "visual" concert hall are discussed.

4:40

1pAA8. Sound levels in rehearsal and medium sized concert halls; are they too loud for the musicians? Anders Christian Gade (Partner Gade & Mortensen Akustik, A/S Hans Edvard Teglers Vej 5, 3rd. Floor DK 2920, Charlottenlund, Denmark, acg@gade-mortensen.dk)

After the EU directive related to sound exposure in the work environment became valid also for the music industry in February 2008, managers of symphony and opera orchestras in Europe should now pay serious attention to the sound levels to which their musicians are exposed. In this context, it is often discussed whether some halls are simply too small to accommodate the large sound power output of a symphony orchestra. Based on measurements of sound exposure levels of musicians according to ISO 9612 in both performance and smaller rehearsal halls, as well as room acoustic measurements in a number of small sized halls that we have designed, it is discussed whether this is likely to be true.

5:00

1pAA9. Acoustic enhancement in the Aylesbury theatre with the CARMEN[®] electroacoustic system. Christophe Rougier, Isabelle Schmich (Centre Scientifique et Technique du Bâtiment, 24 rue Joseph Fourier, 38400 Saint Martin d'Hères, France, christophe.rougier@cstb.fr), Helen Butcher (Arup Acoustics, Parkin House, 8 St Thomas Street, Winchester, SO23 9HE), and Delphine Devallez (48 avenue Victor Hugo, 92100 Boulogne Billancourt, France)

The 1200 seat Aylesbury Waterside Theatre opened in October 2010 in the UK. The theatre needed to be flexible enough to accommodate events and performances from pop to classical music, as well as opera and theatre. To host the different performances in the best acoustic conditions, it has been decided to design an acoustics adapted to amplified music ($RT = 1,1s$ at mid frequencies) and to install an acoustic enhancement system in order to adapt it for other music events. The CARMEN[®] electroacoustic enhancement system, designed by CSTB, has been chosen and installed. This paper presents the design and results of the installation of the Carmen system in the Aylesbury Waterside theatre. It details the CARMEN[®] electroacoustic design and explains the tuning and fine-tuning session with musicians. Detailed explanations are given for the use with orchestral music. Measurement results and the subjective evaluation with the feedback of acousticians and musicians are finally presented.

5:20

1pAA10. The calculation of impulse responses in concert halls below 80 Hz. Wolfgang Ahnert (AFMG Technologies GmbH, wahnert@afmg.eu)

Computer Simulation of room acoustics in large and medium-size venues has become a standard in the acoustic design process. But the Ray or Beam Tracing methods used in all such simulation programs cannot be applied at low frequencies. Here the rules of wave Acoustics must be considered. This work introduces a practical and accurate software-based approach for simulating the acoustic properties of concert and other large halls based on FEM. A detailed approach to obtain complex transfer functions is presented. By means of an inverse FFT impulse responses are obtained and compared with Ray Tracing results. It is shown that the results calculated with FEM extend the fine structure of Ray Tracing results at low frequencies. Also, it is understandable that the FEM simulation software can help to avoid modal phenomena and to place absorbers and diffusers in order to improve the acoustic quality of the hall.

5:40

1pAA11. Applications of large-scale finite element sound field analysis onto a music hall using ensemble averaged surface normal impedance. Toru Otsuru, Reiji Tomiku, Noriko Okamoto, Takeshi Okuzono, and Kusno Asniawaty (Oita University, 700 Dannoharu, Oita 870-1192, Japan, otsuru@oita-u.ac.jp)

To analyze the sound field in a practical room with complicated boundaries, the authors have developed large-scale finite element sound field analysis in both frequency and time domains. Although the surface normal impedance values of boundaries are required in the modeling process of the analysis, insufficient amount of the impedances are available to date. Then, to provide rather practical boundary conditions for numerical simulations on room acoustics, the authors have also proposed the concept and theoretical background of ensemble averaged surface normal impedance including the fundamental measurement technique. Herein, a brief summary of the finite element sound field analysis is given first. Next, the concept of the ensemble averaged surface normal impedance is explained. Then, several application analyses of a music hall's sound frequencies are conducted to show the resulting accuracy of the Large-scale finite element sound field analysis.

6:00

1pAA12. Objective analysis of concert hall design using ISO3382-1. Mike Barron (Fleming & Barron, Combe Royal Cottage, Bathwick Hill, Bath BA2 6EQ, m.barron@btinternet.com)

ISO3382-1 (originally issued in 1997) provides five basic objective measures for assessing concert hall acoustics: reverberation time, Early Decay Time, early-to-late sound index (C80), early lateral fraction and Strength (G) or total sound level. Optimum values for the objective measures have been proposed by several authors. But how much trust can one place in this approach? The author's book Auditorium acoustics and architectural design, 2nd edition, contains 16 case studies of concert halls with both subjective questionnaire ratings and objective measurements. This data can be used to assess the value and validity of these objective measures to offer an answer to the question: how reliable is design according to ISO3382-1, as currently used by many acousticians at the design stage? The analysis in fact uses the further analysis tool involving comparisons of measured levels with revised theory of sound level distribution.

6:20

1pAA13. A model to predict measurement uncertainties due to loudspeaker directivity and its validation. Ingo Witew, Tobias Knüttel, and Michael Vorländer (Institute of Technical Acoustics, RWTH Aachen University, Neustr. 50, D-52066 Aachen, Ingo. Witew@akustik.rwth-aachen.de)

In order to improve the understanding of uncertainties in measuring the acoustics in auditoria, the influence of a sound source's directivity is investigated. In previous work a model to predict the uncertainties when measuring room impulse responses with sources of a given directivity pattern has been developed. As a result, properties of the measurement environment, i.e. the size of the room, its reverberation as well as the sound scattering behaviour of the room surfaces, were identified to be significant secondary influences. Through extensive series of scale measurements data has been collected in a reverberation room to validate the model prediction. By introducing adjustable partition panels, absorbing and sound scattering surface elements the secondary influence factors were carefully controlled over a large range of values. After a brief explanation of the uncertainty model the results of the validation measurements will be presented. The significance of the different influence factors on the measurement uncertainty will be discussed.

1pAA14. Coupled volumes and statistical acoustics: preliminary results of an improved analytical model. Paul Luizard (Audio & Acoustics team, LIMSI-CNRS, 91403 Orsay, France & LAM team, Université Pierre et Marie Curie, Paris, France, paul.luizard@limsi.fr), Ning Xiang (Graduate Program in Architectural Acoustics, School of Architecture, Rensselaer Polytechnic Institute, Troy, New York 12180), and Brian FG Katz (Audio & Acoustics team, LIMSI-CNRS, 91403 Orsay, France)

Reverberation chambers, coupled to the main audience hall, make it possible to change the acoustics of a hall through the size and location of coupling surfaces. This passive device provides an acoustical variability which exceeds the possibilities offered by heavy curtains and moving reflectors. Since architects and acousticians are interested in predicting models as design tools, several approaches have been developed, using different simulation methods. This study proposes an improvement to analytical models of sound energy decay in coupled rooms, integrating temporal aspects proposed by Cremer and Müller as well as spatial components described by Barron's revised theory. Distances from the primary source located on stage as well as from secondary sources, which are the apertures between reverberation chambers and the main room, to the same receiver are included. Results from this analytical model are compared to those from ray-tracing software and scale model measurements, all based on the same simple shoebox geometry. While single volume rooms generally provide exponential sound energy decays, coupled volumes can present non-exponential decays under specific conditions. Hence adapted quantifiers are used to determine the characteristics of the obtained room impulse responses from the different methods.

MONDAY AFTERNOON, 14 MAY 2012

S224 + S225, 2:00 P.M. TO 7:00 P.M.

Session 1pBA

Biomedical Acoustics: Ultrasound Enhanced Drug Delivery

Constantin C. Coussios, Cochair
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Juan Tu, Cochair
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Invited Papers

2:00

1pBA1. Magnetic microbubbles for localised imaging and drug delivery: Development, characterisation and preliminary application in vivo. Eleanor Stride, Joshua Owen (Institute of Biomedical Engineering, Department of Engineering Science, Old Road Campus Research Building, University of Oxford, OX3 7DQ, UK, eleanor.stride@eng.ox.ac.uk), Helen Mulvana (Imaging Sciences Department, Imperial College London, London, UK), Quentin Pankhurst (The Royal Institution of Great Britain, London, UK), Mengxing Tang (BioEngineering Department, Imperial College London, London, UK), and Robert Eckersley (Imaging Sciences Department, Imperial College London, London, UK)

The use of coated microbubbles in therapeutic applications, in particular drug delivery and gene therapy, has become a highly active area of research. There remain, however, some significant challenges to be overcome in order to fully realise the potential of microbubbles in these applications. In particular, the difficulty in controlling the concentration of microbubbles at a given site and in ensuring sufficient proximity between bubbles and target cells, has frequently led to disappointing results from in vivo studies. Recent work has indicated that incorporating magnetic nanoparticles into the microbubble coating may provide an effective strategy for overcoming these challenges. Further investigation to fully understand the mechanisms of enhancement and hence optimise the delivery protocols is however required and this is the aim of the present study. Results will be presented from flow phantom studies demonstrating manipulation of bubble suspensions under physiological flow conditions, from high speed imaging used to investigate the dynamic behaviour of single microbubbles and theoretical modelling conducted to support and interpret the experimental findings. Finally results demonstrating in vivo transfection in a mouse model will be presented confirming successful localisation of the transfection site.

2:20

1pBA2. Ultrasonic microbubbles: vehicles for molecular imaging and localized drug delivery. Hairong Zheng (Shenzhen Institute of Advanced Technology, Chinese Academy of Sciences, 1068 Xueyuan Avenue, Shenzhen University Town, Shenzhen, P.R. China, hr.zheng@siat.ac.cn)

Recent biomedical ultrasound development taking advantage of unique properties of ultrasound contrast microbubbles on ultrasonic molecular imaging, drug delivery and therapy has opened up powerful emerging applications. The intense radial oscillation of microbubbles makes them several thousand times more reflective than normal body tissues and emits significantly stronger acoustic signal. Besides, the oscillation resonance that microbubbles produce has several special properties that can be exploited to improve diagnoses and even for drug delivery and therapy. Moreover, selective ultrasound excitation can spatially-manipulate the microbubbles through the

mechanism of acoustic radiation force and also destroy microbubbles to release the drug. In this talk, we present our current progresses on ultrasonic microbubbles as acoustic probes for molecular imaging, precise transportation micro-particles (drug), microbubbles and single cell to any specified location with MEMS technique by acoustic radiation force, as well as microbubble contrast agents as vehicles for ultrasound-mediated drug delivery in vitro and in vivo. The mechanism of the precise particle transportation using acoustic radiation force will also be discussed.

2:40

1pBA3. Cavitation-enhanced delivery of a self-amplifying oncolytic adenovirus for tumour-selective cancer therapy. Robert C. Carlisle (Dept of Oncology, University of Oxford, Old Road Campus Research Building, Oxford OX3 7DQ, UK, robert.carlisle@oncology.ox.ac.uk), James Choi (Institute of Biomedical Engineering, Dept. of Engineering Science, Old Road Campus Research Building, Oxford OX3 7DQ, UK), Leonard W. Seymour (Dept of Oncology, University of Oxford, Old Road Campus Research Building, Oxford OX3 7DQ, UK), and Constantin-C. Coussios (Institute of Biomedical Engineering, Dept. of Engineering Science, Old Road Campus Research Building, Oxford OX3 7DQ, UK)

Unlike conventional gene therapy, oncolytic adenoviruses selectively infect and replicate within cancer cells, potentially enabling systemically administered yet highly targeted self-amplifying cancer therapy. Until recently, therapeutic efficacy was hindered by limited extravasation of the virus to poorly vascularized tumour regions and by liver toxicity beyond a certain dose. In the present work, co-injection of the virus with contrast agent microbubbles (SonoVue) and exposure of the tumour to ultrasound using a set of optimized parameters (0.5 MHz, peak rarefactional pressure 1.2 MPa, pulse length 50,000 cycles, pulse repetition frequency 0.5 Hz) result in inertial cavitation, which is found to enable increased extravasation and improved distribution of the virus throughout the tumour. Stealthing of the virus using a novel polymer coating results in improved circulation times, yielding a 30-fold increase in tumour viral expression at 3 days relative to delivery without ultrasound. Post-injection survival of mice bearing subcutaneous human breast cancer cell tumours (ZR75.1) of initial volume in excess of 30 mm³ is extended from 22-42 days for the virus alone to 22-80 days in the presence of inertial cavitation (n=7). Ultrasound-enhanced delivery mediated by inertial cavitation is thus expected to play a key role in the clinical application of oncolytic virotherapy.

3:00

1pBA4. The impact of microbubbles on measurement of drug release from echogenic liposomes. Jonathan Kopechek (Biomedical Engineering, University of Cincinnati, Cincinnati, OH, 45267, kopechek@bu.edu), Kevin Haworth (University of Cincinnati, Cincinnati, OH, 45267), Kirrith Radhakrishnan (Biomedical Engineering, University of Cincinnati, Cincinnati, OH, 45267), Shaoling Huang (University of Texas Health Sciences, Houston, TX), Melvin Klegerman (University of Texas Health Sciences, Houston, TX), David McPherson (University of Texas Health Sciences, Houston, TX), and Christy Holland (Division of Cardiovascular Diseases, Internal Medicine, University of Cincinnati, Cincinnati, OH 45267)

Echogenic liposomes (ELIP) are under development to enable ultrasound-triggered drug delivery. The mechanisms of ultrasound-mediated drug release from ELIP are not well understood. The effect of cavitation activity on drug release from ELIP was investigated in flowing solutions using two fluorescent molecules: a lipophilic drug (rosiglitazone) and a hydrophilic drug substitute (calcein). ELIP samples were exposed to pulsed Doppler ultrasound from a clinical diagnostic ultrasound scanner at pressures above and below the inertial and stable cavitation thresholds. Control samples were exposed to Triton X-100, a detergent (positive control), or to flow alone (negative control). Fluorescence techniques were used to detect release. Encapsulated microbubbles reduced the measured fluorescence intensity. This effect should be considered when assessing drug release if microbubbles are present. Release of rosiglitazone or calcein compared to the negative control was only observed with detergent treatment, but not with ultrasound exposure, despite the presence of inertial or stable cavitation activity. Thus, cavitation activity did not correlate with release of rosiglitazone or calcein from ELIP using a clinical diagnostic ultrasound scanner. These findings lay the foundation for future studies of ultrasound-mediated drug delivery with ELIP.

3:20

1pBA5. Piezoelectric effect of cell's membrane. Qian Cheng and Meng-Lu Qian (Institute of Acoustics, Tongji University, Shanghai 200092, China, q.cheng@tongji.edu.cn)

In this paper, the piezoelectric effect of cell's membrane at nano-scale is preliminary investigated. For a eukaryotic cell, either it or every organelle in it is enclosed in a similar membrane made of the phospholipid bilayer. A lot of cell's physiological activities, such as signal transduction, ion transport and macromolecules delivery, realize through the membranes. Fundamentally, the realization of these physiological functions originates from the physical properties of the phospholipid bilayer. Here, the detection results of the dynamic piezoelectric effect of the plasma membrane and the nuclear envelope of rat A7r5 aorta smooth muscle cell at nano-scale using PFM are present. The results verify that cell membrane is piezoelectrically active due to ordered arrangement of polar phospholipid molecules in the liquid crystalline state. Consequently, this indicates that the ultrasound acting on the membrane structure will lead to the change of membrane potential, suggesting the piezoelectricity of cell membrane may play key roles in physiological activities of cells, further in drug/gene delivery, cancer treat, and so on. This work is supported by the National Natural Science Foundation of China (No. 10804085 and 11174223)

3:40

1pBA6. Feasibility study of using macrophages as drug delivery carriers for drug-loaded phase-change droplets. Chih-Kuang Yeh (Department of Biomedical Engineering and Environmental Sciences, National Tsing Hua University, 101, Section 2, Kuang-Fu Road, Hsinchu, Taiwan 30013, ckyeh@mx.nthu.edu.tw)

This study investigated the acoustic droplet vaporization (ADV) of perfluoropentane (PFP) droplets in single droplet-loaded macrophages (DLMs) by insonation with single three-cycle ultrasound pulses. Transient responses of intracellular ADV within a single DLM were observed with synchronous high-speed photography and cavitation detection. Ultrasound B-mode imaging was further applied to

demonstrate the contrast enhancement of ADV-generated bubbles from a group of DLMs. The PFP droplets incorporated in a DLM can be liberated from the cell body after being vaporized into gas bubbles. Inertial cavitation can be simultaneously induced at the same time that bubbles appear. The coalescence of bubbles occurring at the onset of vaporization may facilitate gas embolotherapy and ultrasound imaging. Macrophages can be potential carriers transporting PFP droplets to avascular and hypoxic regions in tumors for ultrasound-controlled drug release and ADV-based tumor therapies.

4:00–4:20 Break

Contributed Papers

4:20

1pBA7. Acoustic cavitation and sonoporation involved in ultrasound-assisted gene transfection with polyethylenimine in vitro. Juan Tu, Dong Zhang, Qian Li, and Tingbo Fan (Key Laboratory of Modern Acoustics (Nanjing University), Ministry of Education, Nanjing, Jiangsu, 210093, P.R. China, juantu@nju.edu.cn)

It has been shown that, acoustic cavitation-induced sonoporation should play an important role in the enhancement of gene/drug delivery. However, obstacles still remain to achieve controllable sonoporation outcomes. In the current work, MCF-7 cells mixed with PEI: DNA complex were exposed to 1-MHz ultrasound pulses with varied acoustic peak negative pressure, total treatment time, and pulse-repetition-frequency (PRF). The IC activities were detected using a passive cavitation detection (PCD) system and quantified as inertial cavitation dose (ICD). The DNA transfection efficiency was evaluated using flow cytometry and the cell viability was examined by PI dying assessment. Then, scan electron microscopy was used to investigate the sonoporation effects on the cell membrane. The results show that: (1) the ICD generated during US-exposure could be affected by US parameters; (2) the pooled data analyses demonstrated that DNA transfection efficiency initially increased linearly with the increasing ICD, then tended to saturate instead of trying to achieve a maximum value while the ICD kept going up; and (3) the measured ICD, sonoporation pore size, and cell viability exhibited high correlation among each other. All the results indicated that ICD could be used as an effective tool to monitor and control the US-mediated gene/drug delivery effect.

4:40

1pBA8. Characterization of an innovative drug carrying photoacoustic contrast agent: fluorescent polymer microcapsules. Guillaume Lajoinie, Erik Gelderblom (University of Twente/POF (Physics of Fluids), Enschede, Netherlands, g.p.r.lajoinie@utwente.nl), Ceciel Chlon, Marcel Bohmer (Philips Research, Eindhoven, Netherlands), Srirang Manohar (University of Twente/BMPI (Biomedical Photonic Imaging), Enschede, Netherlands), and Michel Versluis (University of Twente/POF (Physics of Fluids), Enschede, Netherlands)

Local drug delivery is studied to cross biological barriers and reduce the systemic side effects. Multimodal agents are being developed to combine step-response activation with the monitoring of its triggered release. The release can be triggered by acoustical or thermal means, e.g. using thermo-sensitive liposomes. Here we design an optically triggered microcarrier with well-controlled release precision and, in addition, a strong acoustic response in the far field, making the carrier a highly specific photoacoustic agent. The novel biocompatible microcapsules with a shell of fluorinated poly-L-lactic acid mixed with a fluorescent dye were produced with hexadecane oil core as drug-carrier reservoir. Single capsules were excited by a pulsed laser and their responses were monitored through combined ultra-high-speed imaging and sensitive acoustic detection. The experiments support a model where the polymer heats up through dye absorption thereby inducing the shell

destruction and the vaporization of the surrounding water, resulting in the core release. Beyond the classical pulsed laser photoacoustics, capsules also respond to CW laser excitation by emitting a continuous signal, which offers promising opportunities for real-time photoacoustics. The subsequent study shows that the prolonged response results from repeated vaporization cycles and a complex interaction of the laser with the dyed polymer.

5:00

1pBA9. Microseconds vaporization dynamics of superheated droplets upon triggering with focused ultrasound. Oleksandr Shpak (University of Twente, The Netherlands, o.shpak@utwente.nl), Tom Kokhuis (Erasmus MC, The Netherlands), Brian Fowlkes (University of Michigan), Nico de Jong (Erasmus MC, The Netherlands), and Michel Versluis (University of Twente, The Netherlands)

Liquid emulsion nanodroplets composed of perfluorocarbon (PFC) and a drug (Doxorubicin) are currently being studied as a potential highly efficient system for tumor imaging and for local drug delivery. The nanodroplets have the ability to extravasate through hyperpermeable tumor blood vessel walls, and to accumulate in interstitial tissue. The extravasated droplets can be triggered and vaporized with focused ultrasound, converting them into gas bubbles while the encapsulated drugs are released during the explosive evaporation of the droplet. Single and double emulsions of PFC-in-water and oil-in-PFC-in-water upscaled to 5-10 μm size were prepared and the nucleation and growth of the vapor bubbles ($f=3.5$ MHz, $P_{-}=4.5$ MPa) was imaged at frame rates of up to 20 Mfps with the Brandaris ultra high-speed imaging facility. The recorded images provide new and detailed insight in the physical mechanisms associated with the vaporization dynamics. This include droplet deformation and oscillatory motion along with surrounding fluid with an amplitude of 200-400 nm, rapid growth of a vapor nucleus with a speed of 40 m/s and consecutive oscillations and collapse of several bubbles.

5:20

1pBA10. Nonlinear viscous stress modification in the lipid-coated contrast agent microbubble dynamic model. Qian Li, Juan Tu, and Dong Zhang (Key Laboratory of Modern Acoustics (Nanjing University), Ministry of Education, Nanjing, Jiangsu, 210093, P.R. China, lilucky10@yahoo.cn)

In the existing shell models for lipid-encapsulated microbubbles, the viscous shell terms always have the linear form, which assumes that the viscous stresses acting inside the lipid shell are proportional to the shell shear rate with constant coefficient of proportionality. In the present work, a modified dynamic model is proposed for the lipid-coated ultrasound contrast agent (UCA) bubbles by taking into account the nonlinear viscous properties of a lipid monolayer coating. The dynamic responses of the UCA bubbles exposed to 1-MHz ultrasound pulses with varied driving pressures were measured using a modified flowcytometry system. By fitted the measured bubble dynamic curves with the proposed model, it has been verified that the use of the nonlinear theory for shell viscosity allows one to more accurately model the complicated UCA microbubble rheological properties.

5:40

1pBA11. Optical characterization of individual liposome-loaded microbubbles. Ying Luan, Telli Faez (Department of Biomedical Engineering, Thoraxcenter (Faculty Building), Ee 2302, Erasmus Medical Center, Rotterdam, the Netherlands, y.luan@erasmusmc.nl), Erik Gelderblom (Department of Physics of Fluids, University of Twente, Drienerlolaan 5, 7522 NB Enschede, the Netherlands), Ilya Skachkov (Department of Biomedical Engineering, Thoraxcenter (Faculty Building), Ee 2302, Erasmus Medical Center, Rotterdam, the Netherlands), Bart Geers, Ine Lentacker (Laboratory of General Biochemistry & Physical Pharmacy, Ghent University, Harelbekestraat 72, B-9000 Ghent, Belgium), Antonius van der Steen (Department of Biomedical Engineering, Thoraxcenter (Faculty Building), Ee 2302, Erasmus Medical Center, Rotterdam, the Netherlands), Michel Versluis (Department of Physics of Fluids, University of Twente, Drienerlolaan 5, 7522 NB Enschede, the Netherlands), and Nico de Jong (Department of Biomedical Engineering, Thoraxcenter (Faculty Building), Ee 2302, Erasmus Medical Center, Rotterdam, the Netherlands)

Newly developed liposome-loaded (LPS) microbubbles [1] were characterized by comparing their oscillating response with standard phospholipid-shelled (BARE) microbubbles using the ultra-high speed imaging camera (Brandaris 128). 73 LPS bubbles and 41 BARE bubbles of diameters ranging from 3 μm to 10 μm were insonified by narrow band pulses with a driving frequency ranging from 0.5 MHz to 4 MHz and an acoustic pressure from 5 kPa to 100 kPa. Shell elasticity of LPS bubbles (0.17 ± 0.1 N/m) was nearly the same as that of BARE bubbles (0.19 ± 0.1 N/m) for all investigated bubble sizes. Clear difference of shell viscosity was found for bubbles larger than 6 μm . Averaged viscosity of LPS bubbles (2.5×10^{-8} kg/s) was almost twice of that of BARE bubbles (1.4×10^{-8} kg/s). A second finding for LPS bubbles was the dominant “expansion-only” behavior (70% of LPS bubbles), while this was only 13% for BARE bubbles. Results from this study will facilitate future preclinical studies and clinical applications of LPS bubbles for ultrasound triggered drug delivery system. Reference 1. Geers, B., et al., *J Control Release*, 2011. 152(2): p. 249-56.

6:00

1pBA12. Stable inertial cavitation with a confocal ultrasonic device for drug release from nongaseous sonosensitive liposomes. Cyril Lafon, Jean-Louis Mestas, Jacqueline Ngo, Lucie Somaglino, Jean-Martial Mari, Sabrina Chesnais (INSERM, LabTau, Université de Lyon, 151 Cours Albert Thomas, 69003, Lyon, France, cyril.lafon@inserm.fr), Esben Nilssen (Epitarget, Forskningsveien 2A, 0373 Oslo, Norway), and Jean-Yves Chapelon (INSERM, LabTau, Université de Lyon, 151 Cours Albert Thomas, 69003, Lyon, France)

Encapsulating chemotherapeutic agents in liposomes improves targeting and efficacy of treatments against some tumors. The present work aims at evaluating if sonosensitive liposomes combined with cavitation for drug delivery enhance efficacy and reduce toxicity. Two focused beams were combined for stabilizing the cavitation cloud and an imaging probe used for guidance. Each 1MHz focused transducer had a 5cm diameter and focal length. Exposure conditions were $10.8\text{kW}/\text{cm}^2$ Isppa, 250Hz PRF and 1% duty cycle. Phosphatidylcholine-based nongaseous liposomes were loaded with Doxorubicin. To control for mechanical tissue damage, AT2 Dunning tumors on rats were first exposed to ultrasound only. Treatment induced temperature rose below 0.5°C . The tumor growth was not significantly slowed down by ultrasound, but histological examination of tumors evidenced large areas of necrosis which resorbed one week after ultrasound. The new liposomes were compared with conventional HSPC-based liposomes in terms of efficacy and toxicity on the same tumor model. Ultrasound led to equivalent efficacy when applied on HSPC-based liposomes,

while the new liposomes were efficient only with concomitant cavitation. We present a confocal ultrasound set-up able to provide sufficient inertial cavitation for drug release from a nongaseous liposome with reduced systemic toxicity. Eureka-labelled project (E!4056) funded by NRC, FFN and OSEO.

6:20

1pBA13. Modeling the interaction of tandem bubbles for biomedical applications. Todd A. Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, Mark F. Hamilton (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029, hayta@arlut.utexas.edu), Chao-Tsung Hsiao, Jin-Keun Choi, Sowmitra Singh, Georges L. Chahine (Dynaflo, Inc., 10621-J Iron Bridge Rd., Jessup, MD 20794), Georgy N. Sankin, Fang Yuan, David Piech, and Pei Zhong (Dept. of Mech. Eng. and Mat. Sci., Duke Univ., 101 Sci. Dr. Durham, NC 27708)

Recent experiments, motivated by ultrasound-mediated drug and gene delivery, have utilized laser-generated tandem microbubbles to produce directional and targeted membrane poration of individual cells in microfluidic systems [Sankin et al., *Phys. Rev. Lett.* **105**, 078101 (2010)]. Two models describing the dynamics of coupled bubbles between parallel plates have been applied to understand these observations. The first approach is based on the Boundary Element Method in both 2D and 3D coordinate systems for bubbles bounded by finite rigid plates. Deformation of the bubble surfaces is taken into account, capturing phenomena such as bubble jetting and fragmentation [Hsiao et al., *Ultrasound Med. Biol.* **36**, 2065-2079 (2010)]. The second approach is semi-analytic, accounts for fluid compressibility and elasticity of the plates, but is limited to spherical bubble pulsation [Hay et al., *J. Acoust. Soc. Am.* **129**, 2477(A) (2011)]. Observations of tandem bubble interaction with adjacent biological cells and their potential for controlling cell poration will be discussed. Comparisons between simulation results obtained from the two models, as well as comparisons between the models and experimental measurements, will be presented. [Work supported by NIH grant nos. DK070618 and EB011603 (UT), 2R44EB005139-02A1 (DFI), DK052985 and RR016802 (Duke).]

6:40

1pBA14. Combination of magnetic resonance-guided focused ultrasound and polymer-modified thermosensitive liposomes for cancer therapy. Terence Ta (Boston University, 44 Cummington St, Boston, MA, terencet@bu.edu), Eun-Joo Park, Nathan MacDannold (Brigham and Women's Hospital, 221 Longwood Ave, Boston, MA), and Tyrone Porter (Boston University, 110 Cummington St, Boston, MA)

In this study, the response of solid tumors implanted in rat hindlimbs to doxorubicin (DOX) released locally from a novel polymer-modified thermosensitive liposome (pTSL) was investigated. The pTSL was engineered to release encapsulated DOX at lower thermal doses than traditional thermosensitive liposomes. Rat mammary adenocarcinoma cells were implanted in the hindlimb of healthy rats and allowed to grow to at least 100mm^3 . DOX-loaded pTSL were injected intravenously and allowed to accumulate in the tumor interstitium over several hours. MR-guided 1.7-MHz focused ultrasound (MRgFUS) was used to heat the tumor volume and trigger the release of encapsulated DOX. Acoustic parameters (i.e. acoustic power, pulse duration, etc.) were identified to heat and maintain tumors at 40°C or 43°C for five minutes. Treatment with DOX-loaded pTSL and MRgFUS-mediated heating significantly reduced the rate of tumor growth. The response of tumors to DOX released from pTSL at 43°C was comparable to the response of tumors treated with unencapsulated DOX. The results of this study demonstrate that solid tumors can be treated successfully with DOX-loaded thermosensitive liposomes and MRgFUS with negligible toxic effects.

Session 1pEAa

Engineering Acoustics: Acoustic Well Logging and Borehole Acoustics I (Poster Session)

Xiuming Wang, Cochair
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Hailan Zhang, Cochair
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Contributed Papers

All posters will be on display from 2:00 p.m. to 3:00 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 2:30 p.m. and contributors of even-numbered papers will be at their posters from 2:30 p.m. to 3:00 p.m.

1pEAa1. A downhole forward-looking ultrasonic imaging system. Weijun Lin, Hongbin He, Lei Liu, and Hailan Zhang (Institute of Acoustics, Chinese Academy of Sciences, linwj@mail.ioa.ac.cn)

Inspection and evaluation of casing damages is important in oil fields. Compared to normal casing damage detecting equipments which can only measure the radius of casing around the equipment, the downhole forward-looking ultrasonic imaging system can detect casing damage in front of the equipment. A phased-array transducer was used in this imaging system, the FPGA and DSP were introduced to reduce the size of downhole equipment significantly, and the image of the casing is generated via a series of signal processing, which correctly reflects the real shape of the casing. In addition, we analyze the factors influencing ultrasonic image quality. It shows that the influences of the three kinds of inconsistency among channels on ultrasonic imaging are trivial. The possible ranges of these inconsistencies are given. Several experiments were completed on damaged casing models by using the downhole forward-looking ultrasonic imaging system. The results show that the methods introduced in this article are valid and this downhole forward-looking ultrasonic imaging system is a complement to existed casing damage detection equipments. (This work was supported by the National Natural Science Foundation of China, 10874202, 11134011) Key Words: Casing damage, Phased array, Ultrasonic

1pEAa2. Numerical analysis on the acoustic well logging transducers with various fluid load. Qiuying Chen, Jiansheng Cong, Xiuming Wang, and Hailan Zhang (Institute of Acoustics, Chinese Academy of Science, No. 21, 4th Northwestern Ring RD, Haidian District, Beijing 100190, P.R. China, chenqiuying@mail.ioa.ac.cn)

It is numerically analyzed how the acoustic attributions of monopole well logging transducer change when excited in different fluid load media, such as water, silicone oil and mud with different components. The logging transducer is a radially polarized piezoelectric cylindrical tube with both ends shielded, being used as either transmitter or receiver. For the radially resonant mode of the transducer, the resonant frequency, transmitting voltage response and receiving sensitivity of the transducer are affected by the medium density and velocity. When the medium density is constant, the resonant frequency and the transmitting voltage response increase with the increasing of medium velocity, while the receiving sensitivity decrease with the increasing of medium velocity. When the medium velocity is constant, the resonant frequency decrease with the increasing of medium density, while the transmitting voltage response and receiving sensitivity increase with the increasing of medium density. For the given acoustic impedance, medium with different density and velocity have different effect on the transducer, which implies that the acoustic impedance of fluid load can't independently affect the acoustic attributions of transducer. The analysis results above have certain reference significance for the site operations of acoustic logging.

1pEAa3. Performance analysis for acoustic well logging receivers. Jiansheng Cong, Qiuying Chen, Qian Wei, and Xiuming Wang (Institute of acoustics, Chinese Academy of Sciences, No. 21, 4th Northwestern Ring RD, Haidian District, congjs@mail.ioa.ac.cn)

The structure of acoustic logging receivers can change their performance directly. In this article, impedance characteristics and receiving sensitivities of three kinds of acoustic logging receivers were numerically analyzed with the finite element method. The modeling results showed that: The piezoelectric tube transducer in radial vibration mode had higher receiving sensitivity in the frequency range of 8-20 kHz with some ups and downs, while its sensitivity changed great below 5 kHz. The laminated circles transducer in bending vibration mode had higher sensitivity and better flatness below 4 kHz, but its sensitivity and flatness changed lower from 8 kHz to 20 kHz. The rectangular laminated piezoelectric transducer in the length stretching mode had better flatness below 20 kHz. In addition, acoustic well logging receivers in the future will have higher sensitivity, wider frequency bandwidth and smaller volume, which was pointed out in the article.

1pEAa4. A logging while drilling acoustic isolation technology by varying thickness of drill collars at a distance greater than wavelength. Yuanda Su, Xiaoming Tang, Baohai Tan (China University of Petroleum, Qingdao, Shandong 266555, syuanda@sina.com), and Yukun Qin (China Petroleum Logging (CPL) Co., Ltd, Xi'an, 710075, China)

A key technology for logging while drilling (LWD) acoustic measurements is the design of an acoustic isolator to suppress tool waves propagating along the drill collar, such that acoustic signals from earth formations can be effectively measured under LWD conditions. Up to now, the LWD acoustic isolation is achieved by periodically cutting grooves along the drill collar between acoustic transmitter and receivers. Such a technique, although it is effective, reduces the mechanical strength of the drill collar and adds cost to the manufacturing and maintenance of the LWD tool, hindering the application of the LWD acoustic technology. We have developed an LWD acoustic technology that does not use the groove-cutting design. We utilize the inherent frequency stopband for extensional wave propagation along a cylindrical pipe and effectively broaden the stopband by combining drill pipes of different cross-section areas whose lengths are greater than a wavelength but are shorter than the transmitter-to-receiver distance. After propagation through the combined drill collar system, the stopband in the collar extensional wave is significantly widened and the wave amplitude in the stopband is substantially reduced. Making LWD acoustic measurements in this widened stopband allows for recording acoustic signals from the surrounding formation.

Session 1pEAb

Engineering Acoustics: Biomedical Transducers

Ling Xiao, Cochair
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Contributed Papers

2:00

1pEAb1. Automatic drowsiness detection system using autoregressive coefficients and neural network. Hyungseob Han, Dajung Kim, and Uipil Chong (University of Ulsan, 680 - 749, overhs@naver.com)

One of the main reasons for serious road accidents is driving while drowsy. For this reason, drowsiness detection and warning system for drivers has recently become a very important issue. Monitoring physiological signals provides the possibility of detecting features of drowsiness and fatigue of drivers. One of the effective signals is to measure electroencephalogram (EEG) signals. The aim of this study is to extract drowsiness-related features from a set of EEG signals and to classify the features into three states: alertness, drowsiness, sleepiness. This paper proposes a neural-network-based drowsiness detection system using Autoregressive (AR) coefficients as feature vectors and Multi-Layer Perceptron (MLP) as a classifier. Specifically, the proposed method estimates AR coefficients using EIV (Errors-In Variables) providing an accurate estimation in a noisy process and linear predictive coding (LPC) analysis not considering noise. Samples of EEG data from each predefined state were used to train the MLP program by using the proposed feature extraction algorithms. The trained MLP program was tested on unclassified EEG data and subsequently reviewed according to manual classification.

2:20

1pEAb2. Combing spatial impulse response with angular spectrum to simulate pulse ultrasound field and the analysis of simulation errors. Wentao Wu (BeiJing Beisihuan Road, No. 21, wuwentao@mail.ioa.ac.cn), Yi Lv, and Dong Wang

In ultrasound imaging field, the simulation of ultrasound imaging is very important. The calculation of ultrasound field is supporting for the theory of ultrasound imaging and improving quality of ultrasound imaging. Recently, the simulation method of ultrasound imaging is mainly based on spatial impulse response. It is researched how to calculate the ultrasound imaging by combing pulse angular spectrum with spatial impulse response to simulate ultrasound field quickly, And the relationship of pulse angular spectrum and spatial impulse response is researched theoretically. The numerical error of simulation ultrasound field in this method is given. It is proved that the method combing spatial impulse response with angular spectrum is correct and effective. This method provides the base for designing ultrasound imaging system exactly and doing some research in nonlinear ultrasound imaging simulation.

2:40

1pEAb3. The application of compressive sensing in synthetic transmit aperture imaging. Yi Lv, Wentao Wu, and Ling Xiao (Institute of Acoustics, Chinese Academy of Sciences, lvyi8512@126.com)

Synthetic transmit aperture (STA) imaging requires huge data amount due to its demand for high image quality, thus this increases the need for high performance hardware and limits the flexibility of the post-processing stages. Compressive sensing (CS) theory shows that the signals can be reconstructed from an extremely smaller set of measurements than what is generally considered necessary by Nyquist/Shannon theorem. In this paper,

the CS theory is applied to STA imaging by sparse representation of the image in k-space. This method is tested with Field II simulation and the result shows that the quality of image is maintained with reduced data size by the application of CS theory with reduced sampling rates.

3:00

1pEAb4. Estimation of blood velocity vectors using two generalized beam signals. Ling Xiao, Xiaohui Meng, and Wentao Wu (Institute of Acoustics, CAS 100190, xling@mail.ioa.ac.cn)

Traditional Doppler methods measure only the velocity component along the ultrasound beam direction, and a flow transverse to the beam is not displayed. The lack of information on the beam-flow angle creates an ambiguity that can lead to large errors in velocity magnitude estimates. Different triangle techniques have been proposed, which basically perform multiple measurements of the Doppler frequency shift originating from the same region. In this work, a generalized model is introduced for triangle and non-triangle techniques, in which two ultrasound beams with known relative orientation are directed toward the same vessel. The velocity vector can then be obtained under the condition, when the phase variations of the two beams are linear independence as the functions of the scatterer's movement direction. A novel vector estimator is proposed under the framework of the model. It uses only real received signals and their Hilbert transformation and is simulated by Field II, showing suitable for implementation in steerable linear array transducers.

3:20

1pEAb5. Optimization design of the send-receive combined compound bar piezoelectric transducer. Xiaohui Meng, Junlin Wang, and Ling Xiao (Institute of Acoustics, CAS 100190, mengxh@mail.ioa.ac.cn)

Send-receive combined compound bar piezoelectric transducer has been widely used in underwater acoustics and ultrasonic fields. The optimization of the send-receive combined compound bar piezoelectric transducer is studied. The effect of the position and the dimensions of the piezoelectric ceramic elements on the resonance frequency, the anti-resonance frequency and transmitting and receiving response are analyzed. The conclusions are beneficial to the optimization of send-receive combined compound bar piezoelectric transducer.

3:40

1pEAb6. Novel concept of therapeutic array transducer element using coresonance between oscillations of hemispherical piezoceramic shell and water sphere. Kenji Otsu, Shin Yoshizawa, and Shin-ichiro Umemura (Tohoku University, 6-6-05 Aoba, Aramaki, Aoba-ku, Sendai 980-8579, Japan, otsu@ecei.tohoku.ac.jp)

For therapeutic array transducers, it is required to reduce the electrical impedance of their elements so that the transducer can produce high ultrasonic power at a relatively low drive voltage. For this purpose, a new concept of concave hemispherical piezoceramic transducer element using its breathing mode has been investigated. The vibrational behavior of such a transducer is theoretically, numerically, and experimentally analyzed. Both resonance frequencies of the breathing-mode oscillation of a piezoceramic

spherical shell and the volume oscillation of a water sphere are not only inversely proportional to their diameters, but also very close to each other at the same diameter. Numerical simulation of the transducer element showed high acoustical coupling achieved by the coresonance between the piezoceramic and the water sphere half enclosed by the shell. To confirm the effect by the coresonance, simulation replacing water by virtual materials, having the same acoustic impedance as water but different longitudinal velocities, was performed. The electrical impedance curves of the concave shell were very sensitive to the longitudinal velocities of the virtual materials, whereas those of the convex shell remained unchanged, which strongly support the hypothesis. Experimental results with a prototype transducer element will also be discussed.

4:00–4:20 Break

4:20

1pEAb7. Unique gel-coupled acoustic physiological transducer for health and performance monitoring. Michael Scanlon (US Army Research Laboratory (RDRL-SES-P), 2800 Powder Mill Road, Adelphi, MD 20783, michael.v.scanlon2.civ@mail.mil)

The U.S. Army Research Laboratory developed a unique gel-coupled acoustic physiological monitoring transducer that exploits acoustic impedance matching between the sensor and the skin. This optimizes the transmission of body sounds into the sensor pad, yet significantly rejects ambient airborne noises due to an impedance mismatch. Experiments have shown significant ambient noise reduction in a high-noise anechoic chamber test. The sensor's sensitivity and bandwidth produce excellent signatures for detection and spectral analysis of diverse physiological events such as heartbeats, breaths, wheezes, coughs, blood pressure, activity, and voice for communication. The health and performance of soldiers, firefighters, and other first responders in strenuous and hazardous environments can be continuously and remotely monitored with body-worn acoustic sensors. Comfortable acoustic sensors can be built into a helmet suspension or personal protective gear, or in a strap around the neck, chest, and wrist. Pulse wave velocity (PWV) transit-time between neck and wrist acoustic sensors can indicate systolic blood pressure on a beat-by-beat basis. Larger torso-sized arrays can be used to acoustically inspect the lungs and heart, or built into beds for sleep monitoring. Acoustics is an excellent input for sensor fusion, and combining acoustics with electrical-potential sensors such as electrocardiograms and electroencephalograms can produce interesting results.

4:40

1pEAb8. A three-dimensional orientation method for estimating a concave spherical array in phased array high intensity focused ultrasound. Junlin Wang (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, 100190, wangjunl@mail.ioa.ac.cn), Xiaodong Wang, Zhongde Yu, and Ling Xiao

The pseudo-inverse matrix method can be used to get complex excitation vector through the forward propagation operator from the surface of the array to the set of control points before the scheduled complex pressure at the control points is formed. The relative positions of the elements in the array should be estimated accurately since the elements are randomly placed with high-frequency and large-size. The high precision three-dimensional coordinates of elements are obtained by ways of comparing phase differences measured and calculated. The ideal distribution of focused sound field validates the precision of this way in which the measured phase differences are regarded as the initial phases of the elements.

5:00

1pEAb9. The accomplishment of the field of the phased array high intensity focused ultrasound using pseudo-inverse method. Zhongde Yu (Institute of Acoustics, No. 21, Beisihuan Road, Beijing, China, yuzhongde@sina.com)

In this paper, the two-focused field of phased array intensity focused ultrasound have been accomplished using the pseudo-inverse method. To

reducing grating-labs, the elements in array are randomly placed. The forward propagation operator from the surface of the array to the set of control points should be known when we use the pseudo-inverse method to get the excitation vector. Because of the element placing with high-frequency and large-size, the relative positions of element need to be estimated accurately. A method to get the high precision three-dimensional coordinates of the element are proposed and so the excitation vector have been get to accomplish the field of two-focus.

5:20

1pEAb10. Ultra low power low noise amplifier circuit design for electrically evoked compound action potential measurement of cochlear implant. Feng Hong, Ping Li, and Ling Xiao (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, 100190, China, hongf01@gmail.com)

Cochlea implant is composed of sound processor and implant that can provide a sense of sound to people who are deaf or deeply hearing-impaired. Electrically Evoked Compound Action Potential (ECAP) Measurement, which is an effective way of monitoring the status of the auditory nerve, plays a vital role in the usage of cochlear implant. Design of an ultra low power low noise amplifier circuit for ECAP measurement based on Commercial Off The Shelf (COTS) is described and some experimental results of ECAP from guinea pigs are also presented. The circuit is flexible and transplantable that can be widely used in other medical implant devices.

5:40

1pEAb11. Electrode configuration influences electrically evoked compound action potentials of guinea pig. Li Meng (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, 100190, China, msandy@163.com)

The intracochlear electrode array of cochlear implants is used to electrically stimulate the residual hearing auditory nerve of profound sensorineural hearing loss. The present products used to implant possess configuration with the different contact style and different separation between the contacts. This study investigated the effects of electrode configuration on the auditory nerve compound action potentials in response to electric stimulation. We also investigated the channel interaction of the different electrode configuration. Adult guinea pigs were used in acute experimental sessions. We implanted three kinds of electrode array either (1) a narrow spacing banded array consisting of a tapered silicone elastomer carrier with a linear series of banding contacts; or (2) two wider spacing arrays consisting of a tapered silicone elastomer carrier with oval-shaped contacts. The electrically evoked compound action potential (ECAP) was recorded from the intracochlear. ECAP latency functions indicated that the electrode array with narrow spacing and banded contacts generated shorter latency than the electrode array with wider spacing and oval-shaped contacts. We also observed that the electrode array with banded contacts had greater ECAP amplitude than the electrode with oval-shaped contacts.

6:00

1pEAb12. Design and implementation of the electronic system of phased array high intensity focused ultrasound. Xiaodong Wang (Institute of Acoustics, Chinese Academy of Sciences, wangxiaodong@mail.ioa.ac.cn)

Phased array technology is the development direction of the high intensity focused ultrasound. In this paper, we will discuss the design and implementation of the electronic system of phased array high intensity focused ultrasound. The structure of the system is distributed, composed of PC and some controlling units, which communicate with CAN bus. Each unit receives data from the PC and controls the phase and amplitude of acoustic emission signals. Specifically, we will discuss some key technology of the electronic system, such as how to control the phase and amplitude of the emission signal, use the time reversal and pseudo-inverse matrix to get the phase of the emission signals.

Session 1pED**Education in Acoustics: Teaching Acoustics on Both Sides of the Pacific II**

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Invited Papers

2:00

1pED1. University of Hartford undergraduate acoustical engineering programs and teaching philosophy. Michelle C. Vigeant (Acoustics Prog. and Lab., Dept. of Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., W. Hartford, CT, 06117, vigeant@hartford.edu), and Robert D. Celmer

The University of Hartford is a small private institution that consists of seven schools and colleges, and is located in West Hartford, Connecticut, USA. A more detailed overview of the acoustical engineering curriculum and laboratory facilities will be discussed. Graduating high school seniors who wish to pursue the study of acoustical engineering at the University of Hartford have two ABET-accredited program options: (1) the Bachelor of Science in Mechanical Engineering (BSME) with Acoustics Concentration and (2) the interdisciplinary Bachelor of Science in Engineering (BSE), Acoustical Engineering & Music, which requires acceptance into the University's music conservatory, The Hartt School. These acoustical engineering programs are within the Mechanical Engineering Department, which is part of the College of Engineering, Technology and Architecture. Both programs require a number of theoretical courses in acoustics and vibrations. These theoretical courses are balanced with hands-on real-world design projects and a number of these projects are done for non-profit organizations to expose our students to service learning. Students participate in these industry-sponsored full-semester design projects at both the sophomore (2nd year) and senior (4th year) levels. As a result, students leave the program with a solid foundation of both the theory and real-world applications of acoustical engineering.

2:20

1pED2. Underwater acoustics education in Harbin Engineering University. Desen Yang (Acoustic Science and Technology Laboratory, Harbin Engineering University, Harbin 150001 China, xiukun_li@yahoo.com.cn), Xiukun Li, and Yang Li (Acoustic Science and Technology Laboratory, Harbin Engineering University, Harbin 150001 China)

Underwater acoustic engineering is a discipline for graduate students' study which is currently famous at 6 schools across China, two of which may offer the undergraduate-level program. Underwater acoustic engineering specialty of Harbin Engineering University derives from the first sonar specialty in China built in 1953, which is the earliest institution engaging in underwater acoustics education in Chinese universities. There are three education program levels in this specialty (undergraduate-level, graduate-level and PhD-level), and students may study underwater acoustics within any of our three programs. In this presentation, descriptions of underwater acoustics education programs, curriculum systems, and teaching contents of acoustics courses will be introduced.

2:40

1pED3. Acoustical education in architectural engineering program at the University of Nebraska. Siu-Kit Lau and Lily M. Wang (University of Nebraska-Lincoln, 1110 S 67th St, Omaha NE 68182-0816, slau3@unl.edu)

An increasing number of schools are offering Architectural Engineering (AE) programs; currently there are 20 schools across the United States. However, only few of these AE programs include acoustics as a main option in their curricula. A comprehensive review of U.S. graduate programs in a report of "Technology for a Quieter America" by the United States National Academy of Engineering found that there is not sufficient training for students in acoustics in the U.S. This presentation will review the Nebraska Acoustics Group, housed within the AE program at the University of Nebraska which began in 1998. To cope with the needs, students could study acoustics within any of our five engineering degree programs (BSAE, MAE, MEng, MS, and PhD). There are currently two AE faculty out of 13 who focus in acoustics at Nebraska, and the program regularly offers at least six recurring acoustics courses. Descriptions of the acoustics courses, the research interests of the Nebraska Acoustics Group, and where our graduates are to date will be given. Specifically highlighted will be the theme of our acoustics group: to promote the advancement and science of architectural acoustics by closely tying our coursework and research to practice in the 'real-world'.

3:00

1pED4. Teaching acoustics in Communication University of China. Lingyun Xie and Zihou Meng (Communication University of China, Beijing, 10024, xiely@cuc.edu.cn)

There are three specialties with relation to acoustics in Communication University of China. Each specialty has different main courses. The key is how to make good interdisciplinary education between these specialties, so that the students can share different experience and knowledge from both Arts and Science. Practicing is also an important element for acoustics education. The university has good relationship with the industry. The students often have their practical courses and experiments in the corporations and factories. It is a good chance for them to turn their knowledge from books into actual products and experience. In the recent years, more and more students enjoy the benefit from this education strategy in the Communication University of China.

3:20

1pED5. Teaching acoustics in Mexico. Fernando J. Elizondo Garza (Acoustics Laboratory, Mechanical and Electrical Engineering School, Nuevo Leon State Autonomous University. P.O. Box 28 "F", Cd. Universitaria, C.P. 66450, San Nicolas de los Garza, N.L., Mexico., fjelizon@hotmail.com), and Sergio Beristain (Acoustics Laboratory, National Polytechnic Institute, Mexico City. President of the Mexican Institute of Acoustics. P.O. BOX 12-1022, Narvarte, 03001, Mexico, D.F., Mexico)

Mexico has been too slow to create a so called "critical mass" of acousticians, goal still not reached. There are a number of reasons for this condition, such as: a very low government budget to develop science and technology (approximately 0.4 % of the Mexican GNP in 2011); a low cultural level which promote "self formation"; the perception of acoustics/audio as a field somehow considered as a luxury, well beyond food, dress and housing; a little interdisciplinary activity among professionals; the consideration by the education institutions of acoustics as a complimentary knowledge within some technical professions, instead as a field in its own value; the difficulty derived by the low number of specialists for the consolidation of postgraduate programs; and the frequent economical and political crisis that make almost impossible the long term planning among people and companies. In this lecture the education programs and the periodic events on acoustics in Mexico will be described. Short term perspectives are a little discouraging, but thanks to the personal activities of some distinguished acousticians through professional societies and specialized congresses, it is possible to gather researchers, disseminate the Mexican acoustics research activities, and promote the formal education on acoustics in Mexico.

Contributed Paper

3:40

1pED6. Undergraduate acoustics education in Nanjing University. Dong Zhang and Xiaojun Qiu (Department of Acoustic Science and Engineering, School of Physics, Nanjing University, Nanjing 210093, China, dzhang@nju.edu.cn)

The acoustics in the Department of Acoustic Science and Engineering, Nanjing University, has been evaluated as a state key subject for cultivating acoustic talents from undergraduate students to post-doctoral fellows. Our department prepares undergraduates for entry-level positions in the acoustic

field and further education at the master's level. Our educational programs are organized around two overlapping areas: Acoustics and Acoustical signal processing. Core courses include fundamentals of acoustics, acoustic measurement, and acoustic transducer. A series of advanced undergraduate courses have also been developed to provide students with formal training in acoustics, including: electronic acoustics, ultrasonics, architectural acoustics, audio signal processing, active noise control, and etc. The goal of this program is to prepare students with both solid foundation in acoustic science and capabilities in acoustic engineering.

4:00–4:20 Break

Invited Paper

4:20

1pED7. Basic acoustics for graduates: an experience in Uruguay. Alice Elizabeth González Fernández (IMFIA-Facultad de Ingeniería - 11300, elizabet@fing.edu.uy)

The Acoustics area has a very secondary place in the degree studies of Engineering and Architecture in Uruguay. Just a quick overview on environmental and room acoustics is provided for taking in account this issue in the Degree Project. As noise issues are becoming increasingly important in the working market, many professionals are interested on a specific training in this area. Teaching basic concepts for graduates is sometimes not an easy challenge: no one wants to ask questions that would bring out any technical weakness. Some young professionals are invited each year to attend the course to enhance the profiting of the course of basic acoustics for graduates. As they have been working on environmental acoustics during their undergraduate studies, they have no difficulty to manage with room acoustics. When they are invited to join the course, they are asked to have an active participation in classes and to cooperate with their peers, especially in applied issues.

Contributed Paper

4:40

1pED8. Acoustic education and its exploration at NPU. Kean Chen, Yonghu Yang, Yong Liang, Xiang Zeng, and Kunde Yang (School of Marine Engineering, Northwestern Polytechnical University, Xi'an Shaanxi 710072, China, kachen@nwpu.edu.cn)

School of Marine Engineering (SME), Northwestern Polytechnical University (NPU) offers two undergraduate degree programs bearing on the field of acoustics, including information countermeasure and environmental engineering, of which the former is focused on underwater acoustic signal and information processing and the latter on environmental acoustics and

noise control. The SME also provides graduate degree programs involving Acoustics, Underwater Acoustic Engineering, Environmental Engineering and Environmental Science leading to a master's degree, among which the former two also possess competency of doctoral education. As a key subject, the education respect to acoustics has developed its own characteristics of teaching reform. The SME is actively involved in fostering the talent of engineering and internationalization among the students. On the one hand, the school is devoted to improve practical and experimental skills of the students. On the other hand, recent years, Acoustics program at the school is gradually promoting its international teaching program.

Invited Papers

5:00

1pED9. Graduate education: Meeting the needs of the next generation of professionals in architectural acoustics. Ning Xiang, Jonas Braasch, Todd L. Brooks, and David Sykes (Graduate Program in Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, New York, 12180, xiangn@rpi.edu)

In the fields of architectural-, physical- and psycho-acoustics the pace of change results from research, materials science and professional practice. Integrating the latest advances into pedagogy poses challenges for educators who are charged with training future experts and leaders, many of whom do not have technical backgrounds. To meet this need, the Graduate Program in Architectural Acoustics at the School of Architecture at Rensselaer Polytechnic Institute has re-shaped its pedagogy using "STEM" (science, technology, engineering and mathematics) methods enabling individuals from a broad range of fields to succeed in this rapidly changing field. RPI's curricula in architectural acoustics-leading to both M.S. and Ph.D. degrees -includes intensive, integrative hands-on experimental components that fuse theory and practice in a collaborative environment- a "STEM" method. The program has attracted graduate students from a variety of disciplines- including individuals with B.Arch., B.S., or B.A. degrees in Architecture, Music, Engineering, Audio/Recording Engineering, Physics, Mathematics, Computer Science, Acoustics, Electronic Media, Theater Technology and related fields. Following completion, most graduates pursue careers in acoustical consulting where an integrated understanding of complex, technical phenomenon is essential. RPI's curricula covers: Architectural Acoustics, Applied Psychoacoustics, Engineering Acoustics, Aural Architecture, and Sonics Research Laboratories.

5:20

1pED10. Theoretical acoustics course for postgraduates in Nanjing University. Jian-chun Cheng (Department of Acoustic Science and Engineering, School of Physics, Nanjing University, Nanjing 210093, China, jccheng@nju.edu.cn)

There is a one academic year theoretical acoustics course for postgraduate students in the Department of Acoustic Science and Engineering of Nanjing University. The course introduces the physical principles and mathematical methods for acoustics in fluids, and the main objective is to deepen understanding of acoustical principle for postgraduate students in the field of acoustics following their undergraduate fundamental acoustics course. The contents include acoustic waves in ideal fluids, acoustic radiation in infinite space, acoustic scattering and diffractions, propagation and radiation of acoustic waves in dust, acoustic fields in enclosed space, acoustic waves in dissipative fluids, acoustic waves in layered fluid, acoustic waves in moving fluids, propagation of acoustic waves with finite amplitude, and effects generated by acoustic waves with finite amplitude. More details can be found in the book title "Principle of Acoustics in Fluids" published by Science Press in China in 2012.

Contributed Papers

5:40

1pED11. Development of education for acoustics in Hong Kong. C.F. Ng (H.K. Polytechnic University, cecfng@polyu.edu.hk), C.L. Wong, LiXi Huang (Hong Kong Institute of Acoustics), and Y.N. Au Yeung (Hong Kong Institute of Vocational Education (Morrison Hill))

Acoustical issues have been important subjects in Hong Kong since its progressive development into a cosmopolitan city. Its densely populated characteristics and vibrant nature are challenges often encountered by professionals of various fields involved in the design and operation of a variety of infrastructures and facilities, from the early stage of landuse planning to daily maintenance of plants and equipment. Acoustical issues have also become livelihood issues as people are demanding better quality of living in terms of their acoustical environment. Hong Kong has been putting efforts to cope with the need to address acoustical issues by the most fundamental means, i.e. education for almost all walks of life, aiming to promote

knowledge as well as good practices so as to build a "sound" environment. This paper will describe the development of education for acoustics to meet the challenges in an extremely active city like Hong Kong. The partnering efforts among professional associations, academic institutions, and the industries in fostering professional knowledge and enhancing continual training will also be covered.

6:00

1pED12. Acoustic design education for general liberal arts students. Akira Nishimura (Tokyo University of Information Sciences, 2658501, akira@rsch.tuis.ac.jp)

Products and services concerning acoustic design are widely available, and the designers who create and maintain them are generally specialists in acoustic design. Therefore, acoustic design education is important, and much effort has been made by many colleges and schools. However,

education for not only designers, but also users of acoustically designed environments is important. For example, knowledge of time alignment of loudspeakers for 5.1 channel surround sound is important for producing a better sound field. Scientific and acoustic knowledge can enrich and improve quality of life by utilizing acoustical technologies, products, services, and environments. An acoustic design class for general students enrolled in liberal arts education has been introduced at the Tokyo University

of Information Sciences. The main topics of the class are physical and psychological aspects of sound, noise and noise control, introduction to building acoustics, hearing impairment and hearing aids, soundscapes, music therapy, how to use audio products, sound pictograms, and sound design for video. A questionnaire survey administered to students showed that the class was useful for learning new schemes, technologies, and products concerning acoustic design in their daily life.

MONDAY AFTERNOON, 14 MAY 2012

S227, 2:00 P.M. TO 5:20 P.M.

Session 1pHT

Hot Topics: 3-D Sound II

Yang Hann Kim, Cochair
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Contributed Papers

2:00

1pHT1. Beaming teching application: recording techniques for spatial xylophone sound rendering. Miloš Marković, Esben Madsen, Søren Krarup Olesen, Pablo Hoffmann, and Dorte Hammershøi (Section of Acoustics, Department of Electronic Systems, Aalborg University, Fredrik Bajers Vej 7 B5, 9000 Aalborg, Denmark, *mio@es.aau.dk*)

BEAMING is a telepresence research project aiming at providing a multimodal interaction between two or more participants located at distant locations. One of the BEAMING applications allows a distant teacher to give a xylophone playing lecture to the students. Therefore, rendering of the xylophone played at student's location is required at teacher's site. This paper presents a comparison of different recording techniques for a spatial xylophone sound rendering. Directivity pattern of the xylophone was measured and spatial properties of the sound field created by a xylophone as a distributed sound source were analyzed. Xylophone recordings were performed using different microphone configurations: one and two-channel recording setups are implemented. Recordings were carried out in standard listening room and in an anechoic chamber. Differences between anechoic and reverberant xylophone sound for binaural synthesis are examined. One-channel recording approach with binaural synthesis for spatial xylophone sound rendering is proposed. One channel recording is processed to define multiple source positions for xylophone width representation. Binaural synthesis was used for the reproduction. This leads to spatial improvements mainly in terms of the Apparent Source Width (ASW). Rendered examples are subjectively evaluated in listening tests by comparing them with binaural recording.

2:20

1pHT2. Evaluation of dynamic binaural reproduction system for live transmitted xylophone recording. Esben Madsen, Miloš Marković, Søren Krarup Olesen, Pablo Hoffmann, and Dorte Hammershøi (Section of Acoustics, Department of Electronic Systems, Aalborg University, Fredrik Bajers Vej 7, 9220 Aalborg Ø, Denmark, *em@es.aau.dk*)

For a special teaching application of the telepresence research project BEAMING, a scenario of a remote teacher (the Visitor) teaching a local student to play a xylophone through an embodiment is defined. In order to achieve this, a system is required to record, transmit and render the sound of the xylophone to the teacher in a dynamic scene. In an implementation of such a system, the xylophone is recorded using a mono recording technique. The signal is then processed to spread out the sound of the distributed sound source as multiple point sources in the virtual scene experienced by the

Visitor. Finally head tracking allows for a dynamic binaural rendering of the xylophone sound. The goal of this paper is to evaluate the realism of this virtual (auditory) representation of a real xylophone. A listening test is designed to compare characteristics of a real physical xylophone in front of the listener with a rendering using the described system. The evaluation is done with a basis in methods previously used for evaluating the subjective sensation of presence in virtual reality systems, mainly based on questionnaires.

2:40

1pHT3. Calibration aspects of binaural sound reproduction over insert earphones. Pablo Hoffmann, Milos Markovic, Søren Krarup Olesen, Esben Madsen, and Dorte Hammershøi (Aalborg University, Aalborg Ø - 9220, Denmark, *pfh@es.aau.dk*)

Earphones are nowadays widely adopted for the reproduction of audio material in mobile multimedia and communication platforms, e.g. smartphones. Reproduction of high-quality spatial sound on such platforms can dramatically improve their applicability, and since two channels are always available in earphone-based reproduction, binaural reproduction can be applied directly. This paper is concerned with the theoretical and practical aspects relevant to the correct reproduction of binaural signals over insert earphones. To this purpose, a theoretical model originally developed to explain the acoustic transmission to and within the open ear canal is revisited [Møller, *Appl. Acoust.*, 36, 171-218 (1992)]. The model is modified accordingly in order to investigate the aspects of the transmission within the blocked ear canal that are significant to the calibration required to preserve the natural spatial cues that exist during normal hearing conditions, i.e. during an open-ear-canal situation. To evaluate the validity of the theoretical considerations outlined in this paper, measurements are conducted using an IEC711 occluded-ear simulator with a number of different types of insert earphones.

3:00

1pHT4. Effects of in-phase and anti-phase head rotation of a remote avatar robot on median plane localization. Yōiti Suzuki, Yoshitaka Ikeda (Research Institute of Electrical Communication, Tohoku University, Katahira 2-1-1, Aoba-ku, Sendai 980-8577, Japan, *yoh@riec.tohoku.ac.jp*), Makoto Otani (Faculty of Engineering, Shinshu University, Nagano, Japan), and Yukio Iwaya (Research Institute of Electrical Communication, Tohoku University, Katahira 2-1-1, Aoba-ku, Sendai 980-8577, Japan)

Dynamic cues induced by a listener's movement markedly improve sound localization (e.g. Kawaura, 1991; Iwaya, 2003). Using a simplified version of an avatar robot called TeleHead (Toshima, 2004), this study

investigated effects of horizontal head rotation on median plane localization. The head of our robot can rotate horizontally, synchronously following a listener's head rotation. Sound signals at the robot's two ears in an anechoic room are captured and reproduced for a listener in a remote soundproof room. The robot rotation was controlled to have in-phase or anti-phase rotation with the listener's head rotation with a ratio between the rotation magnitudes of a listener and the robot of 0.05, 0.1, or 1.0. Results show that the anti-phase dynamic cue increases front-back confusion when the ratio is 1.0, but the localization was little affected when it was 0.05 or 0.1. In contrast, the in-phase dynamic cue suppresses front-back confusion significantly, irrespective of the rotation magnitude. Consequently, the sound localization accuracy can be improved considerably if the robot head's direction of rotation in a remote site and that of a listener are identical, even if the robot head rotation magnitude is as little as 5 % of the listener's. (Work supported by MEXT, Japan)

3:20

1pHT5. A perceptual analysis of off-center sound degradation in surround-sound reproduction based on geometrical properties. Nils Peters (International Computer Science Institute, 1947 Center Street, Berkeley, CA, nils@icsi.berkeley.edu), and Stephen McAdams (Schulich School of Music, McGill University, 555 Sherbrooke Street West, Montreal, Quebec, H3A 1E3, Canada)

Surround-sound reproduction is usually limited to a position where the listener maintains optimal perception of the reproduced soundfield. To improve the reproduction quality at off-center listening positions (OCPs), a better understanding of the nature of the perceived artifacts is necessary. Based on the geometrical relationships of a listener to the loudspeaker in a surround setup, an OCP can be characterized with three attributes: time-of-arrival differences, sound-pressure-level differences between the signal feeds, and the direction of the arriving wavefronts. Two listening experiments were conducted to elicit the perceptual effects of the off-center sound degradation of each of these three attributes in qualitative and quantitative terms. The five most often qualitatively described artifacts are related to the position of sound sources; their distance and depth; reverberation and envelopment; their spread and width; and sound coloration. The quantitative study found that off-center sound degradation is primarily caused by the level differences of the loudspeaker feeds. The time-of-arrival differences have a stronger perceptual effect on percussive sound material than on sustained sound material. In two out of three musical excerpts, off-center sound degradation was primarily correlated with artifacts related to the reproduction quality of reverb and envelopment.

3:40

1pHT6. Listening test for three-dimensional audio system based on multiple vertical panning. Toshiyuki Kimura and Hiroshi Ando (National Institute of Information and Communications Technology, 2-2-2, Hikaridai, Seika-cho, Soraku-gun, Kyoto, 619-0288 Japan, t-kimura@nict.go.jp)

In this paper, the novel three-dimensional (3D) audio system is proposed. The proposed system is based on Multiple Vertical Panning (MVP) method and matches to the glasses-free 3D display system in which the size of screen is very large. The vertical position of sound images is synthesized by the panning between two loudspeakers placed at the top and bottom of screen. The horizontal position of sound images is controlled by the position of two loudspeakers. By the proposed system, multiple listeners can simultaneously feel the sound images at the position of objects depicted by the 3D display system. In order to evaluate the auditory performance of the proposed system, the listening test was designed by using the loudspeaker array in which twenty-seven loudspeakers were aligned on the vertical line. Sound

images were synthesized by the panning between two loudspeakers placed at the top and bottom of the loudspeaker array. Twelve subjects listened to a sound and reported the position of synthesized sound images. As a result, it was indicated that subjects could feel the synthesized sound images at the position between two loudspeakers placed at the top and bottom of the loudspeaker array.

4:00–4:20 Break

4:20

1pHT7. Realization of sound space information acquisition system using a 252ch spherical microphone array. Shuichi Sakamoto, Jumpei Matsunaga (Research Institute of Electrical Communication and Graduate School of Information Sciences, Tohoku University, 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi, 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Satoshi Hongo (Sendai National College of Technology, 48 Nodayama, Medeshima-Shiote, Natori-shi, Miyagi 981-1239, Japan), Takuma Okamoto (Graduate School of Engineering, Tohoku University, 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi, 980-8577, Japan), Yukio Iwaya, and Yōiti Suzuki (Research Institute of Electrical Communication and Graduate School of Information Sciences, Tohoku University, 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi, 980-8577, Japan)

Sensing of high-definition 3D sound-space information is important to realize total 3D spatial sound technology. Nevertheless, conventional methods cannot sense comprehensive 3D sound-space information at a listening point properly and precisely so that the information can be reproduced simultaneously for many individual remote listeners facing in different directions. To cope with this problem, we proposed a sensing method of 3D sound-space information based on symmetrically and densely arranged microphones called SENZI (Symmetrical object with ENchased Zillion microphones) (Sakamoto *et al.*, 2008). In the system using SENZI, sensed signals from the respective microphones are simply weighted and summed to synthesize a listener's HRTF, reflecting the listener's facing direction. This method is expected to sense 3D sound-space information comprehensively in accordance with the head motion of listeners who are listening in remote places. Dynamic cues provided by the listener's motion are important to render sound localization correctly and stably (e.g., Kawaura *et al.*, 1991; Iwaya *et al.*, 2003). We developed a system using a 252-ch spherical microphone array and FPGAs. This presentation introduces a method of realizing this system as a real-time system using results of analysis related to the accuracy of the synthesized sound-space information of the system.

4:40

1pHT8. An assessment of the simulation and auralization quality of the Virtusound platform. D. Alarcão (CAPS, DEEC, Instituto Superior Técnico, TULisbon, Av. Rovisco Pais, 1 P-1049-001 Lisbon, Portugal, diogo.alarcao@ist.utl.pt), and J. L. Bento Coelho

The Virtusound platform was developed for real time room acoustics simulation and auralization through binaural technology. The calculation module uses an accelerated mirror image source method and a time-dependent radiosity method for the computation of the binaural room impulse responses. Informal tests have shown the system to be accurate and the rendered tri-dimensional sound to be quite realistic. However, a more comprehensive assessment was seen to be required in order to guarantee the quality of the platform. This paper presents therefore a first assessment on the capabilities of the Virtusound system in terms of the predicted objective parameters and in terms of the produced auralizations.

Session 1pNSa

Noise: Noise Source Localization II (Lecture/Poster Session)

David Woolworth, Cochair
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Jun Yang, Cochair
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S.K. Tang, Cochair
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Contributed Papers

2:00

1pNSa1. Acoustic sources joint localization and characterization using compressed sensing. Francois Ollivier, Antoine Peillot (UPMC - Institut d'Alembert, 2 place de la gare de ceinture 78210 Saint-Cyr-l'Ecole, France, francois.ollivier@upmc.fr), Gilles Chardon, and Laurent Daudet (UPMC - Institut Langevin - LOA 10, rue Vauquelin 75005 Paris, France)

In this work, a Compressed Sensing (CS) strategy is developed in order to jointly achieve two complementary tasks regarding sound sources: localization and identification. Here, the sources are assumed sparse in the spatial domain, and greedy techniques are used for their localization. The case of coherent sources located in a plane is studied both numerically and experimentally at different frequencies. Results show that, in this framework, CS source localization is reliable using a significantly smaller number of microphones than classical techniques (standard or high resolution beamforming techniques), while overcoming some of their pitfalls. We then use a similar technique for the identification of the source nature, i.e. its radiation pattern, and here the sparsity domain is extended to a basis of elementary radiating functions. We present simulation and experimental results using calibrated sources and measurements performed with a 3D array of 80 randomly distributed microphones. This study investigates the limitations of Compressed Sensing in terms of resolution and reliability of the identification, with respect to the number of sensors, the signal to noise ratio and the density of the reconstruction region.

2:20

1pNSa2. Ray based virtual time reversal method for the localization of sound sources in reverberant fields. Zeng Xiangyang and Song Qianqian (College of Marine Engineering, Northwestern Polytechnical University, Xi'an, 710072, China, zengxy@nwpu.edu.cn)

Localization of sound sources in reverberant fields is significant for the research areas such as noise sources recognition, meeting speaker tracking system, intelligent robot design. Microphone arrays are usually used, however, in most applications in rooms it is practical and economical to decrease the microphones. In this paper, using only one microphone a virtual time reversal algorithm based on the ray-tracing method has been developed according to the reciprocity theorem and the time reversal invariance of linear wave equation. The algorithm has been validated by the localization experiments in a real room and a virtual room. Then the performance of the algorithm under the conditions of various reverberation time, source-receiver distance and sound reflection times has been investigated according to a specified ratio.

2:40

1pNSa3. Diagnosis and characterization of low frequency noise source for a cable car system. Wei-Hui Wang (National Taiwan Ocean University, whwang@mail.ntou.edu.tw), and Chieh-Yuan Cheng (HanSound Technology Co., Ltd.)

Noise emission generated by a cable car system in operation condition normally becomes a problem widely disturbing the residents living in extremely quiet environment. The noise source identification and the sound field simulation are discussed and addressed in this article. To identify the noise sources from the tower post of a cable car system, the spectral level of the structure-borne vibration, the near-field sound and the far-field sound are measured and analysed. Unexpectedly, it is found that there exists some special narrow band peak frequencies in the range 50~80 Hz and its multiples of all the measured vibration and sound level spectra. The fundamental peak frequency is identified as the frequency of the periodic regular uneven wire rope surface passing through the sheaves of cable wheels. Which depends on the running speed of the cable, the number of cable strands, the pitch of strands. Besides, the fundamental peak frequency appearing in the sound level spectra away from the surface of the tower post also depends on the vibrational mode of the post whether pertaining to radiation mode or not near the fundamental exciting frequency band. This can be clarified and illustrated by the sound field simulation analysis. Keywords: low frequency noise, cable car, structure-borne sound radiation

3:00

1pNSa4. Identification and location of the distribution of elementary sources based on phase conjugation method. Ting Li and Sheng Li (State Key Laboratory of Structural Analysis for Industrial Equipment, School of Naval Architecture, Faculty of Vehicle Engineering and Mechanics, Dalian University of Technology, Dalian 116024, P.R. China, litingyouxiang@sina.com)

Phase conjugation method can achieve the back propagation and adaptive focusing. It can be used for acoustic source localization. Localization of the distribution of elementary sources is discussed with the discrete phase conjugation planar arrays and the discrete phase conjugation sphere array numerically. Here, it is discussed how both of the shape of array and the distance between phase conjugation array and initial source can influence the spatial resolution. Three variants of phase conjugation arrays are studied: Phase conjugation array made of monopoles, dipoles, or both of them. Corresponding to the three variants, analysis is performed in terms of evanescent and propagative waves and an acoustic sink of the three variants, which absorbs the outgoing wave of the time-reverse wave, is also discussed. The

interference pattern of the wave generated by the initial source and the time-reverse wave is used to identify the combination acoustic source.

3:20

1pNSa5. Study on suppression of background noise using near-field acoustic holography with single layer microphone array. Huancai Lu and Yulai Song (Key Laboratory of E&M, Ministry of Education & Zhejiang Province, Zhejiang University of Technology, Hangzhou, China 310014, huancailu@zjut.edu.cn)

A study was carried out to suppress background noise in non-free field generated by target sound source and noise source based on near-field acoustic holography with single layer microphone array. The acoustic pressures in non-free field are expressed as superposition of incoming and outgoing spherical wave functions. The coefficients of those spherical wave functions are determined based on the principle of Helmholtz Equation Least-Squares (HELs) method. The sound field was then separated once both the coefficients of incoming and outgoing spherical wave functions obtained. The error incurred in the process of inverse calculation was minimized via least-squares method as utilized in HELs. Numerical simulations were conducted to validate the approach, in which the non-free field was generated by different sound sources with analytical solutions, such as dilating sphere, oscillating sphere, and vibrating simply-supported thin plate. Those sound sources perform as target source and background noise source alternatively. Experiments in non-free field generated by omnidirectional speaker and JBL speaker was also conducted to examine the validity and accuracy of the approach. The results from both simulations and experiments show that the approach is capable to reconstruct the target sources and suppress the background noise with satisfactory accuracy in low frequency range.

3:40

1pNSa6. Sound source localization based on laser measurement of air vibration. TianHao Cui, XiaoBin Cheng, and HongLing Sun (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, cth@mail.ioa.ac.cn)

This paper proposes a novel approach to locate sound source using laser measurement by testing multi-point air vibrations simultaneously. In this approach, a laser beam is generated in a free space and backscattered by various scattering points. All the backscattering is assumed to take place simultaneously since light travels much faster than sound. The backscattering time intervals can be measured and the locations of scattering points in the space can be calculated. This method could be regarded as a substitute of a receiving array with n - sound transducers, based on which, an algorithm is presented to locate the sound source. Experimental results show that the proposed method exhibits a high locating precision.

4:00–4:20 Break

4:20

1pNSa7. Detection of noise sources in monitoring systems. J. Wierzbicki and W. Batko (AGH University of Science and Technology, Department of Mechanics and Vibroacoustics, Cracow Poland, wierzbic@agh.edu.pl)

One of the most important issue during continuous long-term environmental measurements in open space is connected with automatic

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

1pNSa10. Near-field acoustic holography of cyclostationary sound fields. Zhimin Chen (Institute of Noise & Vibration, Naval University of Engineering, Wuhan 430033, P.R. China, czm12345678@yeah.net), Hongchun Chen (91663 Troops, Qingdao 266012, P.R. China), and Haichao Zhu (Institute of Noise & Vibration, Naval University of Engineering, Wuhan 430033, P.R. China)

The radiated sound field of rotating machinery or reciprocating machinery has a significant periodically time-variant nature. This is a kind of non-stationary sound field and called cyclostationary sound field. In the conventional planar near-field acoustic holography(PNAH), this kind of sound field

identification of noise sources. It is particularly important in monitoring systems which collect data for creation and verification of noise maps where noise from road, rail and air traffic and from industry is taken into consideration. The determination of sound wave direction in observation point as a data pre-selection can be a first step in automation of noise sources identification process. In many cases due to permanent localization of roads and industrial plants such pre-selection should be sufficient. The idea of spatial sound monitoring system based on 3D microphone with procedures of data acquisition, processing and visualization and first results of noise sources detection are presented.

4:40

1pNSa8. Numerical study on reconstruction of acoustic pressure field based on near-field acoustic holography with spherical array. Huancai Lu and Minzong Li (Key Laboratory of E&M, Ministry of Education & Zhejiang Province, Zhejiang University of Technology, Hangzhou, China 310014, huancailu@zjut.edu.cn)

This paper presents the results of numerical study on reconstruction of acoustic pressure field based on near-field acoustic holography (NAH) with spherical array, while standoff distance, diameter of spherical array, number of microphones vary respectively. Two acoustic pressure fields are analytically generated by two monopole sound sources on the opposite sides of spherical array, and on one side of spherical array but apart at small distance. The accuracy of localization and identification of sound sources at different frequencies with different setup of reconstruction parameters was examined by comparison of the reconstructed results to the analytical results. The simulation of reconstruction of acoustic pressure field based on NAH with spherical array may provide guideline for application of NAH with spherical array in engineering.

5:00

1pNSa9. Near-field acoustic holography of acoustic radiation from structures. Rongfu Mao, Zhimin Chen, and Haichao Zhu (Institute of Noise and Vibration, Naval University of Engineering, Wuhan430033, P.R. China, maorfu@163.com)

There are some difficulties for conventional Near-field Acoustic Holography (NAH) to analyze acoustic radiation from a large scale structure. To solve the problem, a method for NAH of large scale structures was presented. In the method, the normal velocities or sound pressure at a few points on the surface of the structure are measured by transducers, and that at other position on the surface of the structure are calculated by means of the radiation mode theory, then the radiated acoustic field may be analyzed by NAH. Since complex coupling terms no longer appear in the radiation modes, and only a few orders of modes are required to describe the acoustic field at low-to-mid frequencies, the accuracy of NAH analysis may be ensured. Moreover, according to the nesting property of radiation modes, the radiation modes at other frequencies can be replaced by that at maximum frequency, consequently the calculating procedure may be simplified and the calculating speed quicken. Finally, NAH analysis of acoustic radiation from large scale structures was illustrated using a $1\text{m} \times 1\text{m}$ simply supported steel plate. The results show that the radiated acoustic field can be reconstructed accurately under the circumstances of a few measurement points.

is treated as stationary field, so the information relating to the change of frequency with time will be loss inevitably. In this article, the cyclic spectral density(CSD) instead of the complex sound pressure was adopted as reconstructing physical quantity in the PNAH, and the cyclostationary PNAH(CP-NAH) technique was proposed. Meanwhile, focusing on the calculation complex of CSD and the accuracy of the cyclic nature extracted, the gathering slice method of CSD was proposed by referring time aliasing methods on time series. The experiment results illustrate that the cyclic nature of cyclostationary sound field may be extracted directly and the location of the source determined exactly as well.

1pNSa11. To observe and understand the basaltic eruptions with infrasound. Aurélien Dupont (Pusan National University, dupont.aurelien@free.fr)

The frequent eruptions of Piton de la Fournaise volcano (Reunion island) release an important quantity of magmatic gas into the atmosphere and generates infrasonic airwaves. The series of volcanic noise recorded, in the near field, on a microbarometer between 1992 and 2008 bring new constraints on the functioning of the eruptions. The detection and the modelling of the waveforms associated to the overpressurized explosions of gas bubbles leads to conceive the volcanic eruption as a puzzle game. The elementary pieces of the game, the eruptive regimes, are characterized and interpreted in the framework of a two-phase flow. The eruptive gas flow is also quantified. The main flow regime is the Strombolian activity where the infrasound signature come from the slug flow bursting. The tracking of the main source of noise, during the eruptions, shows that the size of the gas pockets which are maximum in the starting stage of the eruption, what corresponds to the Lava Fountain regime, constantly decrease until to disappear with the eruption end: the gas volume fraction constantly decrease in the volcanic conduit during an basaltic eruption. The quantitative analysis of the noise produced by the gas flow allows not only to understand a natural system as complex as a volcano but allows also to better monitor it.

1pNSa12. Surveying the infrasonic noise on a basaltic volcano to understand the eruptive dynamics. Aurélien Dupont (Pusan National University, dupont@pusan.ac.kr)

The frequent eruptions of Piton de la Fournaise volcano (Reunion island) release an important quantity of magmatic gas into the atmosphere and generates infrasonic airwaves. The series of volcanic noise recorded, in the near field, on a microbarometer between 1992 and 2008 bring new constraints on the functioning of the eruptions. The detection and the modelling of the waveforms associated to the overpressurized explosions of gas bubbles leads to conceive the volcanic eruption as a puzzle game. The elementary pieces, the eruptive regimes, are characterized and interpreted in the framework of a two-phase flow. The eruptive gas flow is also quantified. The main flow regime is the Strombolian activity where the infrasound signature come from the slug flow bursting. The tracking of the main source of noise, during the eruptions, shows that the size of the gas pockets which are maximum in the starting stage of the eruption, what corresponds to the Lava Fountain regime, constantly decrease until to disappear with the eruption end: the gas volume fraction constantly decrease in the volcanic conduit during an basaltic eruption. The quantitative analysis of the noise produced by the gas flow allows not only to understand a natural system as complex as a volcano but allows also to better monitor it.

MONDAY AFTERNOON, 14 MAY 2012

HALL C, 2:00 P.M. TO 3:40 P.M.

Session 1pNSb

Noise and ASA Committee on Standards: Annoyance and Health Effects II (Lecture/Poster Session)

Klaus Genuit, Cochair
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K.C. Lam, Cochair
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A. Lex Brown, Cochair
lex.brown@griffith.edu.au

Contributed Papers

2:00

1pNSb1. Measuring the exposure to sound samples in subjective experiments. Liang Yan, Kean Chen (College of Marine Engineering, Northwestern Polytechnical University, Xi'an 710072, China, liyan1832@hotmail.com), Florian Gomez, and Ruedi Stoop (Institute of Neuroinformatics, University and ETH Zurich, Winterthurerstrasse 190, 8057 Zurich, Switzerland)

Traditional measures of environmental noise exposure concentrate on time and power (e.g. Ldn). For short measurements, time is, however, of secondary importance and the approach may come up with misleading results. In this paper, we propose a novel method based on short-term dose values evaluated along the playing time of the sound samples, to solve this problem. A comprehensive study on potentially influencing factors is carried out, discussing the partitioning method for short-term period analysis, the statistical treatment of the short-term dose values and four different frequency weightings. Eleven indices are then used to measure the exposure of

the fixed duration sound sample. This lays the groundwork for the dose-annoyance relationship via subjective experiments.

2:20

1pNSb2. Examining a-weighted and c-weighted sound levels and noise code limits in respect to annoyance due to music sources. David Woolworth (Oxford Acoustics, Inc. 356 CR 102 Oxford, MS 38655, dave@oxfordacoustics.com)

C-weighting has recently been incorporated into noise ordinances to address music sources that incorporate high levels of low frequency pulsing; previously C-weighting was reserved for industrial and transportation noise. It is common for complainants to have valid concerns in regard to audible low frequency noise that does not qualify as a violation based on A-weighted measures and codes. This paper will survey a number of existing ordinances that utilize C-weighting and sampling speed to address these music sources and produce examples of urban propagation and transmission.

2:40

1pNSb3. Changes in oto-acoustic emissions after exposure to live music. Rodrigo Ordoñez, Dorte Hammershøi (Acoustics, Department of Electronic Systems, Aalborg University; Fredrik Bajers Vej 7-B5, DK-9220 Aalborg Ø, Denmark, rop@es.aau.dk), and Jan Voetmann (Voetmann Akustik; Forhåbningssholms Allé 2, 5th, DK-1904 Frederiksberg C, Denmark)

Distortion Product Oto-acoustic Emissions (DPOAE) and Transient Evoked Oto-acoustic Emissions (TEOAE) were measured in subjects before and after attendance to live music. The changes measured were compared to the exposure levels measured at the position of the subject. The main objectives of this experiment were two fold: 1) to assess the validity of the proposed measurement protocol to measure changes in DPOAE and TEOAE after a concert; 2) to test the reliability of the oto-acoustic emission measurement system under field conditions; Initial results shows that it is possible to measure changes in hearing after exposures of relative short duration

The following abstract will be presented in poster format. The poster will be on display and the author will be at the poster from 3:20 p.m. to 3:40 p.m.

1pNSb5. The influence of ambient noise and headphone style on listening volume using a personal stereo system. Shih-Yi Lu (Department of Occupational Safety and Health, Chung Shan Medical University, No. 110, Sec. 1, Jianguo N. Rd., Taichung City 40201, Taiwan (R.O.C.), syluiosh@yahoo.com.tw), Kuei-Yi Lin, and Chiou-Jong Chen (Institute of Occupational Safety and Health, No. 99, Lane 407, Hengke Rd., Sijhih District, New Taipei City 22143, Taiwan (R.O.C.))

It is well known that exposure to excessive noise for long durations can cause a significant noise-induced hearing loss (NIHL). Although most research has focused on occupational sources of NIHL, there is growing concern about the potential damage caused by non-occupational noise exposure such as that from portable stereo system headphones. The purpose of

(<1.5 hours). There are large individual differences both in sound exposure levels as well as in the changes on oto-acoustic emissions produced by similar exposures. Current results will be presented.

3:00

1pNSb4. Study and research of noise in some industrial factory. Yidan Zhu (Beijing Municipal Institute of Labor Protection, Taoranting Road 55, Xicheng District, 100054, blue_clean@163.com)

Hearing loss is an occupational problem happened frequently in industrial factory. In the article, we investigate an industrial factory for several months, including the habit of worker and current noise situation of the environment. The conclusion shows some working area is exposed to high level noise, which is necessary to execute both noise reduction and hearing protection. And the whole region should be separated into different colors to warn the worker of different noise level.

current study was to measure the sound level generated by headphones of portable stereo system, and provide hearing conservation guidelines. Using a B&K Torso and a personal computer, output sound levels across volume control set by thirty participants were measured from headphone driven by music samples of five different genres. Three different styles of headphones (in-ear, circum-aural, supra-aural) were used to determine if styles of headphone influence sound level inside of ears. The study results suggested that the supra-aural headphone used by listener in noisy environment shall set a higher volume, in a result of a larger sound level in eardrum. The authors would like to thank the Institute of Occupational Safety and Health of Taiwan for the support that made the completion of this work possible.

MONDAY AFTERNOON, 14 MAY 2012

HALL C, 4:20 P.M. TO 6:20 P.M.

Session 1pNSc

Noise, Structural Acoustics and Vibration, and Architectural Acoustics: Vibration and Structure-Borne Noise in Buildings

James E. Phillips, Cochair
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C.M. Mak, Cochair
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Invited Papers

4:20

1pNSc1. Sound transmission through stiffened plates. Dayi Ou and Cheuk Ming Mak (Department of Building Services Engineering, The Hong Kong Polytechnic University, Hung Hum, Kowloon, Hong Kong, China, oudayi@gmail.com)

It is known that proper stiffening treatments may improve the sound insulation of plate-like structures. In this study, the sound transmission loss (STL) of a thin plate with stiffening treatments is investigated using a coupled finite element and boundary element method. A stiffened-plate element used in this method allows the plate to have arbitrary elastic boundary conditions and arbitrarily located stiffeners. Numerical studies of the effects of various stiffening conditions on the STL of a window are carried out. The boundary condition of the window lies somewhere between clamped and simply supported. The results can be used to guide the design of windows with better sound insulation performance.

4:40

1pNSc2. Evaluation of annoyance of vibration induced low-frequency noise from rail transit system. Dongxing Mao (Institute of Acoustics, Tongji University, Shanghai, 200092, China, dxmao@tongji.edu.cn), and Bijun Yang (Shanghai Marine Equipment Research Institute)

Vibration from urban rail transit system may cause vibration of buildings in subway system or vibration of bridges in elevated rail system, this vibration from building or bridges will generate so called vibration induced secondary noise with a strong low frequency character. Although the A-weighted level is below the limit of local regulations, frequent complains showed A-weighted level not a proper criteria for annoyance. Many research work have been done on low frequency noise, but diversify results were shown in the literature. In this paper, frequency and level depending property of annoyance from rail transit vibration induced low frequency noise was studied through subjective evaluation, and a LF-weighting curve was proposed on the basis of A-weighting curve with the correction below 500Hz, with mathematical description formulated and correction value tabled. Calculated levels according to the proposed LF-weighting curve showed high correlation with subjective annoyance. Results of subjective annoyance on sound level showed that annoyance growing exponentially with the increase of linear SPL. Thereafter, a comprehensive mathematical relation between annoyance and frequency and strength of noise was built, and proved to be effective in evaluating the annoyance caused by low frequency noise from rail transits.

5:00

1pNSc3. Isolation of structure-borne noise in buildings from exterior sources. George Paul Wilson (Wilson, Ihrig & Associates, Inc., 6001 Shellmound Street, Suite 400, Emeryville, CA, 94608, gpwilson2@gmail.com)

The increasing frequency of construction of new performance, residential and other noise sensitive facilities in locations with high amplitudes of ground-borne noise has required development of effective building isolation design configurations to reduce to acceptable values the structure-borne noise transmitted into the buildings from the foundation. Isolating a large building such as a concert hall or multi-story residence necessarily requires structurally separating the isolated building, or isolated parts of the building, from the foundation. The structural support is then provided by resilient bearings that must properly support the building gravity load, provide a controlled seismic restraint and structural stability, and provide the noise reduction required. The goal of this presentation is to demonstrate that the technology and materials now exist to allow placement of any large noise sensitive building, either a performance facility or a residential building, in any location where it is subject to ground-borne vibration and noise that would normally cause intrusion and be incompatible with the intended occupancy. The development of the gravity load support bearings and the preloaded seismic restraint rubber bearing concepts are presented along with examples of successful applications.

5:20

1pNSc4. Vibration isolation for concert hall next to busy street. James Phillips (Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608, jphillips@wiai.com)

A new, world-class performing arts center is currently being developed in a metropolitan center in the United States. The center will include two grand performance theaters. One of these theaters will be located close to a busy surface street. Vibration measurements conducted at the undeveloped site indicated that groundborne noise from street traffic would be audible within the completed theater unless measures were incorporated into the design to reduce vibration transmitted from the street to the interior of the theater. This will be achieved by structurally separating the performance area of the theater from the surrounding structure by supporting the theater on custom designed, resilient bearing pads. This paper discusses the vibration measurements taken, the projections of groundborne noise and the vibration mitigation measures that were incorporated into the structural design of the theater building to reduce groundborne noise to meet the project design criteria for background noise.

5:40

1pNSc5. A two domain method for design and evaluation of floor-ceiling assemblies using tapping machine impact sound pressure levels. John LoVerde and Wayland Dong (Veneklasen Associates, 1711 16th St, Santa Monica, CA 90404, jloverde@veneklasen.com)

Structure-borne impact noise from footfalls is a common noise complaint in multi-family residential buildings, and is currently described by a single-number metric such as IIC (using a tapping machine) or LA max (using an impact ball). However, research indicates that impact noise is characterized by two independent frequency domains: low frequency thudding and mid- to high-frequency noise from heel clicks, etc. The levels in these two domains vary independently with assembly design, so that two parameters are required to adequately characterize the impact insulation of an arbitrary assembly. The authors have developed a two-parameter system for evaluating impact noise [LoVerde and Dong, J. Acoust. Soc. Am. 122, 2954 (2007), J. Acoust. Soc. Am. 125, 2708 (2009)] intended to improve the design and evaluation of floor ceiling assemblies. Over the past 7 years, this system has been applied to design, evaluation, and testing of many projects. Criteria have been developed, and the real-world use of the proposed system is described, evaluated, and compared with the existing metrics.

Contributed Paper

6:00

1pNSc6. Elevator equipment noise mitigation for high-rise residential condominium. Jack B Evans (JEAcoustics 1705 West Koenig Ln, Austin, Texas 78756, Evans@JEAcoustics.com)

A new high-rise hotel and residential condominium building had elevator equipment rooms between residential spaces. During construction, elevator equipment noise was audible in adjacent unfinished residential spaces. The developer and architect requested evaluation by an acoustical engineer. Investigatory observations with acoustical and vibration measurements were conducted to determine sources and paths of vibration and noise transmission. Construction noise and the unfinished condition prevented measurements reflective of actual future conditions in occupied spaces, but 1/3

octave vibration measurements in the elevator equipment rooms and on wall surfaces of residential space indicated structure borne vibration transmission that could result in re-radiated audible sound. Sound spectrum measurements in the elevator equipment rooms compared with anticipated airborne noise transmission loss through the demising partition indicated potential levels of residual elevator equipment noise in residential spaces. Primary acoustic sources were determined by observations and validated by vibration and measurements in the equipment room. Anticipated noise levels due to airborne sound and structure borne vibration were compared to full-octave background noise Room Criteria (RC) to determine attenuation requirements. Recommendations for noise and vibration mitigation were developed. Mitigation measures implemented by the construction contractor will be enumerated with subjectively determine results.

MONDAY AFTERNOON, 14 MAY 2012

THEATRE 2, 2:00 P.M. TO 7:00 P.M.

Session 1pNSd

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

David Woolworth, Cochair
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Wing Tat Hung, Cochair
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Ulf Sanberg, Cochair
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Invited Papers

2:00

1pNSd1. Development of cleaning machine for drainage asphalt pavement in Japan. Kazuyuki Kubo (Public Works Research Institute, 1-6, Minamihara, Tsukuba, Ibaraki, Japan, k-kubo@pwri.go.jp)

Drainage asphalt pavement has become popular since 1990s in Japan, especially in expressways in order to improve traffic safety in rainy days and national highways in order to reduce traffic noise. It had regarded to be problem that these effects of drainage asphalt pavement could not last longer. For example, it could reduce traffic noise in 3 dB as its initial performance, while this effect would be lost in three years by clogging. To solve this problem, cleaning machines have been developed in Japan, which uses high pressure water and vacuum system to remove dust in drainage asphalt pavement. As a result, these machines are proved to be able to remove dust from drainage asphalt pavement, however, they can't recover its noise reduction performance to the initial level. Adding to say, the speed of these machines were around 5km/h and were regarded not to be appropriate for on-site maintenancework. Therefore, further development has been conducted to improve its workability. Finally, new cleanign machine using only high pressure air was developed. In this paper, short histry of this development and their actual use are reported.

2:20

1pNSd2. Acoustical performance assessment of Swiss low-noise road surface solutions in urban areas. Erik Bühlmann (Grolimund & Partner AG, Thunstrasse 101A, 3006 Bern, Switzerland, erik.buehlmann@grolimund-partner.ch), and Toni Ziegler (Grolimund & Partner AG, Entfelderstrasse 41, 5000 Aarau, Switzerland)

Recently Switzerland has experienced a new momentum in the construction of low-noise road surfaces in order to combat traffic noise in urban areas. Various regions have taken action and developed individual approaches to reduce traffic noise at the source. As a result, new requirements on the acoustical properties or on the void content of road surfaces were imposed, leading to the development of new products. The present work aims at both summarizing and cross comparing the acoustical performance of these products as well as analyzing the data produced to understand how the noise reduction was achieved. A large number of road surfaces (50 thin-layer asphalts, 30 SMA-like surfaces with increased void content) were therefore subjected to acoustical property monitoring using the CPX (close proximity) method. The acoustical performance of these road surfaces was quantified and evaluated in respect to AC and SMA surfaces, commonly used in urban areas in Switzerland and elsewhere in Europe. An approach was developed where measurement data

of surface texture, airflow resistance and sound absorption, combined with SPERoN modeling (a tyre/road surface contact model) was included in the analyses. This approach provided an evaluation of the contribution of isolated factors to the overall noise reduction.

Contributed Papers

2:40

1pNSd3. A prediction method for tire tread pattern noise based on characteristics of single tire tread block noise. Yinxiao Lu, Zhenyi Chen, and Jianmin Ge (Tongji University, No. 1239, Siping Road, Shanghai, China, 200092, 0940106001@tongji.edu.cn)

Tire tread pattern noise is the main source of tire noise. It contains several noise generation mechanisms such as 'air pumping' mechanism, 'air resonant radiation' and 'pipe resonances'. In the past 40 decades, a lot of researches have been made and many achievements have been won. But there is still no good prediction model or method for tire tread pattern noise. The paper provides a new method for tire tread pattern noise prediction. Make a superposition of noise signal of single tire tread block acquired in drum laboratory in time-domain and space. And give a noise prediction of tire which is full of tread block. The result of prediction and measurement will be compared to validate the effectiveness of the method.

3:00

1pNSd4. Two decades of noise control engineering and implementation for the Mass Transit Railway Corporation Ltd. Glenn Frommer (MTR Corporation Ltd, Fo Tan Railway House, No. 9, Lok King St, Fo Tan, Shatin, Hong Kong, gfrommer@mtr.com.hk)

Hong Kong relies on electrically powered railways as the backbone of its mass transit. With 84 heavy rail stations and 218 route length, more than 4.3 million out of Hong Kong's 7 million residents use the Mass Transit Railway (MTR) each working day. Hong Kong is also one of the most densely populated cities in the world. The density of the city, the large number of high-rise residential developments and the city's reliance on railways poses unique challenges for the railway noise control engineer when considering airborne noise, ground-borne noise, building services, speech intelligibility and noise within compartments. Starting with the design development of the Airport Railway, noise control has been successfully applied to all new railway lines and stations since 1992. Though cutting edge at that time, the methods are now standard for railway noise control throughout the region. This paper will provide an overview of the strategies and proven outcomes. A read across to operating railway noise will also be presented. A 'How - to' guideline will also be presented and the issue of noise as energy inefficiency will be discussed.

3:20

1pNSd5. Railway tunnel portal noise. Wilson HO, Banting Wong, Wylog Wong (Wilson Acoustics Limited, Unit 616, Technology Park, 18 On Lai Street, Shatin, Hong Kong, China, who@wal.hk), and Alson Pang (MTR Corporation, MTR Headquarters Building, Telford Plaza, Kowloon Bay, Hong Kong)

Railway tunnel portal noise is a concern in environmental impact assessment for railway projects. Railway noise is amplified inside the tunnel due to multiple reflections of highly sound reflective tunnel walls and such noise is then radiated from the tunnel portal. Noise Sensitive Receivers (NSRs) which are situated in close proximity of railway tunnel portal are adversely affected by the portal noise radiation in addition to the ordinary railway noise. Standard railway noise calculation procedures (Calculation of Railway Noise by Dept of Transport in UK, Transit Noise and Vibration Impact Assessment by Federal Transit Administration in USA, etc.), however, do not include a correction for such portal noise effect. This paper presents the experimental results of railway tunnel portal noise collected from Tai Po Kau tunnel on the East Rail Line in Hong Kong and proposes appropriate the tunnel portal noise corrections.

3:40

1pNSd6. A review of the interior noise and vibration characteristics of modern Chinese high speed train. Fusheng Sui, Anne Shen, Jiumei Cheng, and Minmin Yuan (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, sui@mail.ioa.ac.cn)

In the past few years, extensive experiments have been carried out to investigate the interior noise and vibration characteristics of modern Chinese high speed trains. The relationships between the vehicle noise and vibration sources and their contributions to the interior environment are discussed. Possible airborne and structural borne sound transmission paths are identified. The vibration responses of, and sound radiation from, the roof, the floor and the sidewalls of a typical power car are presented. The noise contribution from each of these components to the overall interior sound pressure levels is examined and results are discussed.

4:00–4:20 Break

4:20

1pNSd7. Use of floating slab track to control noise from rail transportation systems. George Paul Wilson (Wilson, Ihrig & Associates, Inc., 6001 Shellmound Street, Suite 400, Emeryville, CA 94608, gpwilson2@gmail.com)

One of the most effective provisions for control and reduction of way-side noise and vibration from rail transportation systems is floating slab track. A floating slab track support is effective in reducing ground-borne noise and vibration from subway and surface track installations and can be very effective in reducing airborne noise radiated from viaduct and bridge structures. This presentation includes an outline of the development of the design parameters and materials used for the small vertical dimension, relatively light weight and low cost floating slab track support configurations as first installed in the early 1970's. This design concept has been further developed and improved as experience was gained with the early installations and there are now a large number of successful installations in service. There have been numerous developments of variations in the design parameters to accommodate the specific conditions unique to each application regarding geology and nearby occupancies. The development of the design parameters and limitations are presented and example designs are described along with measured noise control effectiveness results.

4:40

1pNSd8. A comparative study of high-speed train noise prediction models. Guo Yanjie (Tongji University, No. 1239, Siping Road, Shanghai, China, 200092, ggairxc@gmail.com), Liang Junhai, Wang Dongzhen (CSR Sifang Locomotive Joint Stock Company Limited, No. 88, East Jinhong Road, Chengyang District, Qingdao Shandong 266111, China), and Ge Jianmin (Tongji University, No. 1239, Siping Road, Shanghai, China, 200092)

The present train noise prediction model in Chinese standards (HJ 453-2008 and HJ 2.4-2009) is only aimed for conventional train. Aerodynamic/rolling noise contribution of high-speed train noise are different from those of conventional train noise, thus, the Chinese prediction model might not be adaptable for the prediction of high-speed train noise. The aim of this paper is to discuss the noise prediction models of high-speed train above 300 km/h. Based on the comparative analysis of different models, our work mainly focus on noise source identification. Firstly the sound pressure level distribution on the surface of high-speed train body is investigated. In order to

overcome the shortage of single line source, the noise source is divided into three line sources of different heights parallel to each other according to the energy distribution of sound source. Then sound pressure level generated by the three line sources at the receiving point is calculated separately. Finally the predicting outcomes are compared with measured results to verify the reliability of the model.

5:00

1pNSd9. The relationship between structural vibration and noise of railway vehicles. Gong Lv (Tongji University, No. 1239, Siping Road, Shanghai, 200092, China, 495382721@qq.com), Junhai Liang, Jinzhu Liu (CSR Sifang Locomotive Joint Stock Company Limited, No. 88, East Jinhong Road, Chengyang District, Qingdao Shandong 266111, China), and Jianmin Ge (Tongji University, No. 1239, Siping Road, Shanghai, 200092, China)

With the rapid development of China's rail-vehicles industry, the property of the interior noise of the vehicles has become an important indicator of its quality. With the increasing speed of rail vehicles recent years, the airborne noise and the structural acoustics generated during the operation have had a great impact on the sound field inside the vehicle. In order to solve this problem, through the analysis of the noise and vibration produced during its operation, a rail vehicle sample is taken as the object of the study. Together with the basic principles of acoustics, the relationship between internal noise and the structural vibration and also the transmitting patterns of the structural vibration can be researched, so that it could provide a reference for the reduction of vibration and noise.

5:20

1pNSd10. Noise and vibration induced by a pantograph of high-speed trains. Zongguang Chen (Institute of Acoustics of Tongji University, 1239, Siping Road, Yangpu District, Shanghai, China, 10chenzg@tongji.edu.cn), Jianmin Ge (Institute of Acoustics of Tongji University, 1239, Siping Road, Yangpu District, Shanghai, China), Junshan Lin, Zhaojin Sun, and Jianqiang Guo (CSR Qingdao Sifang Locomotive and Rolling Stock Co., Ltd, No. 88, East Jinhong Road, Chengyang District, Qingdao Shandong 266111, China)

Pantographs mounted on the roof of the train body are high projections when they work. Along with the raising of the train speed, noise and vibration generated by pantographs are significant. This paper is aimed at evaluating the contribution of pantograph noise to overall noise of high-speed trains. A number of experiments consisting of noise and vibration measurements near and far from pantographs were performed to investigate aerodynamic noise radiated from pantograph and train roof vibration propagated from pantograph. Test data analysis consisted mostly of comparison of noise/vibration in different regions. The results of the experiment indicate that pantographs are main aeroacoustic sources and the train roof vibration which radiating noise into interior is extraordinary when the train speed is 300Km/h.

5:40

1pNSd11. Acoustical device using helmholtz resonator for the high-speed train noise barrier. Hyo-In Koh (Korea Railroad Research Institute, #360-1 Woramdong Uiwang City, hikoh@krii.re.kr), Jun-Ho Cho, and Joon-Hyuk Park (Korea Railroad Research Institute)

This study is primarily aimed at developing a measure to overcome the limited shielding performance of the noise barriers for the high-speed train. Up to the train speed of 300km/h and more the noise incidence angle and the source height change due to the pronounced aerodynamic noise source parts located at the higher positions compared to the height of the conventional rolling noise source. By means of the experimental analysis on the sound radiation characteristics and the sound pressure distribution around the noise barrier, a prototype of an acoustical attachment is produced based on the analytical model calculation and numerical analysis. The principle of

the Helmholtz resonator is used to optimize the acoustical impedance on the surface of the upper edge of the noise barrier. Using the model it was possible to find an appropriate acoustical property for the impedance according to the target frequency, sound incidence angle relative to the barrier and the receiver position. In this paper the results of the experiment in an anechoic room and from the out door experiment are shown and discussed.

6:00

1pNSd12. Acoustical insertion losses of coupled round edge barrier. Ho Ting Ng (Hong Kong Polytechnic University, Hung Hom, Kowloon, Hong Kong, alexhtng@yahoo.com.hk)

In this research project, Finite Element Numerical Modeling Method is used to compute the low frequency acoustical insertion losses of barriers with different edge shapes. Rectangular, Round Edge and Couple Round Edge barriers are included. The coupled round edge barrier is a hollowed round edge barrier with a slit on the round edge and a tube with a slit placed at the center of the hollow space of the round edge. The result shows that the coupled round edge barrier can produce a higher insertion loss on a specific range of low frequency noise in shadow zone. The shadow zone is also enlarged at the same time. The performance of the barrier is improved by the slit on the coupled round edge barrier under the dual resonator effect.

6:20

1pNSd13. The value of quiet areas in providing respite from traffic noise. Abigail L Bristow (School of Civil and Building Engineering, Loughborough University, Loughborough, LE11 2HY, UK, a.l.bristow@lboro.ac.uk), Petrina Rowcroft (URS/Scott Wilson), Paul Shields (URS/Scott Wilson 12 Regan Way, Chetwynd Business Park, Chilwell, Notts, NG9 6RZ), and Stuart Woodin (URS/Scott Wilson)

Prolonged exposure to unacceptable levels of noise is associated with a wide range of adverse impacts on human health, public amenity, productivity and ecosystems. As transport demand and development increases there is an associated reduction in the availability of areas that are perceived to be quiet or tranquil. The beneficial effects of access to quiet areas are not well understood. Critically there is a dearth of evidence on the value of benefits derived from quiet or green areas that offer a respite from traffic noise. Here we review the available evidence and propose a framework to assess the benefits that people derive from quiet areas and conversely the costs of loss of access to such areas. This requires a value to be placed on how residents, workers and visitors value publicly accessible quiet areas.

6:40

1pNSd14. Optimisation of noise reducing device intrinsic performances. Thomas Leissing, Jérôme Defrance, Philippe Jean, Catherine Guigou-Carter (CSTB, thomas.leissing@cstb.fr), and Jean-Pierre Clairbois (A-tech)

The work presented in this paper is part of the QUIESST European project, in which one of the objective is to perform optimisations of noise reducing devices. We present here optimisation results concerning the intrinsic performances of noise barriers. First the limits of these optimisations are determined: this concerns geometrical limitations as well as limitations on the number of materials. The intrinsic performances under interest are calculated using numerical simulations (the Boundary Element Method and the Transfer Matrix Method) in such a way that calculated values are as close as possible to quantities that one could measure using the CEN/TS 1793-4 -5 -6 standards. These simulations lead to reflection, transmission and diffraction performance values, which are expressed as a relative gain (or loss) to a reference noise barrier. The multi-objective optimisation strategy is then detailed and applied to nine coherent noise reducing device families. It is shown that using a specific set of parameters can largely improve the noise reducing device performances, and more importantly, that some selected set of parameters allow one to optimize several objectives simultaneously.

Session 1pPA

Physical Acoustics: Thermoacoustics

Anthony Atchley, Cochair
aaa9@psu.edu

Contributed Papers

2:00

1pPA1. Nonlinear acoustic impedances of thermoacoustic stacks with different structures in resonance pipes. Shu-yu Xiao, Sha Tao, Mei-chen Qiu, Huan Ge, Li Fan, Shu-yi Zhang, and Hui Zhang (Lab of Modern Acoustics, Institute of Acoustics, Nanjing University, Nanjing, 210093, b091120159@smail.nju.edu.cn)

The acoustic impedances of thermoacoustic stacks in the resonance pipes are measured by the method for measuring the nonlinear fluid resistances of porous materials. The thermoacoustic stacks with different structures of plate-type, pipe-type and meshed copper stacks are studied, in which the influences of the porosity and thickness of the stack and the operating frequency are evaluated experimentally. In the evaluations, the velocity variations in the stacks are neglected, so the thicknesses of the stacks must be much shorter than the acoustic wavelengths in the resonance pipe. The measured results show that the resistance of the stack keeps constant when the acoustic pressure level is low, but it increases rapidly with the tendency of quadratic function when the acoustic pressure level increases more than about 130 dB. Furthermore, both the linear and nonlinear acoustic resistances of the stacks increase with the thicknesses, while decrease with the increase of the porosity and/or the operating frequency. Finally, it is believed that the results of the influences of the structures and parameters of stacks on the acoustic impedances can be used in the nonlinear model of the thermoacoustic refrigerator.

2:20

1pPA2. Study of the acoustic impedance characteristics of linear alternator used in thermoacoustic generator. Jianwei Zhang (Graduate School of the Chinese Academy of Sciences, 100039, zhangjw1205@sina.cn), Zhengyu Li, and Qing Li (Technical Institute of Physics and Chemistry, CAS, 100190)

A thermoacoustic generator is a long-life, high efficient generator, it is composed of thermoacoustic heat engine and linear alternator. As a resonance system, the acoustic impedance match between both of them greatly influences the whole system's performance. In order to test the acoustic characteristics of the linear alternator, an experimental system was set up in TIPC. In this system, the displacement of alternator's piston and the pressure in the generator were measured. Correlation arithmetic was used to analyze the impedance characteristics of the linear alternator at several mean pressures. The analysis let us know the characteristics, which the linear alternator would be in the whole machine. It could be used to design the thermoacoustic generator. Some results were attained. They showed the experimental system could effectively work. The design of thermoacoustic generator has benefited from it. The authors gratefully acknowledge the Natural Science Foundation of China (Grant No. 10804114).

2:40

1pPA3. Investigation on shapes of the resonator related to external acoustic field in an open traveling-wave thermoacoustic generator. Xiujuan Xie (Key Laboratory of Cryogenics, Technical Institute of Physics and Chemistry, Chinese Academy of Sciences, Beijing 100190, China, xiexujuan@mail.ipc.ac.cn), Shaoqi Yang, Lihua Zhou (Key Laboratory of Cryogenics, Technical Institute of Physics and Chemistry, Chinese Academy of Sciences; Zhong guan cun dong lu 29, Haidian District, 100190, China; Graduate School of the Chinese Academy of Sciences, Beijing, 100049, China), and Qing Li (Key Laboratory of Cryogenics, Technical Institute of Physics and Chemistry, Chinese Academy of Sciences; Zhong guan cun dong lu 29, Haidian District, 100190, China)

Based on thermoacoustic effect, an open traveling-wave thermoacoustic generator could realize conversion between heat power and acoustic power, and then radiate sound into air space. The generator consisted of a loop tube and an open resonator. The shapes of resonator had tremendous effect on the external acoustic field radiated from the thermoacoustic generator. Uniform structure function was derived for different shapes of resonator. Acoustic wave equation taking the incident and reflected wave into account was established to acquire impedance distribution along the resonator. Based on the assumption of the point source, the external acoustic field far away from the system was obtained. A model between uniform structure function of resonator and the external acoustic field was obtained due to the impedance matching around the open end. The external acoustic field 0~1m far away from the system was measured experimentally, which verified the applicability of the model. Therefore, the optimal structural coefficient related to the highest SPL 1m far away from the open end was acquired.

3:00

1pPA4. Oscillation of sound wave in the straight-tube type miniature thermoacoustic system with closed-closed ends. Kenji Shibata (Doshisha University, Kyoto, Japan, dtl0171@mail4.doshisha.ac.jp), Shin-ichi Sakamoto (University of Shiga Prefecture, Shiga, Japan), Kentaro Kuroda, Yosuke Nakano, Takeshi Onaka, and Yoshiaki Watanabe (Doshisha University, Kyoto, Japan)

Downsizing of thermoacoustic system is discussed in order to be applied to the electronic equipments. In our previous studies for the miniature thermoacoustic system, the stable oscillation is successfully realized on the straight-tube with opened-closed ends type although the inner working fluid cannot be shut in. To realize the working fluid be shut in the tube and keeping the stable oscillation, a new system is combined of two tubes with different cross-sectional area. The large sized system is applied instead of the small sized one because of the measurement difficulty. As the results, it was confirmed that the efficiency of the energy conversion was improved by

using the system which combines two kinds of tubes with different cross-section tube. The reason of the efficiency improvement was estimated that the resonance became to be strong by the overlap of two kinds of resonances. This estimation was confirmed by changing the ratio of length of connecting tubes with different cross-section. The efficiency of the energy conversion was improved by controlling of the acoustic field. Those were realized by connecting of two tubes with different cross-sectional area.

3:20

1pPA5. Investigation on streaming sources in thermoacoustic prime mover. Richard Paridaens, Smaïne Kouidri, and Fathi Jebali Jerbi (Limsi-Cnrs BP 133 91403 Orsay cedex France, richard.paridaens@imelavi.fr)

Thermoacoustic devices either prime mover, heat engines or refrigerators are not known for their high efficiency. Even though these systems have many advantages regarding environmental constraints, they are not yet used in the industrial applications. Energy conversion efficiency improvement of thermoacoustic systems is now in the priority of the thermoacoustic community. One of the reasons of the relative low efficiencies is in the physical understanding which is not well achieved. The appearance of steady mass flow of second order usually called streaming and superimposed to the oscillating flow in these systems is shown as an important dissipating energy phenomenon. From energy consideration and despite their low level, this DC flow involves heat transfer to the wall which is undesirable loss mechanism. This phenomenon which is a quite old topic is still widely investigated experimentally and theoretically. The design, construction and performance measurements of the traveling wave thermoacoustic engine will be presented and discussed. A non-linear acoustic approach has been developed in order to determine the contribution of the different sources of streaming generation. The purpose is to emphasize on the physical interpretation of each source.

3:40

1pPA6. Acoustic field measurements in a standing wave thermoacoustic refrigerator using time-resolved particle image velocimetry. Philippe Blanc-Benon and Emmanuel Jondeau (Laboratoire de Mécanique des Fluides et d'Acoustique, UMR CNRS 5509, Ecole Centrale de Lyon, Université de Lyon, France, Philippe.Blanc-Benon@ec-lyon.fr)

A standing-wave thermoacoustic refrigerator consists of a stack of plates placed in an acoustic resonator. Two heat exchangers are located at each stack extremity. The thermoacoustic effect takes place in the thermal and viscous boundary layers along each plate of the stack. It results in a heat transport along the plates and in a temperature difference between the two stack ends. In such devices, the full understanding of the heat transfer between the stack and the heat exchangers is a key issue to improve the global efficiency of these devices. The aim of this work is to investigate the vortex structures, which appear at the ends of the stack and modify the heat transfer. Here, the aerodynamic in the gap stack-exchanger is characterized using a time-resolved particle image velocimetry technique. Measurements are performed in a device operating at a frequency of 200 Hz. Instantaneous velocity fields are recorded at a frequency of 3125 Hz (ie 15 maps per acoustic period). Measurements show that vortex shedding occur at high pressure levels, when a nonlinear acoustic regime prevails, leading to an additional heating generated by viscous dissipation in the gap and a loss of efficiency.

4:00–4:20 Break

4:20

1pPA7. Low temperature drive of a straight tube thermoacoustic system filled with mixture gases by using numerical calculation. Yosuke Nakano (Doshisha University, Kyoto Japan, yosuke547@gmail.com), Shin-ichi Sakamoto (University of Shiga Prefecture, Shiga Japan), Kentaro Kuroda, Kenji Shibata, Takao Tsuchiya, and Yoshiaki Watanabe (Doshisha University, Kyoto Japan)

As the temperature ratio of the both ends of the stack increases gradually and reaches a critical value, sound waves begin to oscillate. This temperature ratio is called the onset temperature ratio. The thermoacoustic systems can use waste heat effectively, and are applied as electrical generation and

cooling systems. However, the required temperature to drive the current thermoacoustic systems is higher than the temperature of waste heat, and energy conversion efficiency of the systems is just only a few percent. In order to use of the systems effectively, these need to be improved. The onset temperature ratio and energy conversion efficiency are dependent on the geometry of the systems and working fluids. In this report, it is focused on the gas in the systems, and the mixture gas of argon and helium was used as working fluids. The influence of the onset temperature ratio and energy conversion efficiency was investigated, when the mixture ratio was changed. These are calculated by using the linear stability theory and a transfer matrix method. As a result, the mixture ratios which give the minimum onset temperature ratio and the maximum energy conversion efficiency have been calculated.

4:40

1pPA8. Particle Swarm Optimization method in thermoacoustic problems. Hussein Chaitou, Philippe Nika, and Guillaume Layes (Institut Femto-ST / UMR CNRS 6174, Parc Technologique, 2 avenue Jean Moulin, 90000 Belfort, France, hussein.chaitou@gmail.com)

Thermoacoustic engine systems convert heat power into acoustic power which is useful to pump heat or to generate electricity. To construct a robust and useful thermoacoustic device, both the acoustic power produced and the exergetic efficiency of this device should have acceptable and meaningful values. In order to attain this objective, an optimization study is strongly recommended and required. In the literature of thermoacoustic research, we found only some limited synthetic optimization methods. This paper presents a new study that incorporates the Particle Swarm Optimization (PSO) method for the first time in the thermoacoustic research in order to optimize the two objective functions, i.e. the acoustic power and the exergetic efficiency. The importance of using the PSO method in thermoacoustic research is highlighted and extensively investigated. In addition, significant conclusions, which are useful for the design of new thermoacoustic engines, are discussed.

5:00

1pPA9. Factors influencing on matrix H of acoustic field modulation in the thermoacoustic system. Lihua Zhou (Key Laboratory of Cryogenics, Technical Institute of Physics and Chemistry, Chinese Academy of Sciences, Beijing, 100190, China; Graduate School of the Chinese Academy of Sciences, Beijing, 100049, China, zlh13@163.com), Xiujian Xie, and Qing Li (Key Laboratory of Cryogenics, Technical Institute of Physics and Chemistry, Chinese Academy of Sciences, Beijing, 100190, China)

In the system of acoustic field modulation in the regenerator by double loudspeakers (J. Acoust. Soc. Am. 130(5), 2709-2720 (2011)), the relationship among the pressure, velocity and driving voltages is described by a matrix H. H is a function of configuration and location of regenerator, operating frequency and temperature gradient between heat exchangers. The modulation range of acoustic field will vary with the variation of H. In this paper, the influences of factors (y_0/l_0 , L, f, dT_m/dx) on the matrix H are researched. The order of these factors effecting on matrix H is given and discussed. The results show that location L and frequency f have important influence on matrix H than y_0/l_0 and dT_m/dx , which are applicable in acoustic field modulation.

5:20

1pPA10. The influence of thermoacoustic regenerator on a traveling-wave acoustic field. Gang Zhou, Xin Huang, and Qing Li (Key Laboratory of Cryogenics, Technical Institute of Physics and Chemistry, Chinese Academy of Sciences, Beijing, 100190, China, zhougang@mail.ipc.ac.cn)

Thermoacoustic device can realize conversion between the heat energy and acoustic energy. Regenerator is the core of a thermoacoustic engine or refrigerator, which consists of smooth or tortuous porous media, such as parallel plates or stainless stacked-screen. Due to regenerators, the practical acoustic field is neither a pure standing wave nor a pure travelling wave. In fact, the position, structural parameters of the regenerator and temperature gradient between heat exchangers will strongly influence the oscillating pressure and velocity distribution in the acoustic field, which will bring about different performance for a practical thermoacoustic engine or refrigerator. In this paper, based on linear thermoacoustic theory, a mathematical model of a $1/2$ wave-length duct with a regenerator driven by speakers will

be built and the influence of the regenerator on the acoustic field distribution will be simulated and analyzed. This research is helpful for comprehensively understanding coupling mechanism between the acoustic field and the regenerator. The work was supported by the National natural science foundation of China (Grant no. 10904154).

5:40

1pPA11. Modulating of traveling-standing wave acoustic field in the thermoacoustic resonator. Xin Huang, Gang Zhou, and Qing Li (Key Laboratory of Cryogenics, Technical Institute of Physics and Chemistry, Chinese Academy of Sciences, Beijing, 100190, China, huangxin@mail.ipc.ac.cn)

In this paper, a double-loudspeakers model to modulate the acoustic field in the resonator is presented. The acoustic field is modulated by changing the driving conditions, including the amplitude and the phase difference of the driving voltages. A numerical simulation for the acoustic field in the resonator without regenerator is carried out. The calculation results indicate that any traveling-standing wave acoustic field can be obtained by changing the driving conditions. The acoustic field in the resonator with regenerator is also simulated. It is found that the acoustic field in the regenerator can be modulated in a wide range and several targeted acoustic field conditions can be obtained feasibly, which are useful for achieving the optimal thermoacoustic conversion. An experimental device has been constructed and tested. The acoustic field are measured and reconstructed under different driving conditions. The experiment results are in good agreement with the simulations. The device provides a platform for our further study on the thermoacoustic characteristic of regenerator in different acoustic field. This work was supported by the National Natural Science Foundation of China (Grant No. 10904154).

6:00

1pPA12. Real-time measure system of linear alternator efficiency in Thermoacoustic electricity generator. Zhengyu Li (Key Laboratory of Cryogenics, Technical Institute of Physics and Chemistry, Chinese Academy of Sciences, 100190, lizhengyu@mail.ipc.ac.cn), Jianwei Zhang (Graduate School of the Chinese Academy of Sciences, 100039), Gang Zhou, Zhongjun Hu, and Qing Li (Key Laboratory of Cryogenics, Technical Institute of Physics and Chemistry, Chinese Academy of Sciences, 100190)

Thermoacoustic electricity generator could convert heat to electric power. It is composed of thermoacoustic engine and linear alternator. In TIPC, a traveling wave thermoacoustic electricity generator had been built and tested. Long resonator was replaced by linear generator in the thermoacoustic electricity generator. There were mechanical impedance, acoustic impedance and electric impedance. Only adjustment parameter was electric load when the generator ran. Electric load could affect the acoustic impedance of linear alternator, which was close related with efficiency from acoustic power to electric power. In order to optimize the electric load in real time, a kind of on-line measure method was supposed and tested. Volume flow rate and pressure were measured by resistance strain gauge and piezo-electric pressure sensor respectively. Correlation arithmetic was used to evaluate the acoustic power. On-line analysis of acoustic power and electric power made it possible to measure efficiency. Some tests were done to verify this technology. The improvement of electric power has benefited from the technology. The authors gratefully acknowledge the Natural Science Foundation of China (Grant No. 10804114).

MONDAY AFTERNOON, 14 MAY 2012

S423, 2:00 P.M. TO 6:00 P.M.

Session 1pPP

Psychological and Physiological Acoustics and Animal Bioacoustics: Open Challenges in Auditory Scene Analysis II

Mounya Elhilali, Cochair
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Invited Paper

2:00

1pPP1. Auditory scene analysis: a competition between auditory proto-objects? Susan Denham (University of Plymouth, Drake Circus, Plymouth PL4 8AA, UK, S.Denham@plymouth.ac.uk)

Many sound sources can only be recognised from the pattern of sounds they emit, and not from the individual sound events that make up their emission sequences. We propose a new model of auditory scene analysis, at the core of which is a process that seeks to discover predictable patterns in the ongoing sound sequence. Representations of predictable fragments are created on the fly, and are maintained, strengthened or weakened on the basis of their predictive success. Auditory perceptual organisation emerges from the competition between these representations (auditory proto-objects). Rather than positing a global interaction between all currently active proto-objects, competition is local and occurs between proto-objects that predict the same event at the same time. The model has been evaluated using the auditory streaming paradigm, and provides an intuitive account for many important phenomena including the emergence of, and switching between, alternative organisations, the influence of stimulus parameters on perceptual dominance, switching rate and perceptual phase durations, and the build-up of auditory streaming.

Contributed Paper

2:20

1pPP2. Tuning the Hopf cochlea towards listening. Florian Gomez, Victor Saase, Nikolaus Buchheim, Richard Bumann (Institute of Neuroinformatics, University and ETH Zurich, Winterthurerstrasse 190, CH-8057 Zurich, fgomez@ini.phys.ethz.ch), Liang Yan (College of Marine Engineering, Northwestern Polytechnical University, Xi'an 710072, China), and Ruedi Stoop (Institute of Neuroinformatics, University and ETH Zurich, Winterthurerstrasse 190, CH-8057 Zurich)

In a Hopf cochlea, coupled Hopf oscillators of individual frequency each, account for the active amplification of the auditory input. All salient nonlinear aspects of hearing can be traced back to the physical properties of the Hopf oscillators. At each location along the cochlea, the amplification

strength is effectively governed by a single real parameter characterizing the distance of the Hopf oscillator from the Hopf-bifurcation point. Using these parameters, given a mixture of input signals (e.g., a set of musical instruments) it should be possible to tune the cochlea towards a single sound component. Introducing an autocorrelation-based tuning measure, we demonstrate the tunability of the Hopf Cochlea on recorded real-life instruments of different timbres and pitches. Despite the strongly nonlinear and therefore interaction-prone nature of the device, strong and simple tuning patterns permit an easy tuning to sounds of varying pitch. Our insights may prove essential for gaining further understanding in the problem of selective auditory attention in a multi-source environment, commonly known as the "cocktail party environment".

Invited Papers

2:40

1pPP3. Analyzing objects through time. Barbara Shinn-Cunningham (Center for Computational Neuroscience and Neural Technology, Boston University, 677 Beacon St., Boston, MA 02215, shinn@cns.bu.edu)

Most researchers accept that the act of preparing to listen for a source with a particular attribute (e.g., from a particular location or a particular speaker) causes preparatory changes in how subsequent sound inputs are processed, and thus how an auditory scene is analyzed. However, the dynamics of how humans parse an auditory scene are complex and depend upon not only this kind of volitional attentional modulation but also automatic processes. Moreover, perceptual automatic processes depend both on the recent statistical properties of input sound (e.g., regularities that determine whether an input is unexpected / novel versus predictable) and, as at least based on results from our own lab, on what perceptual object was the focus of attention in the preceding moments. For instance, we find that once a given stream of sound is the focus of attention, subsequent sound elements that are perceptually similar are perceptually enhanced in an obligatory process, even in the absence of volitional attentional focus. Similarly, changes in sound attributes that cause perceptual discontinuities disrupt processing of auditory inputs. These factors, which strongly impact human processing of auditory scenes, will be discussed and contrasted with the processing governing many machine algorithms for auditory scene analysis.

3:00

1pPP4. Effect of source-motion and self-motion on the resetting of auditory scene analysis. Hirohito Kondo (NTT Communication Science Laboratories, NTT Corporation, Atsugi, Kanagawa 243-0198, Japan, kondo.hirohito@lab.ntt.co.jp), Daniel Pressnitzer (UMR 8158, CNRS and Université Paris Descartes, Paris F 75006, France), Iwaki Toshima, and Makio Kashino (NTT Communication Science Laboratories, NTT Corporation, Atsugi, Kanagawa 243-0198, Japan)

Auditory scene analysis needs to parse the incoming flow of acoustic information into perceptual streams, such as distinct musical melodies or sentences from a single talker. Previous studies have demonstrated that the formation of auditory streams is not instantaneous: rather, streaming builds up over time and can be reset by sudden changes in the acoustics of the scene. The present study examined the effect of changes induced by voluntary head motion on streaming. A telepresence robot in a virtual-reality setup was used to disentangle all potential consequences of head motion: changes in acoustic cues at the ears, changes in apparent sound location, and changes in motor processes. The results showed that self-motion induced resetting of auditory streaming. An additive model analysis further revealed that resetting was largely influenced by acoustic cues and apparent sound location rather than by non-auditory factors related to head motion. Thus, low-level changes in sensory cues can affect perceptual organization, even though those changes are fully accounted for by self-motion of the listener. It is suggested that our results reflect a widely distributed neural architecture for the formation of auditory streams.

Contributed Paper

3:20

1pPP5. Empirical AM-FM decomposition of auditory signals. Qinglin Meng, Meng Yuan, Jianping Zhao, and Haihong Feng (Shanghai Acoustics Laboratory, Chinese Academy of Sciences, No. 456, Xiaomujiao Road, Shanghai, China, 200032, mengqinglin08@gmail.com)

Cochlea, known as 'fourier analyzer' in auditory system of mammals, codes the mechanical wave to electrical signals tonotopically. Meanwhile, the auditory coder also works in temporal mode, where envelope (E, i.e. amplitude modulation (AM)) and temporal fine structure (TFS, i.e. frequency modulation (FM)) are both significant but not exactly defined. In several recent researches (e.g. [Smith, et. al, 2002]), hilbert transform (HT) was utilized to extract the E and TFS cues. Though HT is mathematically rigorous,

it is lack of clear physical meaning on AM-FM. This research introduces the empirical mode decomposition (EMD) to auditory research field. EMD was designed for analysis of non-linear and non-stationary data including natural sound. It is observed that the outputs of a auditory filter (e.g. Gammachirp), which was born as narrow-band filters, are approximately intrinsic mode functions (IMF) that admit well-behaved hilbert transforms (implying having physical significance). However, non-narrow-band filter (NNBF) output can be decomposed into two obvious IMFs or more, implying that HT is not suited to NNBF. EMD with HT provides auditory researchers one more perspective on AM-FM decomposition of auditory signals. This work is supported by National Natural Science Foundation of China (11104316) and Shanghai Natural Science Foundation (11ZR1446000).

3:40

1pPP6. New evidence for audiovisual speech scene analysis: Low level interaction between auditory streaming and visual cues in speech perception. Frédéric Berthommier (GIPSA-Lab/DPC, 11 rue des Mathématiques, 38402 Saint Martin d'Herès, France, Frederic.Berthommier@gipsa-lab.grenoble-inp.fr), and Jean-Luc Schwartz (GIPSA-Lab/DPC)

We have proposed in the last years that there could exist a level of audiovisual binding (audiovisual speech scene analysis) previously to audiovisual fusion and speech comprehension. In this paper, we propose a new paradigm in which a competition exists between a non-speech auditory stream and an audiovisual speech stream for incorporating the pre-voicing component (PVC) excised from a /b/. The auditory stream is composed of low intensity PVCs at a regular rhythm of about 1Hz, interleaved with /pa/ syllables at about 0.3Hz, forming the speech stream. The listeners' task is an online forced choice between /p/ and /b/, which relies on the level of capture of a PVC within the speech stream (hence perceived as /b/) or within the auditory stream (speech being hence perceived as /pa/). The regular PVC stream has the tendency to capture the common PVC. Surprisingly, the vision of lips movements enhances this capture effect, and the phonetic fusion between the common PVC and the /p/ is disfavored. To characterize this effect, two supplementary experiments have been carried out, suggesting this effect is (at least partly) speech specific. This supports our proposal for an audiovisual speech binding mechanism.

4:00–4:20 Break

Contributed Papers

4:20

1pPP7. Using low-frequency threshold interaural time differences to test models of binaural hearing. Tianshu Qu (Key Laboratory on Machine Perception-Ministry of Education, Peking University, Beijing, 100871, China, qutianshu@gmail.com), and William Hartmann (Michigan State University, 4208 BPS Bldg., East Lansing, MI, 48824)

All models for detecting an interaural time difference (ITD) begin with model cross-correlator cells. Models differ in their inputs (excitatory/inhibitory) and in the distribution of cells with respect to interaural delay and frequency. For instance, the Jeffress model postulates a broad distribution on delay. Also, alternative binaural displays are based on different moments of the cross-correlation. For instance, the centroid is based on the first moment. The different models predict different frequency dependences of the threshold ITD in the limit of low frequency. Limiting behavior was computed for the various models using different assumptions about the frequency dependence of the synchrony of inputs to cross-correlator cells and about the sharpness of the rate-ITD function from the cells. The results were compared with the measured low-frequency functional behavior of ITD thresholds for four human listeners, which ranged from $f^{-0.8}$ to $f^{-1.4}$. These measured exponents disagree with predictions from some combinations of models. In particular, the popular centroid display within the Jeffress model tends to lead to slopes that are steeper than observed experimentally. [Work supported by the National Natural Science Foundation of China grant 61175043 and the Air Force Office of Scientific Research grant 11NL002.]

4:40

1pPP8. Platform for virtual auditory environment real time rendering system. Chengyun Zhang and Bosun Xie (South China University of Technology, No. 381, Wushan Rd., Guangzhou, P.R. China, 510641, zhang.cy@tom.com)

By dynamically synthesizing binaural signals in free-field and reflective environment, a virtual auditory environment (VAE) real time rendering system recreates realistic auditory events or perceptions for listeners. VAE systems have been applied in various fields, such as the research of binaural hearing, multimedia and virtual reality, among others. In present work, a PC and C++ language-based VAE system is designed and implemented. Schemes for improving the performances of the system, including multiple virtual source synthesis, auditory distance perception, dynamic information simulation for multiple degrees of freedom of listener, as well as dynamic characters of the system, are proposed. The results from measurement

indicate that the system is capable of synthesizing 280 virtual sound sources (including free-field sources and image sources for reflections) simultaneously in conventional working mode, or 4500 virtual sources in proposed PCA (principal components analysis) working mode. The update rate is 120 Hz, and the system latency is 25.4ms. A set of psychoacoustic experiments also validate the performance of the system. The function extension of VAE can serve as a flexible and powerful platform for binaural and virtual reality research.

5:00

1pPP9. Subjective evaluation of 5.1 channel signals' reproduction by two loudspeakers with small spacing angle. Dan Rao and Fang Ming (Physics Dept., South China University of Technology, Wusan RD., Guangzhou, China, 510641, phdrao@scut.edu.cn)

In some circumstances (such as looking at TV), it is needed to use two loudspeakers with small spacing angle to reproduce multi-channel surround sound. The performance of reproduction with closely spaced loudspeakers was concerned. In this paper, the performances of two-loudspeaker reproduction were evaluated by a subjective listening test. Using 5.1-channel signals' reproduction with standard five-loudspeaker arrangement as reference, two reproduction methods, downmixing and virtual reproduction were graded according to three attributes, spatial impression, timbre and global impression with picture. Five-grade impairment scale was adopted in test assessment and 17 subjects with listening experiences participated in the test. Test results show that the performance of downmixing method is degraded with decreasing spacing angle of loudspeaker pair, and the performance of virtual reproduction method almost is not affected by spacing angle within the range of less than 15 degrees. In addition, subjective score of virtual reproduction is better than that of downmixing reproduction in all three experimental spacing angles. Therefore, virtual reproduction method can improve the two-loudspeaker reproduction performance of multi-channel surround sound signal compared to the common downmixing method.

5:20

1pPP10. The use of relative weights to assess perceptual segregation in a concurrent profile analysis task. Yi Shen (Department of Cognitive Sciences, University of California, Irvine, CA 92617, shen.yi@uci.edu)

A series of experiments were conducted to address the need of a psychophysical tool to measure the perceptual segregation of concurrent sources. The experiments measured listeners' sensitivity to the spectral profile of a

target sound embedded in a concurrent masker. Both the target and masker were harmonic complexes, which were presented at different fundamental frequencies in order to investigate the effects of this acoustic cue on segregation. The task was designed so that it either strongly encouraged the segregation of the two complexes (task-driven design) or it did not necessarily require segregation (listener-driven design). In both cases, the degree of segregation was assessed by deriving the relative decision weights on the target and masker. Larger differences between the target and masker weights were found as the fundamental frequency difference between the two complexes increased (0.5 – 15 semitones), suggesting that listeners were more successful in selectively attending to the target alone at larger fundamental frequency separations. Although quite different thresholds were obtained for the task-driven and listener-driven designs, the estimates of the decision weights were consistent across the two task designs, indicating that listeners' motivations did not influence the usefulness of the periodicity cue in segregating concurrent sounds.

5:40

1pPP11. Detection of spectral changes induced by a break in sound correlation in younger adults and older adults. Tianshu Qu, Shuyang Cao, Xun Chen, Ying Huang, Xihong Wu (Peking University, qutianshu@cis.pku.edu.cn), Bruce Schneider (University of Toronto Mississauga), and Liang Li (Peking University)

Detecting a transient break in correlation (BIC) between correlated sounds is much easier when presented over two loudspeakers than when presented over two headphones. However, older adults benefit less than younger adults from a change from headphone to loudspeaker presentation (Ear and Hearing, (30) 273-286, 2009), suggesting an age-related reduction in sensitivity to monaural and/or binaural spectral cues provided by comb filtering. In this study, the monaural spectral cues present in the sound field were isolated and extracted, and then presented over headphones to younger adults and older adults with clinically normal hearing. Compared to younger adults, older adults exhibited a reduced sensitivity to the monaural spectral cues, particularly when an inter-loudspeaker time interval was introduced.

MONDAY AFTERNOON, 14 MAY 2012

S222, 2:00 P.M. TO 4:40 P.M.

Session 1pSA

Structural Acoustics and Vibration and Noise: Noise Control Methods for Aerospace Structures I

Gopal P. Mathur, Cochair
gopal.p.mathur@boeing.com

Invited Papers

2:00

1pSA1. The sound insulation of composite cylindrical shells; a comparison between a laminated and a sandwich cylinder. Chongxin Yuan (Technology University of Delft, c.yuan@tudelft.nl), Bert Roozen, Otto Bergsma, and Adriaan Beukers

The fuselages of aircraft are modeled as cylinders in this paper, and the sound insulations of a sandwich cylinder and a laminated cylinder are studied both experimentally and numerically. The cylinders are excited by an acoustic pressure and a mechanical force respectively. Results show that under acoustic excitation, the sandwich cylinder and the laminated one have a similar sound insulation below 3000 Hz, but the sandwich cylinder has a much larger sound insulation at higher frequencies. Under mechanical excitation, the sandwich cylinder is also more beneficial, showing a larger sound insulation above 800 Hz.

2:20

1pSA2. Noise control options for aircraft floors. Jeffrey Weisbeck (ITT Enidine Inc, 7 Centre Dr, Orchard Park NY 14127, Jeff.Weisbeck@itt.com), Samir Gerges, Marcelo Bustamante, and Julio Cordioli (Federal University of Santa Catarina, Mechanical Engineering Department, Noise and Vibration Laboratorio, Florianopolis, SC, Brazil)

Aircraft manufactures seek light weight, cost effective technologies to reduce cabin noise levels. Customers and regulatory bodies are demanding lower noise levels in the cabin. Composite light weight structures, used to increase fuel efficiency, often increase input acceleration levels due to their relatively high stiffness and low damping. Therefore, noise control solutions must provide additional attenuation to meet the challenges of fuel efficiency and lower cabin noise levels. Aircraft floors are large radiating bodies that must be addressed. This paper explores noise control options for aircraft floors. Various isolation and damping methods are investigated. Analytical estimates are compared to empirical measurements for a sample aircraft floor.

2:40

1pSA3. An experimental characterization of the acoustically dissipative properties of light-weight nanocomposite polyurethane foams augmented with carbon nanotubes. Andrew Willemsen (Department of Mechanical Engineering-Engineering Mechanics, Michigan Technological University, Houghton, MI 49931, amwille@mtu.edu)

Flexible, open-cell polymer foams are among the most commonly used and effective materials for passively dissipating noise and vibration. Their unique microcellular structure results in materials which are light weight but still highly absorptive, as well as relatively strong and stiff, making them particularly useful for weight-sensitive applications, such as aircraft cabin noise reduction. "Nanocomposite" polymer foams, which are synthesized from polymer materials containing reinforcing nano-scale fillers, have been shown to have altered morphological and mechanical properties in comparison to conventional counterparts. These same morphological and mechanical properties fundamentally control the acoustic absorption and vibration damping provided by polymer foams. Thus the alteration of these properties by nano-scale reinforcing materials can potentially be exploited to enhance the dissipative properties of these materials. In this study, various composites of polyurethane foam and multi-walled carbon nanotubes were synthesized and then experimentally characterized to observe the effect on noise and vibration dissipation. Sound absorption coefficient and loss factor were measured, along with a number of related material parameters. Results indicate inclusion of carbon nanotubes can increase the ratio of the sound absorption coefficient to weight for polymer foam treatments, dependent on both the carbon nanotube particle size and weight fraction.

3:00

1pSA4. Bi-objective optimization for the vibro-acoustic performance of a double-wall panel. Jie Zhou, Atul Bhaskar, and Xin Zhang (Faculty of Engineering and Environment, University of Southampton, SO17 1BJ, UK, Jie.Zhou@soton.ac.uk)

This paper presents simultaneous optimization of double-walled panels for minimum weight and maximum acoustic transmission loss. A sandwich construction having poroelastic lining in the core is considered. A general formulation — to calculate transmission loss as a function of frequency of the incident wave in the presence of mean external flow on one side of the double-wall panel — is presented. Biot's theory is used to simulate the poroelastic material. Three types of sandwich configurations are considered and the transmission behavior is studied for a range of Mach numbers and over a frequency band. The objective is to simultaneously minimize the sound transmission and the structural weight. Pareto fronts are obtained. The trade-off between weight and acoustic performance is systematically studied.

3:20

1pSA5. Comparative analysis on acoustic radiation modes of typical structures. Lu Dai, Tiejun Yang, Jingtao Du, Yao Sun, Jianchao Dong, and Xinhui Li (Harbin Engineering University 150001, dailu1026@yahoo.cn)

Acoustic radiation modes have received increasing attention in the areas of structural acoustic radiation and active structural acoustic control during the past few years. In this paper, a comparative study on the acoustic radiation modes of several typical structures and their associated radiation efficiencies is presented. The present work is also undertaken to extend the acoustic radiation modes into more complex structures, i.e. a thin cylindrical

shell, since it was rarely described in the literatures. The two analytic approaches for deriving acoustic radiation modes are reviewed briefly first. Numerical examples and comparative analysis are performed from one-dimensional problem to two-dimensional and three-dimensional problems. A grouping characteristic of acoustic radiation modes and their corresponding radiation efficiencies is observed. The shapes of the acoustic radiation modes follow the order of uniform variation, linear, quadratic and high-order variation. It is interesting that the acoustic radiation modes of a cylindrical shell exhibit symmetric and anti-symmetric shapes in the circumferential direction, similarly to its structural modes.

3:40

1pSA6. Effects of non-linear eddy-airfoil interactions on the acoustic radiation of a thin wing. Avshalom Manela (Faculty of Aerospace Engineering, Technion, Haifa, Israel, avshalom@aerodyne.technion.ac.il)

We study the combined effects of flow unsteadiness (incident vorticity) and external forcing (leading edge animation) on the vibroacoustic radiation of a thin rigid wing. Applying potential flow theory, non-linear coupling between wing motion and flow vorticity trajectory is calculated using the method of conformal mapping. At first, the dynamical problem is formulated and studied. The dynamical description then serves as an effective source term to evaluate the acoustic field. The formulation of the aeroacoustic problem is based on a compact-body acoustic analogy, thus avoiding the traditional difficulty in obtaining the weak acoustic far field from direct numerical simulations. The results identify the airfoil as a dipole-type source, and analyse the significance of non-linear eddy-airfoil coupling on the system acoustic signature. The effect of adding elastic degrees of freedom to the wing, in the form of "passive" linear and torsional springs, is analysed as a mean for monitoring the system acoustic radiation.

4:00–4:20 Break

4:20

1pSA7. Numerical simulation of the transmission loss of plates. Rafael Piscoya, Ralf Burgschweiger, and Martin Ochmann (Beuth Hochschule für Technik Berlin, University of Applied Sciences, Luxemburger Str. 10, 13353 Berlin, Germany, piscoya@beuth-hochschule.de)

Numerical simulations for estimating the transmission loss of plates can be an important alternative to measurements when there is no access to transmission loss test facilities. Furthermore, parametric studies and design changes can be made easily and faster. This work presents a method to calculate the transmission loss of plates placed between a source and a receiver room using an iterative approach. The sound radiation due to the vibration of the plates is solved with a Boundary Element formulation while the motion of the plate is determined using a Finite Element formulation with the sound pressure as the exciting force. The starting point is the blocked pressure approximation. The real pressure on the plate and its displacement are obtained after some iterations. If no damping in the plate is considered, poor or no convergence is expected at the resonant frequencies of the plate. This problem is avoided introducing some damping in the plate as well as in its fixation (boundary). With this approach, the use of existing techniques to accelerate the calculations that are already developed for the BEM, e.g. the Fast Multipole Method and for the FEM, e.g. the Model Order Reduction can be directly applied without needing to adapt them to this specific problem

Session 1pSC

Speech Communication: Cross-Linguistic Studies of Speech Sound Learning
of the Languages of Hong Kong (Poster Session)

Estella Ma, Cochair
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Benjamin Munson, Cochair
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Contributed Papers

All posters will be on display from 2:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:40 p.m. to 4:00 p.m. and contributors of even-numbered papers will be at their posters from 4:00 p.m. to 5:20 p.m.

1pSC1. Adaptation of English word-final stops into Korean: Effects of English exposure. Harim Kwon (University of Michigan, Department of Linguistics, 611 Tappan St. 440 Lorch Hall, Ann Arbor, MI 48109, harim@umich.edu)

In Korean, word-final stops are never released. When borrowed into Korean, English stop-final words are sometimes, but not always, adapted by epenthesis of /i/ after the stop. Epenthesis is most frequent for words with coronal stops, that is, words whose stops are arguably most often released [Kang, *Phonology* 20, 219-273 (2003)]. This study investigates Korean listeners' attention to this sub-phonemic cue in borrowing. Korean listeners are predicted to attend to release cues, but their attention should decrease with their L2 exposure to English. Korean monolinguals, late-bilinguals, and early-bilinguals were tested on English non-word stop-final stimuli with and without releases. Their task was to add a suffix to the novel form; of interest was whether epenthetic /i/ was inserted during the task. Overall, released stops were twice as likely as unreleased to trigger epenthesis; however, monolinguals were nearly twice as likely as early bilinguals to insert /i/ after a released stop. This result accords with the "phonological" adaptation of proficient bilinguals [LaCharité and Paradis, *Ling. Inquiry* 36, 223-258 (2005)]. More proficient bilinguals "know" that English stop release is not contrastive and ignore stop release when adapting English words.

1pSC2. Reciprocal perception of Chinese and Korean affricates and fricatives. Bin Li, Sunyoung Oh, Jing Shao, and Lan Shuai (City University of Hong Kong, binli2@cityu.edu.hk)

The distinction between lax and tense sounds, as well as aspiration, is employed in classifying Korean affricates and fricatives. Affricates and fricative in Chinese are mainly distinguished in places and aspiration. Little attention has been given on perception of these two types of consonants. This study investigates reciprocal perception of Chinese and Korean affricates and fricatives, by examining how native speakers of the two languages identify each other's consonants at syllable-initial positions. Predictions are made based on how non-native sounds may be assimilated into the native system. First, tense and lax Korean affricates and alveolar fricative may pose difficulty in consonant labeling for Chinese listeners. Korean alveolar fricatives may be perceived as the aspirated alveolar fricative by Chinese listeners. Second, Chinese retroflex and palate-alveolar affricates may be identified as belonging to same categories by Korean speakers. Third, Chinese alveolar and retroflex fricatives may be perceived as same categories when preceding vowel /i/. Predictions are confirmed in most cases. It is also found that vowel contexts play a role in consonant labeling. Assessment of current speech perception models is also discussed using our findings.

1pSC3. Effects of perceptual training on the ability of elderly adults of Japanese speakers to identify American English /r/-/l/ phonetic contrasts. Rieko Kubo (Japan Advanced Institute of Science and Technology, 923-1292, rkubo@jaist.ac.jp), Reiko Akahane-Yamada (ATR Learning Technology Corporation, 619-0288), and Masato Akagi (Japan Advanced Institute of Science and Technology, 923-1292)

Previous research has demonstrated that auditory perceptual training on young adults improves their ability to identify phonological contrast in foreign languages. However, most research has focused on college-age adults, and little research has considered age-related changes among adults. In this study, training to identify American-English /r/-/l/ was conducted with Japanese speakers in their 30s, 40s, 50s, and 60s. The pretest-post-test design was used for comparing the results with those of young adults (Lively, 1994). As a result of training, although the identification performances improved in every age group, there was an age-related gradual decline in the improvements. Analyses concerning phonetic environments revealed that older adults had more difficulty learning targets in initial consonant clusters than did the younger ones, while had comparable improvements in final position. Komaki et al. (1999) suggested that as a result of listeners' language-specific perceptual strategies, Japanese speakers identify targets least accurately in initial consonant clusters and most accurately in final position. This trend in identification is seen in older adults' learning. These results suggest that the ability to learn new phonological categories is preserved in normal aging from the 30s to 60s and that older adults are more dependent on their first language in learning.

1pSC4. Distributions of [r]-deletion and [w]-substitution in English C[r] clusters by Cantonese speakers. Yizhou Lan and Sunyoung Oh (City University of Hong Kong, ylylan2@student.cityu.edu.hk)

Literature shows that Cantonese speakers often delete [r] in C[r] clusters in English (e.g. [pɪnt] for "print"). However, based on our observation, [w] substitution of [r] is also apparent during speech. Experiment was conducted to see the distribution of deletion and substitution and investigated the effects of the place of articulation and syllable structure on the distribution of C[r] production. Ten non-English-major university students participated in the study. Stimuli were randomized and repeated 10 times in a passage, contrasting consonants between open and closed syllables (C[r]V vs. C1[r]VC1) in three vowel conditions (/i, a, u/). Two-way ANOVA was used for statistics. Preliminary results show that overall, substitution is more dominant than deletion for the clusters, particularly with velars. Deletion, though much less than substitution, is only observed with bilabials whereas alveolar clusters are mostly pronounced correctly. Also, substitution and deletion tend to occur more in closed syllables than in open syllables. Findings

suggest that such distributions are based on the speakers' articulatory strategy of gestural economy.

1pSC5. Cross-dialect and cross-language differences in perception of vowels: A multidimensional scaling study. Jing Yang, Robert Allen Fox, and Ewa Jacewicz (Department of Speech and Hearing Science, The Ohio State University, 1070 Carmack Rd., Columbus OH, 43210-1002, yang.1198@osu.edu)

This study examines the perceptual responses to vowels in two regional varieties of American English typical of Central Ohio (OH) and Western North Carolina (NC). Listeners are monolingual local speakers of each dialect and Mandarin-English bilinguals in Columbus OH. The questions are (1) how English listeners perceive acoustic variations in vowel quality differences not found in their own dialect and (2) whether bilingual listeners are sensitive to such differences not found in their native language. In a multidimensional scaling study, we investigate whether listeners make these perceptual decisions on the basis of acoustic properties or categorical properties of these vowels. Each listener was presented with two sets of 13 vowels /i ɪ e ε æ u ʊ o ɔ̃ ɔɪ a/ in a /hVd/ syllable. One set was produced by an OH male speaker and the other by a NC male speaker. Listeners rated all possible vowel pairs from one set only on a nine-point similarity/dissimilarity scale (there were two subsets of listeners in each group). The resulting dissimilarity matrices were analyzed using INDSCAL. Results will be discussed in terms of differences in perceptual dimensions (vowel coordinates and subject weights) as a function of dialect and L2 background.

1pSC6. Vowel production before 8 years of age: A longitudinal formant analysis. Li-mei Chen, Ya-Chiang Lin, and Tzu-Wen Kuo (National Cheng Kung University, 1 University Rd, Tainan City, Taiwan, leemay@mail.ncku.edu.tw)

Vowel production in two Mandarin-learning children was recorded from birth to 8 years old. The present study is the report of the 8th year. Major findings are: 1) Starting from 5 years of age, vowels with nasal endings show similar frequency as diphthongs. Girl subject used vowels with nasal endings more frequently than boy subject; 2) Decrease of F1 F2 values is continuously observed in boy subject. As to girl subject, up to 8 years of age, no obvious change in formant values was found; 3) F1 values are more stable than F2 values, especially in [i, u]. They appeared to acquire jaw movement sooner than tongue movement. Although a general trend of reduction in variability can be found from 4 years on, there seems to be more fluctuation of F2 values at 7-8 years of age in both subjects; 4) The trend of reduction of vowel areas was found to start at around 4 years old for boy and at 6 years old for girl; 5) No obvious decline in fundamental frequencies was found at 7-8 years of age in both subjects. This investigation was supported through funds from National Science Council in Taiwan (NSC 99-2410-H-006 -102-MY2).

1pSC7. Identification of synthesized Mandarin tones by Northern Vietnamese speakers. Bin Li, Lan Shuai (City Univ. of Hong Kong, Tat Chee Ave., Kowloon Tong, Hong Kong, binli2@cityu.edu.hk), and Thi Thu Ha Pham (Univ. of Social Sciences and Humanities, Vietnam National Univ., Hanoi, Vietnam)

Tones in Northern Vietnamese are distinguished by pitch variations and phonation types, but falling pitch slopes are not considered as a major cue that its speakers rely on in tonal contrast. This may contribute to difficulties that native speakers of Northern Vietnamese are faced with when learning Mandarin tones, especially the distinction between the level (T1) and the falling (T4) tones. To examine how Northern Vietnamese speakers perceive the distinction between T1 and T4 in Mandarin and how well they adapt phonetic cues underlying the non-native distinction, an ABX identification experiment is carried out on perception of synthesized pitches with two inter-stimuli-intervals (ISI) at 500ms and 1500ms. Results suggest a combined effect of pitch slopes and pitch heights, the former of which claims more robust influence on tone perception. However, a reversed pattern is found when the ISI equals 500ms, where pitch heights exert a stronger effect on tone identification than pitch slopes.

1pSC8. Effects of errorless learning on the acquisition of velopharyngeal movement control. Andus Wing-Kuen Wong (Division of Speech and Hearing Sciences, and Institute of Human Performance, University of Hong Kong, draw@hku.hk), Tara Whitehill, Estella Ma (Division of Speech and Hearing Sciences, University of Hong Kong), and Rich Masters (Institute of Human Performance, University of Hong Kong)

The implicit motor learning literature suggests a benefit for learning if errors are minimized during practice. This study investigated whether the same principle holds for learning velopharyngeal movement control. Normal speaking participants learned to produce hypernasal speech in either an errorless learning condition (in which the possibility for errors was limited) or an errorful learning condition (in which the possibility for errors was not limited). Nasality level of the participants' speech was measured by nasometer and reflected by nasalance scores (in %). Errorless learners practiced producing hypernasal speech with a threshold nasalance score of 10% at the beginning, which gradually increased to a threshold of 50% at the end. The same set of threshold targets were presented to errorful learners but in a reversed order. Errors were defined by the proportion of speech with a nasalance score below the threshold. The results showed that, relative to errorful learners, errorless learners displayed fewer errors (50.7% vs. 17.7%) and a higher mean nasalance score (31.3% vs. 46.7%) during the acquisition phase. Furthermore, errorless learners outperformed errorful learners in both retention and novel transfer tests. Acknowledgment: Supported by The University of Hong Kong Strategic Research Theme for Sciences of Learning

1pSC9. Extrinsic context is crucial for talker normalization in Cantonese tone perception. Caicai Zhang, Gang Peng, and William S-Y. Wang (Language Engineering Laboratory, The Chinese University of Hong Kong, Hong Kong, yzcelia@gmail.com)

Previous studies showed that recognizing a phonetic category produced by different talkers relies on both intrinsic (target-internal) and extrinsic (contextual) cues. Extrinsic cues influence perception when intrinsic cues allow more than one phonetic interpretation. A recent study in this laboratory found that the configuration of tone systems (Cantonese and Mandarin) affects the degree of ambiguity of tones associated with intrinsic cues. In Cantonese which has three level tones, an isolated level pitch can be mapped to any of these three categories. Cantonese but not Mandarin listeners were found to confuse tones in a way biased by relative pitch height of different talkers. The present study tested Cantonese listeners on stimuli from four talkers with different pitch ranges (Female High, Female Low, Male High, and Male Low). Syllable /i/ carrying different F0 contours was embedded in a meaningful sentence with cues of a talker's F0 range. This study found enhanced identification accuracy with contextual cues over performance in isolation (92.25% vs. 51.75%), suggesting that extrinsic context facilitates talker normalization. This finding implies that extrinsic cues are especially useful for a language with intrinsically ambiguous phonetic categories. [Research supported by GRF 455911, NSFC 11074267, NSFC 61135003, and a 973 grant 2012CB720700.]

1pSC10. Effects of musical experience on learning lexical tone categories. Tian Zhao and Patricia Kuhl (University of Washington, Institute for Learning & Brain Sciences, MS 357988, Seattle, WA 98195, zhaotc@uw.edu)

The relationship between music and speech processing is of great interest. Lexical tones, contrastive pitch-modulation patterns at the word level, are an ideal tool to explore these relations. Previous studies suggest that musicians exhibit an advantage in discriminating lexical tones. The current study aims to explore whether having extensive musical training is associated with the ability to form robust lexical tone categories given highly variable natural speech tokens. A continuum of pitch contours was created with Mandarin Tone 2 and Tone 3 as the endpoints (see Zhao, Wright, & Kuhl ASA abstract). First, 20 monolingual English musicians and 20 monolingual English non-musicians completed identification and discrimination tasks that established individuals' perceptual boundaries on the continuum. Then, half of the musicians and half of the non-musicians were randomly assigned to an 8-session perceptual training procedure. Lastly, all subjects completed identification and discrimination tasks both with old and new stimuli to

examine changes in perceptual boundaries and generalization. Results will be considered in terms of theories relating speech and music processing. [Research supported by NIH and NSF.]

1pSC11. Tone duration and tonal slope of a cochlear implant child in comparison with a normal hearing child: A longitudinal study of a pair of twins. Li-mei Chen, Ya-Wen Chen, and Yi-Ru Chou (National Cheng Kung University, 1 University Rd, Tainan City, Taiwan, leemay@mail.ncku.edu.tw)

The aim of this study is to observe the tonal acquisition in a pair of fraternal twins from 1 to 3 years old, making a comparison between a child with cochlear implant (Child A) and a child with normal hearing (Child B). Spontaneous data were transcribed and later tone duration and tonal slope were measured by Praat. Four main tones in Mandarin were analyzed: high-level, high-rising, high-falling, and low-falling tones. Major findings are: 1) Tone duration of Child B is longer than that of Child A by 1.2-1.7 times. The average of tone duration in Child A is 250ms-260ms; 2) In Child A, high-level tones show the shortest duration among the four main tones, and high-rising tones are the longest; 3) Tonal slope of Child A is flatter than in Child B. The absolute value of tonal slope is higher in Child B; 4) Children control high-level tone the best, followed by high-rising, then high-falling, and low-falling the last.

1pSC12. Visual displays of the pitch pattern for the CAI self-teaching system to discriminate Chinese tones. Qi Sun, Song Liu, Kazuko Sunaoka, and Shizuo Hiki (Language and Speech Science Laboratory, Waseda University, 1-104, Totsuka-machi, Shinjuku-ku, Tokyo 169-8050, Japan, sunqi@aoni.waseda.jp)

A computer-assisted instruction (CAI) system for self-teaching to discriminate Chinese tones is available for public through the internet (<http://chinesestone.org>) in Japanese, English, and Chinese versions. The design and construction of this system has been reported previously (Hiki et al., *J. Acoust. Soc. Am.*, 120 (5, Pt. 2), 2006, 3168). This system utilizes the displays of the pitch pattern as visual cues in training. The following new functions of the visual displays have been added to the system recently: 1) Essential pitch patterns of 15 bisyllabic words, with every combination of the four tones in Standard Chinese, were drawn on the six whole tone musical scale. By displaying visually the corresponding essential pitch pattern along with the measured pitch pattern, it became easier for the beginners to perceive aurally the tonal characters, which underlay the measured pitch patterns; 2) The bisyllabic word lists comprising only voiced consonants were edited, and the measured pitch patterns not interrupted by unvoiced consonants were displayed visually. These speech samples were useful in the early stages of tone discrimination learning. It was also ascertained that the tone discrimination was stimulated by paying attention to musical pitch perception.

1pSC13. The perceptual sensitivity to the prosodic cues in disambiguation of the ambiguous and the biased ambiguous sentences. Sun mi Kang (Korea Univ., dearsunny@korea.ac.kr), Mi Hye Kim, and Kee ho Kim

This study explores the perceptual sensitivity to the prosodic cues in disambiguating the structurally ambiguous sentences by English native speakers and Korean learners of English. Also, this study aims to investigate that even the biased ambiguous interpretations due to their semantic factors were also affected by the prosodic cues. The perception experiment was

conducted by the cognitive experimental tool, E-prime. In the perception test, one of the meanings of the sentences were presented on the screen, and then the acoustic stimuli were provided to the participants. The acoustic stimuli were given in two phases - the normal stimuli and the reinforced stimuli. The subjects were asked to judge the correspondence between the semantic and acoustic stimulus. The recognition rate increased, while the response time was shortened, when the reinforced stimuli were given. To be specific, English natives sensitively respond to the phrasal accent and phrase final lengthening, while Korean learners are sensitive to the substantial pause. With regarding to the biased ambiguous sentences, Korean learners do not rely on their baseline preference and easily shifted their resolution according to the prosodic factors they heard. While native speakers much rely on their internally fixed interpretation beside of the meticulous prosodic cues.

1pSC14. Phonetic characteristics of school students in Chinese dialect regions. Yali Liu (Communication Acoustics Laboratory, Communication University of China, Beijing 100024, P.R. China, pear-1984@163.com), and Zihou Meng (Communication Acoustics Laboratory, Communication University of China, Beijing 100024, P.R. China)

To study the factors which can affect Mandarin-learning of school students in dialect regions, parents, teachers and school students are investigated by questionnaire respectively. Based on the questionnaire, two attributes are obtained through factor analysis, including Mandarin-learning environment and the correlation of Mandarin and dialect. The speech database in Mandarin dialect is recorded, in which there are 455 speakers in total, including 235 male speakers and 220 female speakers (aged from 8 to 18 years old). The database falls into three groups by age. The first group is from 7 to 11 years old; the second is from 12 to 13 years old; the third is from 14 to 16 years old. The acoustic characteristics of initials, finals, F0 and tones are investigated under the three groups. For initials, peaks of high frequencies are extracted. For finals, formant pattern charts are given. The development of F0 and patterns of contour tones for a syllable are summarized. Based on the result of the factor analysis and acoustic characteristics, the patterns of phonetic errors of different age during Mandarin-learning are proposed. The phonetic analysis may provide some references for assessment and improvement of Mandarin-learning for teenagers in dialect regions.

1pSC15. Cantonese Pronunciation among Hong Kong speakers and American Born Chinese speakers. Sandy Cho, Laura Koenig, and Lu Feng Shi (LIU Brooklyn- 1 University Plaza Brooklyn, NY 11201, sandyycho@yahoo.com)

This study contrasts the production of Cantonese words between native (Hong Kong) immigrant speakers of the language and native first or second generation American-born Cantonese speakers. The word lists were constructed to sample across the tone space and vowel space. Multiple recordings were recorded at different intervals in order to assess variability in sound production. Of particular interest was whether the American-born speakers showed greater variability than the Hong Kong speakers. Vowel formant frequencies and tonal patterns, measured by fundamental frequency contours, were compared between the Hong Kong speakers and the American born speakers. In addition, we evaluated the presence of "lazy tones" in the two groups. This data adds to the sparse literature on Cantonese speakers in America.

Session 1pUWa

Underwater Acoustics and Signal Processing in Acoustics: Advances in Underwater Acoustic Communication and Networking II

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James Preisig, Cochair
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Invited Papers

2:00

1pUWa1. On turbo equalization for mobile multi-input multi-output underwater acoustic communications. Kexin Zhao, Jun Ling, and Jian Li (University of Florida, NEB 465, PO Box 116130, University of Florida, Gainesville, FL 32611, United States of America, kexinzhao@ufl.edu)

This paper focuses on mobile multi-input multi-output (MIMO) underwater acoustic communications (UAC) over double-selective channels subject to both inter-symbol interference and Doppler scaling effects. Temporal resampling is implemented to effectively convert the Doppler scaling effects to Doppler frequency shifts. A variation of the recently proposed generalization of the sparse learning via iterative minimization (GoSLIM) algorithm, referred to as GoSLIM-V, is employed to estimate the frequency modulated acoustic channels. GoSLIM-V is user parameter free and is easy to use in practical applications. This paper also considers Turbo equalization for retrieving the transmitted signal. In particular, this paper reviews the linear minimum mean-squared error (LMMSE) based soft-input soft-output equalizer involved in the Turbo equalization scheme, and adopts a fast implementation of the equalizer that achieves negligible detection performance degradation compared to its direct implementation counterpart. The effectiveness of the considered MIMO UAC scheme is demonstrated using both simulated data and measurements recently acquired during the MACE10 in-water experiment. Acknowledgments: This work was supported in part by the Office of Naval Research (ONR) under Grant No. N00014-10-1-0054. We gratefully acknowledge WHOI for the fruitful collaborations with us to conduct the in-water experimentations and for sharing data with us.

2:20

1pUWa2. Low complexity estimation and equalization of doubly spread underwater acoustic channels. Wen-Jun Zeng and Wen Xu (Department of Information Science and Electronic Engineering, Zhejiang University, Hangzhou, 310027, China, cengwj06@mails.tsinghua.edu.cn)

Underwater acoustic channels are characterized by limited bandwidth, long multipath delay spread, and severe time variation, which make reliable and high-rate communication challenging. Channel-estimate-based equalization is a key technique for compensating for distortions introduced by the channel. In this paper, low complexity algorithms for estimation and equalization of doubly spread underwater acoustic channels are presented. By exploiting the sparsity in the delay-Doppler domain, a fast projected gradient method (FPGM) is developed for estimating the delay-Doppler spread function of a time-varying channel. The FPGM formulates sparse channel estimation as a complex-valued convex optimization using an ℓ_1 -norm constraint. Unlike the conventional methods that split the complex variables into their real and imaginary parts, the FPGM directly handles the complex variables as a whole. A unified framework, which includes the time-reversal, linear MMSE, and decision feedback equalizer, is also proposed for fast equalization of doubly spread channels. By exploiting the special block Toeplitz-like structure of the coefficient matrix, the computational complexity of channel estimation and equalization is on the order of $L \log N$, where L is the dimension of the Doppler shift and N is the signal length. [Work supported by Chinese 863 high-tech program under Grant 2009AA093601]

2:40

1pUWa3. Adaptive multichannel decision feedback equalization using subarray processing. James Preisig (Woods Hole Oceanographic Inst., Woods Hole, MA 02543, jpreisig@whoi.edu)

The adaptive multichannel Decision Feedback Equalizer (DFE) has been shown to be an effective algorithm for enabling reliable high rate acoustic communications in complex and time-varying underwater environments. The choice of the number of channels used in an equalizer presents a performance trade-off. The minimal achievable error that can be realized by the equalizer decreases as the number of channels increases. However, an increase in the number of channels increases the number of filter weights that need to be adapted. Thus, the computational complexity of least squares and Kalman type adaptation algorithms is proportional to the number of

channels squared. In addition, the averaging interval required by an adaptation algorithm in order to achieve good performance grows linearly with the number of parameters. Thus, an increase in the number of M-DFE channels can reduce the rate of channel fluctuation that can successfully be tracked by the M-DFE. The partitioning of an array into sub-arrays which are each independently equalized before combining their outputs can both improve performance and reduce complexity in processing real-world signals. The choice of the size and sensor locations for the subarrays is analyzed. The resulting adaptive subarray multichannel DFE algorithm is compared to other multichannel equalization algorithms.

Contributed Papers

3:00

1pUW4. Sequential analysis for underwater communications. Andrey Morozov (Teledyne Benthos TWR, 82 Technology Park Drive, East Falmouth, MA 02536, amorozov@teledyne.com), and Dale Green (Teledyne Benthos, 49 Edgerton Drive, North Falmouth, MA 02556)

The state-of-the-art in high-rate, single-carrier wideband signaling for acoustic communications is represented by the decision feedback equalizer (DFE). Though often effective, its performance is far from the optimal obtained from soft decision maximum a posteriori probability (MAP) and maximum-likelihood sequence detectors (MLSD). While these algorithms have optimal performance, their complexity increases exponentially with the duration of the channel impulse response. In practice, such methods are only used for multicarrier modulation, after mode filtering or other form of channel shortening, time-spatial pre-processing equalization. The optimal joint channel estimation and data decoding algorithm is derived and analyzed. The combination of pre-processing, channel response shortening equalization, and joint channel and data recovery have shown excellent performance in shallow water acoustic communications experiments. A sequential estimation alternative to MLSD-based decoding is “almost” as effective in a probability sense, given a modest increase in signal-to-noise ratio (SNR). That approach combines very high performance with a small computational burden relative to the MLSD approach. The practical result of the paper is the investigation of the replacement of the DFE with a sequential implementation (FANO) of a likelihood sequence estimator. Comparative performance of the two using at-sea experiments in very shallow water is provided.

3:20

1pUW5. Adaptive equalization, tracking, and decoding for high-rate underwater acoustic communications. Andrew Singer (University of Illinois, Urbana-Champaign, 110 CSL, 1308 W. Main Street, Urbana, IL 61801, acsinger@illinois.edu)

The interaction between equalization and decoding in the form of turbo-equalization has been shown to enable dramatic bit error rate (BER)

improvements in high-rate underwater acoustic communication links. These improvements are particularly pronounced at lower SNR, higher data rate and in highly dynamic environments, where forward error correction can be leveraged to enable the receiver to maintain tracking and data recovery through deep fades or bursts of noise in the received signal. Platform mobility exacerbates these challenges, since the resulting broadband Doppler is manifested as a dynamic dilation and contraction of the modulated waveform, necessitating dynamic resampling at the receiver to preserve symbol timing. In this talk, we present results from recent at-sea experiments in which time-varying Doppler compensation has been integrated into an adaptive turbo equalization receiver for both single-channel and multi-channel receiver systems.

3:40

1pUW6. Experimental studies of support vector machine based blind equalization for shallow water channels. Wu Fei Yun, Zhou Yue Hai, and Tong Feng (Xiamen University, wfyfly@126.com)

Due to extended multi-path spread and rapidly changing characteristics, shallow water acoustic channels pose excessive difficulties to the design of high performance underwater communication systems. While classic blind equalization algorithms such as the constant modulus algorithm (CMA) offer potential solutions to tackle the ISI (inter symbol interference) in underwater scenarios without training sequence, slow convergence rate as well as low noise tolerance limits their practical applications. In this paper, blind equalizer based on support vector machine (SVM) is adopted, which makes use of the excellent generalization ability of SVM to accelerate the convergence, and addresses the carrier phase error with embedded phase lock loop (PLL). Experimental SVM blind equalization results conducted in physical shallow water channels show significant performance improvements, demonstrating the effectiveness of the proposed method.

4:00–4:20 Break

Invited Paper

4:20

1pUW7. Focusing, doppler shifts and bubble screening: Parameterizing surface reverberation in different wind regimes. Grant Deane (Scripps Institution of Oceanography, UCSD, La Jolla, CA 92093-0238, gdeane@ucsd.edu), and James Preisig (Woods Hole Oceanographic Institution, Dept. of Applied Ocean Physics and Eng., Woods Hole, MA 02543)

Advances in underwater acoustic communications systems can be based upon physical insight into the relationship between the acoustic channel and controlling environmental variables, such as wind and waves. In the mid-frequency band (3 kHz - 20 kHz) and at relatively short ranges (order 10 water depths) in reverberant channels, gravity waves focus sound energy incident on the sea surface, creating intensifications, Doppler shifts and phase shifts in the reflected field. Breaking waves entrain bubbles into a sub-surface layer that attenuates and scatters sound, which tends to screen reflections from the surface and lessen the impact of these effects. Observations of the time-varying arrival intensity structure from an experiment conducted in the Martha's Vineyard Coastal Observatory will be presented along with model calculations made using the Kirchhoff approximation. Model calculations of wave-induced Doppler shifts and examples of bubble screening will be discussed. [Work supported by the ONR Ocean Acoustics Program].

4:40

1pUWa8. Numerical simulation and experiment of the shallow water communication channel with time varying multipath effect using time reversal mirror. Yi-Wei Lin, I Putu Andhi Indira Kusuma, and Gee-Pinn Too (No. 1, University Road, Tainan City 701, Taiwan (R.O.C.), ls5028@gmail.com)

The underwater communication channels limitations are mainly due to multipath effect and ambient noise. Multipath effect is a serious problem when underwater communication in the shallow water is considered, because it induced severe inter-symbol interference (ISI). Time reversal mirror (TRM) has been widely used in the underwater communication to overcome

the multipath effect by evaluating impulse response function (IRF); and further, it can reduce bit error rate (BER). In this paper, underwater communication channel is explored in 175 m x 8 m x 4 m Towing Tank. Both numerical simulation and experimental are conducted to verify the effect of time varying multipath. Image method and statistic analysis are used to simulate the time varying multipath effect due to wave propagation in the shallow water. The BER caused by time varying multipath effect is evaluated by adding arrival time lag variation and arrival signal amplitude variation in impulse response function. The results show that time reversal process is effective to reduce BER and overcome the time varying multipath effect.

Invited Papers

5:00

1pUWa9. Coherent communications in snapping-shrimp dominated ambient noise environments. Mandar Chitre, Ahmed Mahmood, and Marc Armand (National University of Singapore, 18 Kent Ridge Rd, Singapore 119227, mandar@nus.edu.sg)

The additive white Gaussian noise (AWGN) model is commonly used in the development of communication systems, and adequately models many noisy environments. However the impulsive noise from snapping shrimp is poorly approximated by this model. The mismatch in model has an adverse impact on the performance of conventional communication systems operating in warm shallow waters. The AWGN model may be replaced by the more general additive white symmetric α -stable noise (AWS α SN) model, which better approximates the heavy tailed noise due to snapping shrimp. When converted to the complex baseband representation, the resulting noise for the AWS α SN case is radically different from its Gaussian counterpart. In this talk some properties of baseband noise for the general AWS α SN case are investigated. These properties can be used to guide design decisions for coherent communication systems operating in warm shallow waters. The baseband noise is generally not isotropic and furthermore the real and imaginary components may be dependent. By varying certain physical parameters different non-isotropic distributions may be attained. The resulting properties can be exploited to design communication systems that are able to provide robust performance in the presence of snapping shrimp noise.

5:20

1pUWa10. Recent experiment results of long-range time-reversal communication in deep ocean. Takuya Shimura, Hiroshi Ochi, and Yoshitaka Watanabe (JAMSTEC, 2-15 Natsushima-cho, Yokosuka-city, 237-0061 Japan, shimurat@jamstec.go.jp)

In the Japan Agency for Marine-Earth Science and Technology (JAMSTEC), a project to develop new autonomous underwater vehicle (AUV) is being planned, which will have the capability to cruise long distances over several hundred kilometers. Achieving acoustic communication with such a long-range AUV, even at a low data transmission rate, will be important. Time reversal is an attractive solution for such a long-range communication, by converging multipath signals and decreasing intersymbol interference (ISI). Thus, we have researched on time-reversal communication in the deep ocean, have proposed a method of combining time reversal and adaptive equalization, and have executed various at-sea experiments in the deep ocean. In the first experiments, both active and passive time-reversal communication were performed at the range of 10 km and it was shown that time reversal could enable communication under many multipath interferences. The subsequent experiments were carried out in various ranges for passive time-reversal communication. In our latest trial, communication at the range up to 1,000 km was demonstrated at the data rate of 100 bps at the frequency of 500 Hz. In this paper, the results of these experiments are described.

Contributed Papers

5:40

1pUWa11. Time reversal based channel tracking for underwater acoustic communications. Menglu Xia and Wen Xu (Department of Information Science and Electronic Engineering, Zhejiang University, Hangzhou, 310027, China, luluxml@gmail.com)

Time reversal processing (TRP) has been proved to achieve temporal compressing when the waveguide environment is invariant. TRP has been exploited in underwater acoustic communications as it can, without any knowledge of the channel, compensate severe inter-symbol interferences (ISI) that are caused by complex multi-path propagation. However, environmental variations occur almost all the time when conducting acoustic communications in the real ocean, and indeed become one of the main factors determining the communication performance. Since the channel is time variant, the ISI can not be removed completely through time reversal. In the present paper, a channel tracking method is developed by using this leftover

ISI after time reversal processing in a slowly time-variant environment. Change of the channel response structure is estimated in terms of time-delay shift and attenuation differences. Simulations of the method applied to synthetic data and field experimental results are both provided to demonstrate the method's feasibility. [Work supported by Chinese 863 high-tech program under Grant 2009AA093601]

6:00

1pUWa12. Study on underwater acoustic voice communication system for divers. Lihua Lei and Feng Xu (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China 100190, lhlei@mail.ioa.ac.cn)

This paper describes the design and sea trial of an underwater acoustic speech transmission system for the continuously increasing need for diver communications. The input speech signal is compressed down to 2.4k bit/s

using a MELP (mixed excited linear prediction) coder. The bit rate is 6k bit/s after channel coding and frame synchronization. A QPSK modulation with differential encoding was chosen to transmit the useful signal. To overcome the phase fluctuation and multi-path effects caused underwater acoustic channel we employed a scheme where phase synchronization and fractional spaced Decision Feedback Equalization (DFE) were jointly optimized by RLS algorithm in the receiver. Two kinds of error correcting schemes were used including convolutional codes (CC) and Reed Solomon (RS) block codes. The whole system has been conducted successfully in very shallow water environments and the short range horizontal acoustic voice communication link performances are evaluated.

6:20

1pUWa13. Doppler experiment for shallow underwater acoustic communication using QPSK and turbo coder. Teiichiro Ikeda, Kunio Hashiba, Shinta Takano (Central Research Laboratory, Hitachi Ltd., 185-8601, Japan, teiichiro.ikeda.hv@hitachi.com), Ryusuke Imai, and Mitsuhiko Nanri (Defense Systems Company, Hitachi Ltd., 101-8608, Japan)

A continuous data link under shallow and towing conditions is essential for UUV operation. In shallow water, inter-symbol interference (ISI) due to multipath fading strongly distorts the carrier signal. In addition, the phase of the signals shifts considerably due to the Doppler shift at a high relative speed (~5-kt) between the mother ship and the UUV. The communication performance degrades significantly compared to static communication. Actually, a 1-kt relative speed between the transmitter and receiver causes a

phase shift of more than $2M\pi$ within 1000 symbols of transported data. In this study, we investigated QPSK acoustic telecommunication systems that incorporate a Turbo coder and digital phase lock loop (DPLL) for the Doppler shift compensation. A towing experiment was conducted in shallow water, and the performance of the system was evaluated. When the receiver was 3 m deep, at speeds of 1.49 and 2.02 kt, we see a considerable number of errors. When the receiver depth was 15 m from the surface, all of the transmitted data were completely reproduced after applying our acoustic communication algorithm.

6:40

1pUWa14. Design and implementation of underwater video transmission system. Chen Weilin and Ren Hao (Harbin Engineering University, 150001, willing1111@126.com)

At first, the paper introduces the basic principles of OFDM system. And then analyzes the structural characteristics of the chip DM642. Taking the design requirements of the system into account, the implementation method in which the main design of underwater image transmission system based on DM642 is accomplished, and the detail designs are given. At the same time, the paper mainly analyzes the address generation process of the frame memory. And then some of the key technologies of PCB designing are introduced. In the software section, the implementations of OFDM technology and algorithm of video compression on the DM642 are highlighted for a specific introduction. Keywords-OFDM; video transmission; H.264; DSP

MONDAY AFTERNOON, 14 MAY 2012

S426 + S427, 2:00 P.M. TO 5:40 P.M.

Session 1pUWb

Underwater Acoustics, Acoustical Oceanography, and Physical Acoustics: Sediment Acoustics of Continental Shelves II

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Contributed Papers

2:00

1pUWb1. The influence of the uncertainty of water depth on the inversion of bottom sound speed based on normal mode group velocities in Pekeris waveguides. Mei Sun (Department of Physics and Electronics, Taishan University, Taian 271021, China, tsusunmei@163.com), and Fenghua Li (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China)

The normal mode group velocities are widely used in geo-acoustic inversion, and the uncertainty of water depth affects the inversion results. In order to reduce this effect, the law of normal modes' group velocities in Pekeris waveguides versus the water depth and the bottom sound speed is studied. Theoretical and simulative results show that if the frequency is around the Airy frequency of a normal mode, the group velocity of the normal mode is not sensitive to the water depth but sensitive to the bottom

sound speed. The group velocities of normal modes are applied to the inversion of the bottom sound speed. Experimental results indicate that the influence of the uncertainty of water depth on the inversion results is significantly reduced if the normal modes whose associated Airy frequencies are near the working frequency.

2:20

1pUWb2. Estimation of shear-velocity profiles using shear source data in marine environment. Hefeng Dong, Thanh-Duong Nguyen (Norwegian University of Science and Technology, NO-7491 Trondheim, Norway, hefeng@iet.ntnu.no), and Kenneth Duffaut (Statoil ASA, Arkitekt Ebbells vei 10, NO-7005 Trondheim, Norway)

This paper estimates seabed shear-velocity profiles and their uncertainties using interface-wave dispersion curves extracted from data generated

by a shear source. The data were collected by 4C ocean bottom cable in a testing experiment of a shear source in the North Sea. The shear source generated a seismic signature over a frequency range between 2 and 60 Hz and was polarized in both in-line and cross-line orientation. Low-frequency Scholte- and Love-wave were recorded. Dispersion curves of the Scholte- and Love-wave for the fundamental mode and higher-order modes are extracted by different time-frequency analysis method. Both vertical (SV) and horizontal (SH) polarized shear-velocity profiles are estimated by Scholte- and Love-wave dispersion curves, respectively. Bayesian approach is used for the inversion. Differential evolution (DE) global search algorithm is applied to estimate the most-probable shear-velocity models. Marginal posterior probability profiles are computed by Metropolis-Hastings sampling. The estimated SV- and SH-velocity profiles are compared and the results provide possibility of studying seabed anisotropy. We thank Statoil ASA for permission to publish the data. The work is partly supported by NFR under Grant No. 186923/I30.

2:40

1pUWb3. A scaled mapping approach for treating sloping interfaces in parabolic equation solutions. Jon M. Collis and Daniel Moran (Colorado School of Mines, Golden, CO, 80401, jcollis@mines.edu)

An area of research in ocean acoustics is on solution techniques for environments with sloping seafloors. In these range-dependent environments the oceanic waveguide varies in the direction of propagation and analytic solutions do not exist. In order to enforce continuity conditions at the seafloor the environment is approximated as a series of range-independent regions. An alternative to this approach, the mapping solution [M. D. Collins et al., JASA 107] applies a change of variables that results in a vertical translation of the environment and makes it easier to enforce ocean bottom interface conditions. An alternative mapping approach that uses a characteristic depth length scale is considered here. In contrast to the earlier mapping solution, this approach uses a change of variables in which the depth variable is scaled relative to the depth of the bottom interface. Along with the simplification of bottom interface conditions in both approaches, extra terms arising from the mapping are neglected, and hence a correction is applied in both to the calculated solution. The new approach is implemented in a parabolic equation solution and is benchmarked against existing solutions. The accuracy of this new approach is established initially for problems involving fluid sediments.

3:00

1pUWb4. The horizontal correlation of long range bottom reverberation in shallow water with inclined sea floor. Shi-e Yang, Bo Gao, and Sheng-chun Piao (Harbin Engineering University, ShuiSheng Building Room 1304, 145 Nantong Street, Harbin 150001, China, yangshie@hrbeu.edu.cn)

The performance of active sonar system is seriously influenced by bottom reverberation in shallow water waveguide. It is important to understand the horizontal correlation of bottom reverberation for active towed-array processing techniques in shallow sea. However, little work had been done for the research on horizontal correlation of distant bottom reverberation. In this paper, a coupled mode reverberation model was applied for the horizontal correlation, and it was investigated as a function of receiving position, time and frequency. Calculations show that transverse correlation is greater than the longitudinal correlation in horizontal space for distant bottom reverberation. The adiabatic mode solution is introduced to derive the mathematic mode for horizontal correlation in the range-dependent waveguide with varying depth and the numerical results indicate that the influence of inclined sea floor on the horizontal correlation should be considered.

3:20

1pUWb5. Inversion of elastic bottom parameters from reflection data. Han-hao Zhu, Hai-gang Zhang, Sheng-chun Piao, and Wei Liu (Harbin Engineering University, ShuiSheng Building Room 1304, 145 Nantong Street, Harbin 150001, China, zhuhanhao@hrbeu.edu.cn)

In this paper, the wave reflection at the interface between sea water and the half-space elastic bottom is considered and the sensitivity of reflection coefficient to the geoacoustic parameters, such as bottom density, P-wave

velocity and its attenuation, S-wave velocity and its attenuation, has also been analyzed. In order to establish a geoacoustic inversion method with the ocean bottom reflection coefficient, a simulated experiment is carried out in the laboratory tank, where a PVC plate is used as the elastic bottom. In the experiment, a high-frequency underwater sound wave is transmitted by a source at fixed position and received using a hydrophone at different position with equal interval. By processing the measurement reflected signals at different receiving positions, the reflection coefficient for different incident angles can be obtained, with which the simulated bottom parameters have been inverted. An inversion method based on the sound transmission loss in water is also accomplished. These two inversion results are compared with the measurement result obtained according to the time delay for received multipath signals which reflect/refract at the liquid/elastic or solid/liquid interface of the PVC plate and the feasibility and reliability of the inversion from reflection data are discussed at last.

3:40

1pUWb6. Assessing shear property variability in shallow water sediments using a wide aperture geophone array. Henrik Schmidt (MIT, Cambridge, MA 02139, henrik@mit.edu), Kodali V. Rao (VASA Assoc., McLean, VA 22102), Patrick Edson, and Peter Stein (Scientific Solutions, Inc., Nashua, NH 03049)

The significance of seabed shear properties to low-frequency propagation in shallow water is well established, and since the early 1980's the measurement of the properties of the seismic interface, or Scholte waves, has been recognized as the most direct tool for determining the shear properties. In Sep. 2011, an experiment was carried out in Singapore harbor aimed at investigating the excitation of Scholte waves by objects dropped from the surface and impacting the seabed. For that purpose, a 100 m aperture array of 10 ocean bottom seismometers (OBS) with logarithmic spacing was deployed in approximately 16 m of water, in an area with close to range-independent seabed stratification. Spherical and cylindrical objects were equipped with 6 degree-of-freedom motion packages which allowed for accurate measurement of the impact forcing. The objects were dropped onto the seabed at various bearings and distances relative to the array, providing a unique, rich data set which provides the opportunity of estimating the statistics of the seabed shear properties. This paper will describe the experimental setup, and the spatial variability of the phase- and group velocity, and attenuation of the Scholte waves and the associated variability of the seabed geoacoustics will be assessed. [Work supported by DSO National Laboratories, Singapore].

4:00–4:20 Break

4:20

1pUWb7. Rapid bottom characterization using reverberation data. Jinrong Wu, Zhendong Zhao, and Erchang Shang (Key Laboratory of Underwater Acoustic Environment, Institute of Acoustics, Chinese Academy of Sciences, 100190, ymwj@yaho.com.cn)

Bottom characterization can be described by Geoacoustic model or bottom reflection model. In this work, simplified bottom reflection model in small grazing angle area was preferred. There are only two parameters P and Q in the model. P contains the phase shift information of bottom reflection. Q shows attenuation characteristics of bottom reflection. Spectrograms of reverberation data show evident coherent striations. The waveguide invariant, Beta, was extracted from these striations. For a Pekeris waveguide, a very simple analytic relation has been given: $\text{Beta} = 1 + P/(k_0 \text{Heff})$, here Heff is the 'effective depth', and $\text{Heff} = H + P/2k_0$. P can be deduced using this simple relation firstly. The new developed energy-flux model (with parameters P and Q) of waveguide reverberation based on Perturbation theory was used to inverse Q from the reverberation average intensity decaying line secondly. 2007 Qingdao reverberation experiment data was analysed to illustrate this rapid bottom characterization technique. The result shows that P and Q can be extracted from the reverberation data rapidly. [This work was supported by NSFC under Grant No. 10874201 and No. 11074271]

4:40

1pUWb8. Dispersion deformation due to bottom model mismatching. Zhendong Zhao, Jinrong Wu, Erchang Shang, and Li Ma (Key Laboratory of Underwater Acoustic Environment, Institute of Acoustics, Chinese Academy of Science, No. 21, BeiSiHuan XiLu, Beijing, China, 100190, zhaozhendong@yahoo.cn)

The development of MFP (Matched Field Processing) has played an important role in the inversion of the sediment parameters. However, how to characterize the effect of sediment on the propagation of sound is still far from the last answer. A popular solution is making use of the GA (Geoacoustic) model. In practice, it is always impossible to pre-know the layout of the real sediment, so a supposed model is used. For the narrow frequency band case, the supposed GA model may predict the sound field well. But in a broadband case, there will be error if the supposed model mismatches the real one, which will lead to dispersion deformation. Besides, the GA model involves many unknown parameters, especially for a complex model. To overcome such problems, a model-free method named RBC (Rapid Bottom Characteristic) based on only two parameters, the phase-shift parameter P and the absorption parameter Q in bottom reflection coefficient, has been provided. The effect of RBC for a broadband sound field is showed by simulation, and the problem of dispersion deformation is solved well. [This work was supported by NSFC under Grant No. 10874201 and No. 11074271]

5:00

1pUWb9. A numerical study of interferometric imaging in underwater acoustics. Yingzi Ying and Ying Wu (Mathematical and Computer Sciences and Engineering Division, King Abdullah University of Science and Technology, Thuwal, Jeddah 23955, Saudi Arabia, yingzi.ying@kaust.edu.sa)

Time reversal acoustics has long been a prevailing concept and became fruitful in underwater acoustics community. While the interferometric

imaging, or interferometry, which migrates the cross-correlations over suitable time interval of the traces received at the array, is mostly used in the reconstruction of subsurface geometry in exploration geophysics. In this numerical study, the interferometric imaging method is used to localize the active underwater targets. The intrinsic relationship between the interferometry and time reversal is revealed by back-propagating the received and time reversed signals into a fictitious waveguide model. When the frequency decoherent parameter is infinitely small, the imaging estimator becomes the conventional Bartlett matched field processing, whereas the Kirchhoff migration is achieved when frequency decoherent parameter is equal to the full band. A comparison of the imaging quality with different frequency decoherent parameters is performed through the normal-mode based simulation, and the results show the interferometric imaging is feasible and effective in underwater acoustics. This research was supported by KAUST start-up package.

5:20

1pUWb10. Short range sound propagation in shallow water and geoacoustic parameters inversion. Xuegang Zhang (Dalian Scientific Test and Control Technology Institute, Dalian 116013, China, xuegangzhang@126.com), Chunxia Meng, Haohao Hu, and Jing Han

The distribution of short range sound field is complicated in shallow water because layered structures of seabed and multiple reflection from surface have significantly effect on field, propagation model is established by using fast field program. Matched-field processing experiments are carried out respectively during two typical hydrographic seasons in north Yellow Sea. The seabed parameters are inverted based on Bayesian theory. The Results show that inverted parameters of two experimental data had a well consistency and multiple layered structure of seabed affected sound field at short range.

MONDAY AFTERNOON, 14 MAY 2012

THEATRE 1, 1:00 P.M. TO 2:00 P.M.
2:30 P.M. TO 3:30 P.M.
4:00 P.M. TO 5:00 P.M.

ELECTROACOUSTIC MUSIC PERFORMANCE

A concert titled "Bioluminescence and the Dream World—A Puppet and Electroacoustic Music Performance" will be performed on Monday afternoon, 14 May, at 1:00 p.m., 2:30 p.m., and 4:00 p.m. in Theatre 1. Each performance is one hour long.

Bioluminescent underwater creatures will emerge from the murky depths and a giant spider will dance across a colorful web. A puppeteer will play the Native American flute while performing a dance with a Kokopelli puppet. A skeleton will cast a spell over a Theresmin cauldron invoking a host of ghosts from the shadows.

This concert features original electroacoustic music by Lydia Ayers incorporating synthesis of Asian and Western musical instruments. Three puppeteers, and three live musicians will accompany the music synthesis on acoustic instruments.