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.

MONDAY MORNING, 14 MAY 2012 HALL A, 9:15 A.M. TO 12:40 P.M.

Session 1aAA

Architectural Acoustics and Signal Processing in Acoustics: Multiple-Microphone Measurements and Analysis in Room Acoustics I

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Sam Clapp, Cochair
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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 beampattern 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 sphercial 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 wtih 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 Prasanga 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 etal ["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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Hairong Zheng, Cochair
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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 (Sate Key Laboratory of Medical Ultrasound Engineering Co-founded by Chongqing and Ministry of Science and Technology, Departmente of Biomedical Engeering, 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 Aramakiji 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 Aramakiji 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 μs, respectively. In the third sequence, the N and P waves were employed in the earlier and later 62.5 μs, respectively, and they were exchanged in the forth sequence. In the results, the amount of cavitation bubbles generated by the third sequence was significantly more than the other three sequences. The cavitatin 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 eastimating 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 ($\triangle\beta$) and attenuation coefficient intercept ($\triangle\alpha0$) averaged in the region of interest, and to construct $\triangle\beta$, $\triangle\alpha0$, and B-mode images simultaneously. During HIFU treatment, bubble activities were visible as strong hyperechoic regions in the B-mode images, causing fluctuations in $\triangle\beta$ and $\triangle\alpha0$ during treatment. $\triangle\beta$ and $\triangle\alpha0$ 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, $\triangle\beta$ and $\triangle\alpha0$ 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, $\triangle\beta$ and $\triangle\alpha0$ 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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Y. Y. Lee, Cochair
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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 Hershcel-Quincke tube, then according to the principle of the Hershcel-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, varies 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 Tracianne 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, eletroacoustics 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, Eletroacoustics 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
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Jung-Woo Choi, Cochair
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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 5angular 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@riec.tohoku.ac.jp), Makoto Otani (Shinshu Univ., 4-17-1 Wakazato, Nagano, Nagano), and Yôiti 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.

12:00

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@hccl.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.

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.

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 Hoso Bunka Foundation.]

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.

Contributed Papers

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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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 stirring 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 gives 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 constrains 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 as 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 adding 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 Zs and short-term amplitude As, we won't loss detail of signal; By use of proposed "period-amplitude" Ps 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., rreger@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 2o 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 quiets 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 nightitme, 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 polysomonography 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 Shimbashi 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 Linkoping, 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 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.

11:20

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, Hunghom, 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 then 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)

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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 possiblemechanisms (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 $BaTiO_3$ 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 $BaTiO_3$ 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 $BaTiO_3$ 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 Taleyarkhan' 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 (~ 40 μm) 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 Ccav.on, is higher than the cavitation-free counts Ccav.off, and the neutron increment $\triangle$C, i.e. the difference between Ccav.on and Ccav.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)

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.

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), Jungsoon 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.

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 nanopaticles 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

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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.]

Contributed Paper

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 *Jepsenetal., 2008* was used as a front end of a model for auditory stream segregation. A temporal coherence analysis *Elhilalietal., 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 (Wyane 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 discountinuous 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.

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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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. Puisan 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@hccl.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 Institue 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, Jielin Pan, and Yonghong Yan (Key Laboratory of Speech Acoustics and Content Understanding, Institute of Acoustics, Chinese Academy of Sciences, guoyuhong@hccl.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@hccl.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 arouse 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@hccl.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@hccl.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, Jielin Pan, and Yonghong Yan (Key Laboratory of Speech Acoustics and Content Understanding, Chinese Academy of Sciences, zhangqingqing@hccl.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@hccl.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@hccl.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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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 acomms 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 acomms 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. Xingyang 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 resluts.

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, quke09@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 inversed 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 be 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, qunyren@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 Musikvereinssaal 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 emphases 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 Jhuan 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 behavioir 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 place 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 analyisis.

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.

6:40

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

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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 loacation 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 Reseach 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 Reseach 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 rarafactional pressure 1.2 MPa, pulse length 50,000 cycles, pulse repletion 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), Kirthi 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 indicate that the ultrasound acting on the membrane structure will lead to the change of membrane potential, suggesting the piezoelectricity of cell membrane maybe 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 sonopration 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 sonopration 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, sonopration 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, Nederlands, g.p.r.lajoinie@utwente.nl), Ceciel Chlon, Marcel Bohmer (Philips Research, Eindhoven, Nederlands), Srirang Manohar (University of Twente/BMPI (Biomedical Photonic Imaging), Enschede, Nederlands), and Michel Versluis (University of Twente/POF (Physics of Fluids), Enschede, Nederlands)

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 thermosensitive 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 Netherland), 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 um 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 inidividual 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$\pm$0.1 N/m) was nearly the same as that of BARE bubbles (0.19$\pm$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 10.8kW/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 (Dynaflow, 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 100 mm^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 analyzes 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 detectiong 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 (Insitute 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 scatter'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, msandyl@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, Natonal 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 Fernandez (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 Naning 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

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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. Yôiti 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 Collage 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 Yôiti 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)

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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, zenggxy@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 vibraitonal 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 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 $1m\times1m$ simply supported steel plate. The results show that the radiated acoustic field can be reconstructed accurately under the circumstances of a few measurement points.

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 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
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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, lyan1832@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 complaintants 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; Forhåbningsholms 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 ($<$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.

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 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 workablility. 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 being made and many achievements have being 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 (MTRCorporation 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 additional 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 wayside 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, ggiairxc@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, Chian, 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@krri.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 porduced 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 dege 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 anechic 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
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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, xiexiujuan@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 filed 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), Shinichi 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, zlht13@163.com), Xiujuan 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 (y0/l0, L, f, dTm/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 y0/l0 and dTm/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, Winterhurerstrasse 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, Winterhurerstrasse 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, Xiaomuqiao 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).

Invited Paper

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'Heres, 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' tasks 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 Univerisity 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 angels. 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 efficiently, 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, amwillem@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 rare 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 epenthesizing /ɨ/ 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 /ɨ/ 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 /ɨ/ 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 ɛ æ u ʊ o ɔ ɑ̃ ɔɪ aɪ aʊ/ 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://chinesetone.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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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 LNlog 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 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

1pUWa4. 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

1pUWa5. 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 resamplingat 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

1pUWa6. 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

1pUWa7. 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 k Hz) 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].

Contributed Paper

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 (AWSαSN) model, which better approximates the heavy tailed noise due to snapping shrimp. When converted to the complex baseband representation, the resulting noise for the AWSαSN case is radically different from its Gaussian counterpart. In this talk some properties of baseband noise for the general AWSαSN 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 Sicence and Technololgy (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 trail 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 corder. 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 2 Mπ within 1000 symbols of transported data. In this study, we investigated QPSK acoustic telecommunication systems that incorporate a Turbo corder 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 ODFM 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 inversed. 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 fordetermining 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 statistic 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, ymwjr@yahoo.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: Beta = 1 + P/(k0Heff), here Heff is the 'effective depth', and Heff = H + P/2k0. 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 Theremin 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.

Keynote Lecture

Invited Paper

8:20

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

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

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

Session 2aAAa

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

Kenneth Roy, Cochair
kproy@armstrong.com

Xiang Yan, Cochair
yx@abcd.edu.cn

Chair's Introduction—9:15

Invited Papers

9:20

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

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

9:40

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

11:00

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

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

11:20

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

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

Contributed Paper

11:40

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

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

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

Session 2aAAb

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

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

Sam Clapp, Cochair
clapps@rpi.edu

Chair's Introduction—9:35

Invited Papers

9:40

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

Contributed Papers

11:00

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

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

11:20

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

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

11:40

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

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

12:00

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

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

12:20

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

Adaptive directional microphone arrays have been widely utilized in hearing aids to help wearers understand speech in noisy environments. However, a practical and objective way of measuring the benefit of this technique is not obvious. One commonly used measure for directional microphone arrays is directivity index (DI), which evaluate the arrays' capability to attenuate a diffuse noise field. However, the DI has rarely been considered for the Adaptive directional microphone arrays because a truly diffuse noise field is required for the measurement. This paper investigates the feasibility of measuring the DI of adaptive directional hearing aids in a diffuse noise field generated by a loudspeaker array playing uncorrelated noise signals simultaneously. The requirement for the degree of diffuseness of the noise field is studied based on numerical simulations for first-order and second-order adaptive microphone arrays. The possibility of measuring the DI in both anechoic and non-anechoic conditions is investigated. The accuracy of the proposed approach is further discussed.

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

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

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

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

Session 2aAAc

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

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

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

Session 2aAO

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

Orest Diachok, Cochair
orest.diachok@jhuapl.edu

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

Contributed Papers

9:20

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

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

9:40

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

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

Invited Paper

10:00

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

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

10:20

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

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

10:40–11:00 Break

11:00

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

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

11:20

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

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

11:40

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

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

12:00

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

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

12:20

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

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

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

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

Session 2aBA

Biomedical Acoustics: Biomedical Ultrasound Imaging Instrumentation

Lei Sun, Cochair
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Qifa Zhou, Cochair
qifazhou@usc.edu

Invited Papers

9:20

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

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

9:40

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

11:00

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

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

11:20

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

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

11:40

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

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

12:00

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

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

12:20

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

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

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

Session 2aEA

Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials I

Michael Haberman, Chair
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Invited Papers

9:20

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

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

9:40

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

Contributed Papers

11:00

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

Acoustic concentrator, focusing acoustic field and enhancing acoustic energy in a region, is presented. The concentrating performances and the scattering properties of the acoustic concentrator with multilayered alternative homogeneous materials at different frequencies are investigated by the frequency response analysis with finite-element method (FEM). The calculation results show that it has an optimized relation between the acoustic concentrating performances in the inner region and the scattering properties in the outer region of these concentrators. A kind of acoustic concentrators constructed by an array of cylindrical rigid scatterers in fluid is proposed. The multiple scattering theory (MST) is introduced to describe the superposition of the external incident acoustic field and the radiative scattered field induced by the array, by which the structure of the array and the performance of acoustic concentrator can be optimized. By the way, the MST can be used to estimate and optimize not only the acoustic concentrators but also other potential acoustic metamaterials. Acknowledgment: This work is supported by National Natural Science Foundation of China (No. 11004099 and 11174142), State Key Laboratory of Acoustics of Chinese Academy of Sciences, and also PAPD of Jiangsu Higher Education Institutions.

11:20

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

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

2a TUE. AM

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

11:40

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

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

12:00

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

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

12:20

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

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

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

Session 2aED

Education in Acoustics: Engaging in Effective Teaching Methods in Acoustics

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

11:00

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

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

11:20

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

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

11:40

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

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

12:00

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

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

Session 2aHT

Hot Topics: Community Noise Policy Development I

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

Invited Papers

9:20

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

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

9:40

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

11:00

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

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

11:20

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

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

11:40

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

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

12:00

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

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

Session 2aMU

Musical Acoustics: Asian Wind Instruments

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Invited Papers

9:40

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

11:00

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

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

11:20

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

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

11:40

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

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

Session 2aNSa

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

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Contributed Papers

9:20

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

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

9:40

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

11:00

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

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

11:20

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

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

11:40

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

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

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

Session 2aNSb

Noise: Numerical Methods in Noise I

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Contributed Papers

12:00

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

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

12:20

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

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

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

Session 2aNSc

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

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Invited Papers

9:20

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

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

9:40

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

11:00

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

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

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

Session 2aPA

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

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

9:20

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

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

9:40

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

11:00

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

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

11:20

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

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

Contributed Papers

11:40

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

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

12:00

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

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

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

12:20

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

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

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

Session 2aPP

Psychological and Physiological Acoustics: Binaural Hearing and Cochlear Mechanics

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Wilson Ho, Cochair
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Contributed Papers

9:20

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

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

9:40

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

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

11:00

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

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

11:20

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

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

11:40

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

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

12:00

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

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

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

12:20

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

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

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

Session 2aSA

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

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Invited Papers

9:20

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

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

9:40

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

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

Contributed Paper

10:00

2aSA3. Fault diagnosis of railway roller bearing based on vibration analysis and information fusion. Bin Chen (School of Automation, Beijing University of Posts and Telecommunications, No. 10, West Tu Cheng Road, Haidian District, Beijing 100876, P.R. China; Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, chenbin@mail.ioa.ac.cn), Zhaoli Yan, Xiaobin Cheng, and Wei Liu (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China)

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

Invited Paper

10:20

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

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

10:40–11:00 Break

Contributed Papers

11:00

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

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

11:20

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

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

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

11:40

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

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

12:00

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

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

12:20

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

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

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

Session 2aSC

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

Puisan Wong, Chair
pswresearch@gmail.com

Contributed Papers

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Previous research has shown that roughness perception for vowels is influenced by the waveform amplitude modulation. It was observed that sinusoidal modulation (constant modulation depth of 100%) between 20 Hz and 70 Hz has the greatest impact on perceived roughness of vowels [Shrivastav & Eddins, 2011; JASA, 129, 2661]. The present experiment examined the effects of modulation depth on the perception of roughness for two of 10 vowels used by Shrivastav & Eddins (2011). Synthetic copies of two vowel stimuli selected from the Sataloff/Heman-Ackah database were generated using a Klatt synthesizer. Sinusoidal amplitude modulation was superimposed on each vowel using three modulation frequencies (20 Hz, 30 Hz, and 45 Hz) and five modulation depths (0, -5, -10, -15, and -20 dB). Listeners judged the perceived roughness of each of the 30 speech stimuli (2 talkers x 3 modulation frequencies X 5 modulation depths) by matching it to a comparison sound. The comparison sound consisted of a sawtooth wave, mixed with speech-shaped noise, and amplitude modulated with a raised (power of 4) cosine wave. Results will show how depth of amplitude modulation affects perceived roughness and will help to develop predictive models to quantify roughness. Research funded by NIH.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Session 2aSP

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

Ning Xiang, Cochair
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Chair's Introduction—9:15

Invited Papers

9:20

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

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

9:40

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

Contributed Papers

11:00

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

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

11:20

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

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

11:40

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

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

12:00

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

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

12:20

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

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

Session 2aUW

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

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Fenghua Li, Cochair
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Contributed Papers

9:20

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

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

9:40

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

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

10:00

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

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

10:20

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

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

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

10:40–11:00 Break

11:00

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

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

11:20

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

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

11:40

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

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

12:00

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

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

12:20

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

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

Session 2pAAa

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

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Chair's Introduction—1:55

Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

Session 2pAAb

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

Angelo Campanella, Cochair
a.campanella@att.net

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

Chair's Introduction—1:55

Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

Contributed Papers

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

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

5:20

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

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

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

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

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

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

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

Session 2pBA

Biomedical Acoustics and Physical Acoustics: Subharmonic Contrast Imaging

Michel Versluis, Cochair
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Jeffrey Ketterling, Cochair
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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

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

3:20

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

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

2p TUE. PM

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

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

Contributed Papers

5:00

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

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

5:20

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

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

Session 2pEA

Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials II

Michael Haberman, Chair
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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

Contributed Papers

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

2pEA9. Experimental study of backscattering enhancement using acoustic double-positive metamaterials. Wenlin Hu (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, huwenlin1984@yahoo.cn), Yuxian Fan, Peifeng Ji, and Jun Yang (ditto)

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

5:20

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

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

5:40

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

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

6:00

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

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

6:20

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

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

6:40

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

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

Session 2pHT

Hot Topics: Community Noise Policy Development II

Marion Burgess, Cochair
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Aaron Lui, Cochair
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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20–6:00

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

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

Session 2pMU

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

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2p TUE. PM

Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

Contributed Papers

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

The acoustics of the instrument recording involves acoustic characteristics of the instrument as a sound source, the microphone characteristics for

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

5:00

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

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

5:20

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

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

Session 2pNSa

Noise: Numerical Methods in Noise II

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Contributed Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

2pNSa5. Development of a practical method for determining noise contribution from a large extruded panel using sound intensity method and optimized finite element model. Anne Shen, Jiumei Cheng, and Fusheng Sui (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, anne.xy.sh@gmail.com)

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

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

5:40

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

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

6:00

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

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

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

Session 2pNSb

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

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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

3:40

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

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

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

4:00–4:20 Break

Contributed Papers

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

6:00

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

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

Session 2pPA

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

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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

Functional heterogeneity among different cells of an organism (brain cell, heart cell etc.) is brought about by expression of different subsets of genes drawn from the same genetic template (DNA). Therefore, to understand the molecular basis of normal (healthy) and abnormal (diseased) cell behavior, one must measure gene expression. The typical procedure is to homogenize large numbers (>1,000) of cells together, isolate the first product of gene expression (RNA), reverse transcribe the RNA to a cDNA copy, then identify and quantify gene-specific cDNAs. Unfortunately, the gene expression profile obtained represents an average across all combined cells and cell behaviors rather than any particular cell. Ideally one would measure gene expression in a single cell; however the very small amount of labile RNA obtainable from a single cell mostly degrades before it can be measured. We have developed an acoustic microstreaming-based device ("micromixer") which improves mixing of solutions within microliter volumes. Here we show application of "micromixing" to standard laboratory reverse transcription reactions significantly improves conversion of single-cell amounts of RNA to cDNA. Micromixing is therefore a low-cost and easy-to-use technology that is compatible with and can be added to standard laboratory hardware, software, reagents and expertise to enable better gene expression measurement from single cells.

3:00

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

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

Contributed Papers

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

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

Session 2pPP

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

Ruth Litovsky, Cochair
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Liang Li, Cochair
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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

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

Session 2pSA

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

Zhuang Li, Cochair
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Hongwei Liu, Cochair
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Wilson Ho, Cochair
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Invited Paper

2:00

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

A highway composed of surface and elevated sections opened with very little traffic and without noise barriers along the elevated sections to help meet the relatively stringent noise control targets. However, some noise control measures that were added to the basic design included increased height to the guard barrier, low noise road surface pavement, and green buffer zones. Now authorities wish to determine the traffic flow conditions for which noise barriers would be required on the elevated sections. Of course this result can be determined by iteratively stepping through the variables, but this would take great effort. This paper presents a semi-theoretical semi-empirical method that simplifies the problem. Let K be the difference between noise from surface and elevated sections of the road. The noise level is calculated for the surface sections including such factors as speed, percent buses, percent of traffic at night, and other noise related factors from transportation engineering. Then, the noise level of the elevated sections is estimated from the levels calculated for the surface sections added to K. This estimated level is compared to the noise limit, and from this one can establish the traffic flow which would require noise barriers on the elevated sections.

Contributed Papers

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

6:00

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

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

6:20

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

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

6:40

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

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

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

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

Session 2pSC

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

Charles B. Chang, Chair
cbchang@umd.edu

Contributed Papers

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

This study examined the effects of prosodic position on duration and F0 of the six tones in Si-Xien Hakka. Each of the tones was placed in the initial, medial, and final positions of a three-syllable phrase/clause, which constituted the first part of a sentence with the [$\delta\delta\delta$, $\delta\delta\delta$] structure. The results showed that F0 lowering and lengthening were found at domain final position. The final lengthening allowed the offset of the low-rising tones to reach a higher F0 target, not attainable by those in other prosodic positions. Also in this final position the high level tones exhibited a real flap shape due to the greater coarticulation resistance caused by the following boundary breaks. Furthermore, the F0 decreasing rate of the falling tones was found to vary according to domain position: the F0 descended the fastest at domain medial position and the slowest at final position, except for the high checked tones, which showed the same rate at domain-initial and –final positions. This reveals that the vibration of the vocal folds, reflected by the F0 decreasing rate, started from the initial position, gradually speeded up to the highest rate at medial position and then slowed down till the pre-boundary domain-final one.

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

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

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

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

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

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

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

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

2p TUE. PM

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Session 2pSP

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

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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

The unknown geoacoustic environment often limits sonar-processing performance. Thus, there is a need for better estimates of the geoacoustic parameters. From the start of geoacoustic inversion more than two decades ago this has been model-based. Advanced propagation codes are based on ray tracing, wavenumber integration, normal modes or parabolic equations. The environment model is parameterized using ocean sound-speed and sediment sound-speed, density and attenuation. In some cases range-dependent models has been used. Initially classical least squares were used. But early on stochastic optimization methods as simulated annealing and genetic algorithms became popular. Understanding of uncertainties is important for the development of the methods. A major effort has been addressed using Bayesian methods and Markov chain Monte Carlo methods. Uncertainties are used to find the most important parameters and the obtained geoacoustic inversion uncertainties can be mapped into sonar performance assessment. In a typical experimental scenario, the data arrives sequentially and new improved estimates can be obtained by updating previous estimates at significantly less computational expense than if each data observation were treated independently. Current efforts are focused on model-based geoacoustic inversion sequentially using sequential methods such as particle filters or Kalman-family of filters.

Contributed Papers

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

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

5:20

2pSP10. Design and deployment of infrasonic sensor monitoring network in China. Haonan Feng, Yichun Yang, and Chunlian Men (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, fhn02212005@163.com)

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

5:40

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

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

6:00

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

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

6:20

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

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

6:40

2pSP14. Dialectal alarm words recognition based on a hybrid model of Hidden Markov Models and the BP Neural Network. Ling Lu (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, luling12345678@yahoo.cn), Xiangyang Zeng (Institute of Environmental Engineering, College of Marine Engineering, Northwestern Polytechnical University, Xi'an 710072, China), Xiaobin Cheng, and Zhaoli Yan (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China)

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

Session 2pUWa

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

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Peter Worcester, Cochair
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Invited Paper

2:00

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

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

Contributed Papers

2:20

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

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

2:40

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

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

6:00

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

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

6:20

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

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

6:40

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

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

Session 2pUWb

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

Marcia Isakson, Cochair
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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

Session 3aAA

Architectural Acoustics and Noise: Acoustics in Concert Halls II (Lecture/Poster Session)

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Chair's Introduction—8:55

Contributed Papers

9:00

3aAA1. Practice and cogitation on acoustics design of professional concert hall. Song Yongmin and Zhang Kuisheng (Zhang Kuisheng Acoustical Design and Research Studio, 11F, No. 268, Shimen Er Road, Shanghai 200041, China, asong1102@163.com)

Development overview on design and construction of professional concert hall since the 1990s in China is summarized in this paper. Acoustics design cases of professional concert hall completed by Zhang Kuisheng acoustical design and research studio over the past decade are described. Then four designs among those professional concert halls including the architectural characteristics and acoustical parameters are introduced. Those are Hunan musical hall, concert hall in China conservatory, Beijing, philharmonic hall in Xiamen international conference center and Yangzhou musical hall. Finally, the preferred reverberation time, plane shape, volume per seat, acoustical material and diffuser design in concert hall are discussed in detail. Those successful experience and considered suggest are put forwards for sharing.

9:20

3aAA2. The acoustical design of Xiamen University Concert Hall. Yuezhe Zhao and Shuoxian Wu (State Key Laboratory of Subtropical Building Science, South China Univ. of Tech., Guangzhou 510640, arzhyzh@scut.edu.cn)

Xiamen University is located in the Fujian Province, South East of China. Xiamen University Concert Hall is the essential musical education site for the college students, as well as for holding important meetings. The concert hall has 560 seats. It has a comparatively long reverberation time to offer a fullness and reverberant sound field for music performance. At the same time, rich early reflections from the ceiling and the side walls together with the enough signal to noise ratio guarantee speech intelligibility when using a loudspeaker system. With this conception in the acoustical design of the concert hall, the use of changeable absorptive components was avoided which may influence the brightness and liveness of the chamber music hall. After the completion of the building, acoustical measurements were taken and the measurement results are also given in this paper. Since autumn 2010 when the concert hall was finished, music performance and meeting events have been successfully held in the hall. It has become the main hall for the musical education and performance, as well as the meeting center for the college.

9:40

3aAA3. Prediction of acoustical parameters for open plan offices according to ISO 3382-3. Jens Holger Rindel (Odeon A/S, Scion-DTU, Diplomvej 381, DK-2800 Kgs. Lyngby, Denmark, jhr@odeon.dk)

In the new international standard ISO 3382-3 the measurement procedure for open plan offices is described and a number of new room acoustical parameters for the objective evaluation are defined. Among the new parameters are the privacy distance and the distraction distance, both derived from the STI (speech transmission index) as a function of distance from the sound source. The final evaluation is a balanced compromise between a number of parameters that depend on the amount of sound absorption, the application of screens between work stations and the level of background noise. With room acoustic simulation software these measurements can be simulated, thus providing a tool for the acoustical design of open plan offices. The paper presents an example office with a range of alternative acoustical solutions that include different amount of absorption, screens of different height, and different levels of background noise. Also the influence of dynamic background noise from people talking can be taken into account, leading to a shorter privacy distance. This provides a background for the discussion of the efficiency of various acoustical measures in open plan office design.

10:00

3aAA4. Using synthesised musical stimuli to measure room impulse responses in occupied spaces. Francis Li (University of Salford, M5 4WT, UK, f.f.li@salford.ac.uk)

Since Sabine instigated objective parameters a century ago, the advance of room acoustics has been centred around measurement findings. Room impulse responses are of particular importance: nowadays, measurements typically start from impulse responses, individual acoustic parameters are then derived. With existing methods, noisy signals intolerable to audience are employed as stimuli to obtain the impulse responses. Consequently, occupied measurements are rarely carried out. Unoccupied measurements are unreliable and problematic, because they do not accurately represent the acoustic profile under in-use conditions. This paper presents a method that utilises "presto-chirps", short chirps centred on musical notes in an equal temperament scale, to synthesise musical notes. The harmonics might be added by the Volterra kernel convolution, if richer tones are preferred. These musical notes are used to compose synthesised music as test stimuli. Room impulse responses are deconvolved from received signals in musical

note specified sub-bands. Following energy normalisation and superposition broadband impulse responses are obtained. The use of musical stimuli facilitates occupied measurements. The purpose-defined presto-chirp enables the measurement to be completed in a relatively short duration, mitigating time variance problems.

10:20

3aAA5. How the design of a balcony affects the acoustics in an auditorium? L.Y. Cheung and S.K. Tang (Department of Building Services Engineering, the Hong Kong Polytechnic University, Hong Kong, louisa.cheung@connect.polyu.hk)

A 3D-simulation for improving the acoustics in an auditorium was done using room acoustic software, Odeon. The software provides the sound field inside the hall and thus the sound propagation physics was studied. The 3D computer model was done based on an existing auditorium of over 1000 seats. Further simulations have been done by removing the balcony and changing it's the geometry. The effect of different balcony geometry on the sound quality inside the hall was estimated and compared. Apart from reverberant time, the effects on energy ratios and other parameters from simulation results were studied. [L. Y. Cheung is supported by the Hong Kong Polytechnic University.]

10:40–11:00 Break

11:00

3aAA6. Vocal intelligibility and clarity in amplification: challenges for concert hall acoustics. Damian Doria (75 Feather Lane, Guilford, CT 06437, DamianJDoria@gmail.com), Tom Clark (Acme Sound, 23 Park Lane, Norwalk, CT, 06854), Todd L. Brooks (Artec, 114 West 26th Street, New York, NY 10001), and Bob McCarthy (Alignment and Design Inc, 204 Falling Leaves Court, Creve Couer, Missouri 63141)

The modern concert hall presents a range of programming from soloists and small classical ensembles to large orchestras with choir. In many communities the same concert venue is also required to host a range of amplified events from politely-reinforced ensembles to overtly amplified popular music artists. Maintaining intelligibility of vocals and clarity at high amplification levels depends upon a number of factors both within the control of the acoustics and sound system designers and not. This paper will discuss the programming for a typical season at one 1600-seat concert hall, the elements within its design to allow flexibility of program use, and experiences during the process of adjustment, tuning, and optimizing the hall over its first season.

11:20

3aAA7. Acoustic re-radiation of structurally coupled instruments by stage floors and orchestra risers. Clemeth L. Abercrombie, Todd L. Brooks, Damian J. Doria, and Tateo Nakajima (Artec Consultants Inc, New York, cla@artecconsultants.com)

The role that structural vibration plays on concert hall stage floors is a point of continous discussion in several lines of research. It is often discussed, if only briefly, in research relating to the perception of warmth in performance spaces, low-frequency sound propagation, and the musician's acoustic environment. Three effects have been researched: acoustic absorption, acoustic reradiation of structurally coupled instruments, and perceptible tactile stimulation. New measurements have been conducted to explore the role of reradiation by wood stage floor constructions and orchestra risers in two recently opened concert halls. Results of these measurements will be presented and reviewed in the context of previous research. Implications for design of musical environments will be discussed.

11:40

3aAA8. Effects of periodically arranged absorptive materials on acoustic potential energy in rectangular rooms. Hao Liu and Xiaodong Li (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, Beijing, China, liuhao@mail.ioa.ac.cn)

A modal analysis method is applied to compute the acoustic potential energy of a three-dimensional rectangular room, where sound absorptive materials are periodically arranged on the boundaries of the rectangular room. This paper studies the influence of two parameters, including the frequency of the sound source and the number of absorptive material strips, on the acoustic potential energy in detail. This study reveals that both of the two parameters have notable effects on the coupling behavior of room modes. A main result of this paper is that using a periodic arrangement of absorptive materials can be more efficient than totally covering of absorptive materials on corresponding surface in reducing acoustic potential energy under certain conditions. The coupling behavior of room modes can well explain why excess potential energy reduction can be achieved by using periodically arranged absorptive materials.

12:00

3aAA9. Acoustical design of the National Theatre Company of China (NTCC). Chan Chun Huang, You Guo Qin, Xiang Yan, and Peng Wang (Tsinghua University, School of Architecture, Beijing, China, james601129@hotmail.com)

The National Theatre Company of China seats 886, volume 5,400 m3, and reverberation time is 1.2s without audience and orchestra. As part of the design process, measurements on CAD computer and acoustical simulation using Raynoise software and full-sized materials samples were conducted over two year period. The hall in plan is bell shape. The ceiling is a curved cloud, with its U-shape balcony above the main floor (stall), two side rails provide the useful early reflections to the central rear seating area. The unique wall architectural features around the auditorium was analyzed on the models that all interior surfaces combine to distribute sources on the stage uniformly over the seating areas and to yield optimum values for reverberation time–RT, early decay time–EDT, Lateral fraction–LF, and strength–G.

The following abstracts will be presented in poster format. The posters will be on display and the authors will be at their posters from 12:20 p.m. to 1:00 p.m.

3aAA10. Investigation of improved metrics for the characterization of musicians' room acoustical conditions in concert halls. Behzad Ranjbari (Chalmers Room Acoustics Group, Chalmers University of Technology, SE-41296 Gothenburg, Sweden, br@behzadranjbari.com)

A number of metrics for assessing the acoustical conditions for performers on concert hall stages have been proposed, notably by Anders Gade but also others. However, the subjective relevance of existing stage acoustic metrics for musicians, appears mainly to be associated with the communication with the audience rather than with the communication between musicians. No acoustic metrics have been identified to assess the balance between the hearing of others vs. the hearing of one's own instrument, which appears paramount to orchestral musicians. Problems regarding presence of orchestra, directional characteristics of instruments, distances from instruments to ears of musicians, etc., also have been an issue for researchers, making the work difficult, expensive and imprecise. However, in this paper, due to the comparative approach used, some of these problems were removed, since they are basically the result of properties of orchestral arrangement rather than stage conditions and can be assumed similar from one stage to another. In this paper, a number of laboratory experiments as well as measurements on real stages have been studied and a pair of metrics, namely G_{Self} and G_{Others} are suggested to assess the balance between the hearing of self and that of hearing others.

3aAA11. Acoustical design of the new concert hall at Helsinki Music Centre, Helsinki, Finland. Keiji Oguchi (Nagata Acoustics, Hongo Segawa Bldg.3F, 2-35-10 Hongo, Bunkyo-ku, Tokyo 113-0033, Japan, oguchi@nagata.co.jp), Motoo Komoda (Nagata Acoustics, 2130 Sawtelle Blvd., Suite 308, Los Angeles, CA 90025, U.S.A.), Ayako Hakozaki (Nagata Acoustics, Hongo Segawa Bldg. 3F, 2-35-10 Hongo, Bunkyo-ku, Tokyo 113-0033, Japan), Marc Quiquerez (Nagata Acoustics, 75, avenue Parmentier, 75011 Paris, France), and Yasuhisa Toyota (Nagata Acoustics, 2130 Sawtelle Blvd., Suite 308, Los Angeles, CA 90025, U.S.A.)

The Helsinki Music Centre opened in Helsinki, Finland on August 31, 2011. The Music Centre is anchored by the 1,704-seat concert hall, and is supported by 6 small halls, each with a different program, and the Sibelius Academy, the premier conservatory in Finland. The concert hall is the new home for Finland's two major orchestras, the Finnish Radio Symphony Orchestra and the Helsinki Philharmonic Orchestra. The concert hall is configured in the vineyard style with terraced steps in the auditorium, and measures 35m wide, 48m long and 20m tall. To support the stage acoustics, a 12m diameter ensemble reflector is suspended at a height of approximately 15m above the stage. The stage acoustics are also enhanced by motorized orchestra risers. The acoustical design and characteristics of the Concert Hall are reported.

3aAA12. Acoustic design of the Philharmonic Hall in the Shanghai Oriental Art Center. Jingbo Wang and Kuisheng Zhang (Shanghai XianDai Architectural Design (Group) Co., Ltd., Shanghai, 200041, jbwang827@163.com)

The Shanghai Oriental Art Center opened officially on July. 2005 is one of the largest performance art centers in the domestic. It mainly consists of the 2000 seats philharmonic hall, the 1100 seats opera house and the 300 seats chamber music hall, and several rehearsal facilities. It has been become one of the most important landmark buildings in Shanghai as well in PuDong new district. In this paper it is mainly elucidated the features of the architectural acoustics design of the 2000-seat philharmonic hall, the largest specially designed for music performance venue. It is also covered the objective acoustical measurement results and the subjective acoustics evaluation from all circles after opening.

3aAA13. Spring isolators designed for the Melbourne Recital Centre, Melbourne, Australia. Michael Plumb (Embelton, m.plumb@embelton.com)

The subject matter deals with the performance of spring isolators designed to meet the specified isolation requirement for the 1000 seat Dame Elisabeth Murdoch Recital Hall located in Melbourne Australia. Predicted performance of isolators is evaluated, and measured vibration results are presented for isolated and non-isolated sections of the building. Results are presented in vibration reduction versus frequency with good agreement between predicted and achieved vibration reduction for the locations measured. Achieved background noise levels are also presented in order to provide comparison with anticipated outcomes.

3aAA14. Factor analysis of reverberation perception. Zihou Meng and Lu Dai (Communication Acoustics Laboratory, Communication University of China, Beijing 100024, P.R. China, mzh@cuc.edu.cn)

The reverberation perception in a concert hall is affected by many attributes although the reverberation time may be the most important. To study the contribution to the perceived reverberation from different attributes, a series of listening tests are carried out. The semantic meaning of the reverberation is studied with the questionary to audio engineers and architectural acoustician. The factors to be analyzed are selected based on this semantic study. The subjects are asked to judge the running reverberation impression caused by acoustic scenes simulated by measured or virtual room impulse responses. The reverberation time of the room impulses used in the tests to stimulate the sound field are from 0.5 to 6.0 seconds. Besides the reverberation time, the attributes taken into the consideration in the factor analysis include the clarity, the sound level, the energy ration of early to late reflection, the distance between the source and listener, the volume and shape of the room and also the content of the sound. Based on the result of the factor analysis, a functional expression is proposed to describe the relationship between the perceived reverberation and a group of attributes.

WEDNESDAY MORNING, 16 MAY 2012 S424, 8:20 A.M. TO 12:40 P.M.

Session 3aAB

Animal Bioacoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Biosonar I

James Simmons, Cochair
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Hiroshi Riquimaroux, Cochair
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Invited Papers

8:20

3aAB1. Estimating the biosonar detection range of mesopelagic patches by spinner dolphins. Whitlow Au, Marc Lammers (HIMB, P.O. Box 1106, Kailua, HI 96734, wau@hawaii.edu), and Jokob Jung (Bremen University of Applied Science, Berman, Germany)

Spinner dolphins (*Stenella longirostris*) in the near-shore waters of the Hawaiian islands forage on the mesopelagic boundary community (mbc) of organisms consisting of myctophids, mid-water shrimp and small squids. They forage at night and supposedly in a coordinated fashion and in groups between 16 and 24 animals that are divided into pairs. In the search phase of the foraging process, the dolphins are thought to be spaced in a V-shape formation with the tip of the "V" at the deepest depth, and swim parallel to shore hunting for patches of prey that they can encircle and herd into a tight three-dimensional patch. A profiler housing a broadband echo-ranger that

projected dolphin-like biosonar signals was used to measure the target strength of the mbc based on a dolphin's integration window of 264 μs. The bottlenose dolphin, *Tursiops truncatus*, was used as a proxy to estimate the biosonar detection ranges of *Stenella longirostris* searching for mbc patches because only limited acoustic research has been performed with spinner dolphins. Using the sonar equation, the biosonar threshold detection range of spinner dolphins was estimated to be approximately 50 to 64 m, more than sufficient range for the animals to formulate their prey herding behavior.

8:40

3aAB2. FM bats use transform-based biosonar imaging. James Simmons (Brown University, Providence, RI 02912, james_simmons@brown.edu), and Michaela Warnecke (Brown University, Providence, RI 02912)

Big brown bats (Eptesicus fuscus) perceive target range from echo delay using neuronal-delay lines that extract and display the time that elapses between successive occurrences of individual frequencies in each FM broadcast and its subsequent echoes. The bat's delay accuracy is very sharp for echoes containing all broadcast frequencies. Decreasing the bandwidth of echoes reduces the number of frequencies available for estimating delay, which degrades acuity. However, removal of frequencies triggers the operation of a second, parallel estimation process, loosely related to the cepstrum, whereby delay differences between two or three closely-spaced reflections are estimated from the pattern of modulations (peaks and notches) in the echo spectrum. When a wide span of frequencies is removed, as in lowpass filtering, multiple spectral-modulation solutions coexist, which the bat perceives as many different glints distributed across several hundred microseconds. This defocuses the bat's images for the echoes and renders them not capable of clutter masking. The effect of spectral blurring on delay images is opposite to that expected from time-domain blurring, as demonstrated in 3 different delay-discrimination experiments. [Work supported by ONR and NSF]

9:00

3aAB3. Overcompensation of echo attenuation and dual-component biosonar control in echolocation of an Atlantic bottlenose dolphin (*Tursiops truncatus*). Songhai Li, Paul E. Nachtigall, and Marlee Breese (Marine Mammal Research Program, Hawaii Institute of Marine Biology, University of Hawaii, P.O. Box 1106, Kailua, Hawaii 96734, songhai@hawaii.edu)

Transmitting biosonar clicks and auditory evoked potential (AEP) responses triggered by the clicks were synchronously recorded during echolocation in an Atlantic bottlenose dolphin (Tursiops truncatus) trained to wear suction-cup EEG electrodes and to detect targets by echolocation. Three targets with target strengths of –34, –28 and –22 dB were used at a target distance of 2 to 6.5 m for each target. The results demonstrated that the AEP appeared to both transmitting echolocation clicks and echoes during echolocation, with AEP complexes consisting of alternative positive and negative waves. The echo-related AEP amplitudes were obviously lower than the transmitting click-related AEP amplitudes for all the targets at the investigated target distances. However, for targets with target strengths of –22 and –28 dB, the echo-related AEP response amplitudes increased at further target distances, demonstrating an overcompensation of echo attenuation with target distance in the echo-perception system of the dolphin biosonar. Measurement and analysis of transmitting click intensities showed that the click levels increased with target distance (R). The results demonstrated that a dual-component biosonar control system formed by intensity compensation behavior in both the transmission and receiving phases of a biosonar cycle exists synchronously in the dolphin biosonar system.

9:20

3aAB4. Adaptive echolocation behavior in a complex sonar scene. Cynthia Moss (University of Maryland, College Park, MD 20742, cynthia.moss@gmail.com)

Adaptive sonar behavior in bats provides a window to echo information processing and perception of complex scenes. Echolocating bats produce ultrasonic vocalizations and receive a cascade of echoes from a single sonar call when the environment contains multiple reflecting sources. Some echoes may return from food and others from obstacles, and the bat must rapidly sort and identify the sources of echo returns to analyze the sonar scene and accurately control its flight. Big brown bats produce frequency modulated sonar signals that change in direction, duration, bandwidth and repetition rate as they inspect objects at different directions and distances. The work presented here will focus on adaptive sonar behavior in bats as they track a selected prey item in the presence of other objects, both obstacles and other prey. Data suggest that echo stream segregation of targets in clutter is supported by finely tuned adaptive sonar signal control.

9:40

3aAB5. Evolutionary convergence and divergence in bat and toothed whale biosonars. Peter Teglberg Madsen (Aarhus University, Build 1131, Peter.madsen@biology.au.dk), and Annemarie Surlykke (SDU, Nils Bohrs Alle)

Bats and toothed whales have independently evolved the capability to use echolocation to locate, track and capture prey in a 3-dimensional world of darkness in the night sky or at 1000 meters depth. Despite a very distant common ancestor and the vastly different acoustical properties of air and water, bats and toothed whales use a surprisingly similar ultrasonic frequency range from 15 to 200 kHz for echolocation. In this talk we use recent technical advances in field studies to address and compare the acoustic behavior of bats and toothed whales in the wild. We show that both bats and toothed whales switch to high repetition rate buzzing for prey capture, but that bats capture prey after and toothed whales during buzzing. Bats face a much higher absorption and lower impedance in air by which they have prey detection distances that are between 1 and 2 orders of magnitude shorter than those of toothed whales while moving forward at speeds that are 2-3 times higher. That implies that bats have little time and hence potential for prey discrimination in the wild whereas toothed whales have many seconds between detection/discrimination and the time of capture.

10:00

3aAB6. Shandong University—Virginia Tech Biosonar Research in China. Rolf Müller (ME Dept., Virginia Tech, 150 Slayton Ave., Danville, VA 25240 & School of Physics, Shandong Univ., Shanda Nanlu 27, Jinan 250100, China, rolf.mueller@vt.edu)

China is home to a diverse and in many places abundant bat fauna. Among the most diverse and prominent bat families in the country are the horseshoe bats (*Rhinolophidae*) which also have one of the most specialized and capable sonar systems found in nature. Biosonar research at the Shandong University - Virginia Tech International Laboratory seeks to understand the capabilities and the diverse adaptations in the sonar systems of Chinese bats. A common thread of this research is the analysis of the natural variability in biosonar solutions. For example, the International Laboratory has been compiling a digital database of the noseleaves and outer ear shapes of bats from China and neighboring regions. Current research explores biological adaptation pattern in these shapes and their acoustic properties. Other research explores sound production across different species of Chinese bats. The International Laboratory also conducts behavioral experiments, in particular with horseshoe bats. This research has characterized novel dynamic features on the emission as well as on the reception side of the horseshoe bats' biosonar system. The International Laboratory collaborates with a sister lab at Virginia Tech (GLOBES) on the engineering analysis of the biosonar properties it discovers as well as the development of bioinspired devices.

Contributed Papers

10:20

3aAB7. Effects of water levels on distribution patterns of the Yangtze finless porpoises in Poyang and Dongting Lakes, China. Kexiong Wang, Lijun Dong (Key Laboratory of Aquatic Biodiversity and Conservation of the Chinese Academy of Sciences; Institute of Hydrobiology of Chinese Academy of Sciences, Wuhan 430072, China, wangk@ihb.ac.cn), Tomonari Akamatsu (National Research Institute of Fisheries Engineering, Fisheries Research Agency, Kamisu, Hasaki, Kashima, Ibaraki 314-0408, Japan), Satoko Kimura (Graduate School of Informatics, Kyoto University, Kyoto 606-8501, Japan), Shiyong Wang, Zhigang Mei, Songhai Li, and Ding Wang (Key Laboratory of Aquatic Biodiversity and Conservation of the Chinese Academy of Sciences; Institute of Hydrobiology of Chinese Academy of Sciences, Wuhan 430072, China)

Two large freshwater lakes in China (Poyang and Dongting Lakes) are crucial habitats of the Yangtze finless porpoises. The lakes are confronted with threatens from low water levels. For evaluating possible impacts of low water levels on the porpoises in the lakes, the distribution patterns of the animals in the lakes were monitored in different seasons by using a boat-towing A-tag array from 2009 through 2011. The survey routes were almost same among different seasons. The acoustical detection number (i.e. encounter number) of porpoise in every 3-km section was calculated by counting the bearing angle traces of the sonar sources recorded by the array. The numbers in the same section were compared between high and low water level periods. Results indicated that porpoises appeared to congregate in deep water areas in low water level periods, while they tended to disperse toward the near shore waters in high water level periods. The results suggest concentration of individuals during low water level period. The variations of distribution patterns in different water level periods remind us that protection efforts should be focused on different areas according to the changes of water levels in different seasons.

10:40–11:00 Break

11:00

3aAB8. Large reconfigurable microphone array for transmit beam pattern measurements of echolocating bats. Jason E. Gaudette (Ctr. for Biomedical Eng., Brown U., 171 Meeting Street, Providence, RI 02912, jason_gaudette@brown.edu), Laura N. Kloepper (Dept. of Zoology, U. of Hawaii, Honolulu, HI 96822), and James A. Simmons (Dept. of Neuroscience, Brown U., 89 Waterman Street, Providence, RI 02912)

Measurements of the transmit beam patterns in bats have previously been limited to a single cross-sectional plane or averaged over multiple inflight approaches with sparse microphone arrays. No high-resolution measurements have been published to date of individual transmitted beams jointly in azimuth and elevation. Toward this goal, a high density microphone array was designed and constructed using low-cost ultrasonic microphones and custom electronic circuitry. The planar array is 1.83 meters wide by 1.42 meters tall with sensors positioned on a 2.54 cm square grid. The system can record up to 228 channels simultaneously at a 500 kHz sampling rate. Big brown bats (*Eptesicus fuscus*) were trained to echolocate pairs of virtual targets in a two-alternative forced choice discrimination task while their signals were being recorded by the array. Visualizations of the beam patterns during the task will be presented along with some advanced signal processing techniques used in the analysis. [funded by ONR and NUWC, Division Newport]

11:20

3aAB9. Detection on the presence and frequency use pattern of cetacean tonal sound. Tzu-Hao Lin (Institute of Ecology and Evolutionary Biology, National Taiwan University, No. 1, Sec. 4, Roosevelt Road, Taipei 10617, Taiwan, schonkopf@gmail.com), Hsiang-Chih Chan, Chi-Fang Chen (Department of Engineering Science and Ocean Engineering, National Taiwan University, No. 1, Sec. 4, Roosevelt Road, Taipei 10617, Taiwan), Tomonari Akamatsu (National Research Institute of Fisheries Engineering, Fisheries Research Agency, 7620-7 Hasaki, Kamisu, Ibaraki 314-0408, Japan), and Lien-Siang Chou (Institute of Ecology and Evolutionary Biology, National Taiwan University, No. 1, Sec. 4, Roosevelt Road, Taipei 10617, Taiwan)

Passive acoustic monitoring (PAM) had already been proved to assess the presence of many cetacean species successfully. However, the continuous recording makes the manual data analysis difficult. In the present study, an automatic detection algorithm was developed for tonal sounds produced by Indo-Pacific humpback dolphins. The algorithm included a tonal sound detector which found the local spectral peaks and sampled the dominant frequencies every 5.3 ms. A noise exclusion process was used to exclude the spectral peaks with wide -3 dB bandwidth. After filtering the isolated frequency points within specific duration and frequency range, the adopted frequencies of tonal sounds could be obtained. The result showed the algorithm had 70% correct detection and 2.8% false positive based on each 1 sec time bin in 10 field recordings. The first to third quartile of adopted frequencies showed significant difference with those extracted manually, but the differences were only 245-489 Hz in average. The current algorithm performed considerably faster than real time. In the future, it can be applied as a first step in a real-time monitoring.

11:40

3aAB10. Expression of the gap junction protein connexin-36 in the adult big brown bat cochlear nucleus may be involved in temporally precise biosonar processing. Alyssa Wheeler, Carolina Veltri (Brown University, Alyssa_Wheeler@brown.edu), Victoria Flores, Andrea Simmons, and James Simmons (Brown University)

The big brown bat uses biosonar to orient, navigate, and forage. Successful prey capture requires sonar emissions and returning echoes to be encoded and compared in the central auditory system with precision. Behavioral experiments show that E. fuscus can discriminate echo delay, relative timing of harmonics, and echo phase on the order of 3μs or less. The neurobiological specializations underlying this perceptual acuity remain elusive.

New evidence shows that principal cells in the bat's cochlear nucleus (CN) express the neuronal gap junction protein connexin-36. During early postnatal development, connexin-36 expression is seen throughout the AVCN and PVCN of both big brown bats and mice. In adult bats, but not mice, cx36 expression is restricted to a specific population of cells in the ventral AVCN. The retention of cx36 in a discrete population of cells in the adult bat CN suggests that electrical transmission could be involved in processing echolocation sounds with temporal accuracy. Support: NSF grant 0843522 and ONR grant N00014-09-1-0691 to James Simmons.

12:00

3aAB11. Dolphin echolocation—synthetic aperture or "raster scanning"? Matthias Hoffmann-Kuhnt, Mandar Chitre (National University of Singapore, 18 Kent Ridge Road, Singapore 119227, tmsmh@nus.edu.sg), Eszter Mátrai (Ocean Park Hong Kong, Aberdeen Hong Kong SAR, China), Kelvin Yeo (National University of Singapore, 18 Kent Ridge Road, Singapore 119227), and Jason Lee (Ocean Park Hong Kong, Aberdeen Hong Kong SAR, China)

A bottlenose dolphin performing a cross-modal matching-to-sample task was stationed on a biteplate while echolocating on a sample object concealed in an opaque box. This procedure prevented the animal from gaining different aspects of the stimulus. Despite these restrictions on his location the dolphin was still able to recognize the object and successfully perform a match. The echolocation signals emitted by the dolphin were recorded with a rectangular array of 16 hydrophones mounted on a frame and placed between the dolphin on the biteplate and the object inside the box. A custom high-frequency data acquisition system recorded the signals at 500 kS per second and also collected synchronized video from several locations around the animal. The collected data was filtered and processed. The results presented here show that the dolphin was scanning the object and steering his echolocation beam without moving his head thus avoiding the possible acoustic clutter from multiple reflections from the object. The acquired data was also analyzed for the backscattered echo from the object.

12:20

3aAB12. A numerical study of the role of the noseleaf in the Egyptian slit-faced bat. Qiao Zhuang (Shandong Jianzhu University, 250101, zhuangqiao@sdjzu.edu.cn), Xiaomin Wang, and Mingxuan Li (Institute of Acoustics, Chinese Academy of Sciences, 100190)

Around 300 bat species are known to emit their ultrasonic biosonar pulses through the nostrils. This nasal emission coincides with the presence of intricately shaped baffle structures surrounding the nostrils. Some prior experimental evidence indicates that these "noseleaves" have an effect on the shape of the animals' radiation patterns. Here, a numerical acoustical analysis of the noseleaf of a slit-faced bat species is presented to show that all three distinctive parts of its noseleaf ("pit", "upper leaf", "lower leaf") have an effect on the acoustic near field as well as on the directivity pattern. In their effects on the near field, the noseleaf parts showed a tendency toward spatial partitioning with the effects due to each part dominating a certain region. However, interactions between the acoustic effects of the parts were also evident, most notably, a synergism between the frequency-dependent effects of two parts of the noseleaf ("pit", "lower leaf") to produce an even stronger frequency selectivity. supported by the Natural Science Foundation of China (Project No. 10974222), and the China Postdoctoral Science Foundation funded project (No. 20090450599). Bat specimen provided by the School of Physics, Shandong University, China.

WEDNESDAY MORNING, 16 MAY 2012 S428, 9:00 A.M. TO 10:40 A.M.

Session 3aBAa

Biomedical Acoustics: Ultrasound Imaging and Therapy (Poster Session)

Alfred Yu, Cochair
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Dong guk Paeng, Cochair
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Contributed Papers

All posters will be on display from 9:00 a.m. to 10:40 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 9:50 a.m. and contributors of even-numbered papers will be at their posters from 9:50 a.m. to 10:40 a.m.

3aBAa1. A preliminary study of ultrasound blood imaging in the common carotid artery of stroke patients. Tae-Hoon Bok, Qi Kong (Jeju National University, Jeju-si, Jeju Special Self-Governing Province, Rep. Korea, bth012@jejunu.ac.kr), Yun Hee Oh, Jang Jin Lee, Joong Goo Kim, Jay Chol Choi (Jeju National University Hospital, Jeju-si, Jeju Special Self-Governing Province, Rep. Korea), and Dong-Guk Paeng (Jeju National University, Jeju-si, Jeju Special Self-Governing Province, Rep. Korea)

Blood echogenicity is changed by red blood cell aggregation due to hemodynamic and hematological factors depending on a person. A stroke is known as a cerebrovascular accident due to lack of the blood flow. Hence, an ultrasound blood imaging could be the preliminary diagnosis for the stroke patient. In this paper, ultrasound images were acquired from the common carotid artery of stroke patients and the control group by the ultrasound imaging system (Voluson e, GE Healthcare, USA). The numbers of subject were 6 stroke patients and 5 healthy people for the control group, and their ages were 67 ± 17 and 68 ± 3 years old, respectively. The average of blood echogenicity of the stroke patients (54 ± 8) was lower than that of the control group (96 ± 8). The amplitudes of the cycle variation of blood echogenicity were similar for both of the stroke patient (18 ± 6) and the control group (23 ± 6). The preliminary experimental results showed the statistical difference of blood echogenicity between the stroke patients and the control group, and the data would be continuously collected from more volunteers (~20 people for each group) and discuss the data in the conference. [Work supported by NRF-2011-0017984.]

3aBAa2. Multi-dimensional real-time blood flow velocity field measurement in elastic vascular phantoms using ultrasonic particle image velocimetry technique. Ruibo Song, Ming Qian, Lili Niu, Qiaofeng Jin, and Hairong Zheng (Shenzhen Institutes of Advanced Technology, Chinese Academy of Sciences, 1068 Xueyuan Avenue, Shenzhen University Town, Shenzhen, P.R. China, rb.song@siat.ac.cn)

The experiment was implemented to measure the multi-component blood flow velocity in elastic vascular phantoms. Poly (vinyl alcohol) cryogel, PVA-C, is presented as a tissue-mimicking material, suitable for application in ultrasound imaging. The hardness of PVA-C changed with the number of freeze-thaw cycles. Two kinds of PVA phantoms with different shape were made in our experiment. Mechanical tests were performed on cylindrical samples with a pressure column, and Young's modulus were obtained varying from 50kPa to 320kPa depending on the number of freeze-thaw cycles (one to eight).The phantoms of tubular body with different elasticity were fixed in the fluid system. The information of flow field in the vascular phantom was obtained using the newly developed ultrasound velocimetry technique (echo particle image velocimetry). The flow rate and the pressure at the upstream of the vascular phantom are respectively measured by the Flow Meter and the Pressure Transmitter.

3aBAa3. Blood clot characterization by ultrasound Nakagami imaging. Po-Hsiang Tsui, Chieh-Ming Hung, and Chung-Hsin Hsu (Chang Gung University, tsuiph@mail.cgu.edu.tw)

Detection of blood clotting by ultrasound has been widely explored. Many ultrasonic quantitative parameters have been demonstrated to have the ability to characterize clot. This study proposes the ultrasound Nakagami image as a tool to visually characterize the properties of blood clot. Whole blood samples with a hematocrit of 40% were made. A 10-mL whole blood was placed in a tube and 1 mL of 0.2 mol/L CaCl2 solution was added to induce clotting. A 35-MHz focused transducer was used to scan the tube filled with blood before and after blood clot formation for Nakagami imaging. The results showed that whole blood and clot have similar textures in their B-scans, but the Nakagami image behaved well in distinguishing between blood and clot. The Nakagami image before clot formation is based on red-blue-interlaced shading, corresponding to Rayleigh distribution (the Nakagami parameter m approaches 1). However, the clot has more blue shadings in the Nakagami image, representing that the backscattered statistics of blood clot tends to be pre-Rayleigh distributed ($m < 1$). This study suggests that the Nakagami image may be used to characterize blood and clot for clinical studies and diagnosis.

3aBAa4. Observation of blood echogenicity variation in rat arteries using high-frequency ultrasound. Kweon-Ho Nam, Eunseop Yeom, Sang Joon Lee (Center for Biofluid and Biomimic Research, Department of Mechanical Engineering, Pohang University of Science and Technology, San 31, Hyoja-dong, Namgu, Pohang 790-784, Korea, kwonho@gmail.com), and Dong-Guk Paeng (Department of Ocean System Engineering, Jeju National University, Ara 1-dong, 102 Jejudaehakno, Jeju 690-756, Korea)

Previous studies have demonstrated that blood echogenicity is highly variable depending on blood flow velocity and pulsatility mainly due to the variation of red blood cell (RBC) aggregation. However, most of the studies were performed in a mock-flow loop using porcine blood and the results were not fully validated in vivo. Rat was rarely used for investigation of RBC aggregation due to its low aggregation tendency. But high-frequency ultrasound may detect the RBC aggregation from rat blood. The purpose of the present study is to investigate the cyclic and radial variation of blood echogenicty in rat arteries using a high-frequency ultrasound system with a 40-MHz scanner. B-mode images of blood were acquired from various arteries, including carotid artery and abdominal aorta. Blood echogenicity increased at systole and decreased at diastole. The cyclic variation was larger at the vessel center than near the vessel wall. The central hypoechoic zone ('black hole' phenomenon) was observed in carotid arteries. The experimental results from rat arteries in vivo corroborate the previous observations of cyclic and radial variations of blood echogenicity from porcine blood in mock-flow loops. Acknowledgments: This work was supported by the Korea Research Foundation Grant funded by the Korean Government (NRF-2011-0017984).

3aBAa5. Measurement of brain tissue motion using extended autocorrelation strain estimator. Redouane Ternifi, Melouka Elkateb Hachemi Amar, and Jean-pierre Remenieras (UMR INSERM CNRS U930, 10 Bd Tonnelle, 37032 Tours, France, redouane.ternifi@etu.univ-tours.fr)

Pulsatile motion of brain parenchyma results from cardiac and breathing cycles. In this study, transient motion of brain tissue was estimated using an Aixplorer\textregistered imaging system allowing an ultrafast 2D acquisition mode. The strain was computed directly from the ultrafast IQ complex data using the extended autocorrelation strain estimator (EASE), which provides great SNRs regardless of depth. The EASE first evaluates the autocorrelation function at each depth over a set of successive IQ pairs. This estimates the mean change in phase over time, which is proportional to the velocity. A second autocorrelation is evaluated on the results of the first autocorrelation. This estimates the mean change in phase over depth, which is proportional to the strain rate. The developed algorithm was first validated on "in vivo" data acquired at 7.5MHz from the carotid. Tissue velocity and strain rate were estimated on artery wall and adjacent regions. The estimated displacement velocity and displacement of the wall artery were 2.5 cm/s, and 150 μm respectively. The displacement velocity and displacement of the region near the surface were 1 cm/s and 30 μm respectively.

3aBAa6. Body surface scanner for the abdominal sound speed tomographic imaging. Akira Yamada, Kensuke Sasaki (Tokyo Univ. A&T, Koganei, Tokyo 184-8588, Japan, yamada@cc.tuat.ac.jp), and Toshihiko Yokoyama (Seiko Epson, Shiojiri, Nagano 399-0785, Japan)

The ultrasound tomography has been studied for the reconstruction of the abdominal sound speed image to measure the visceral fat area. The method is based on the travel time observations of the sound waves transmitted through the abdominal medium. In the present study, aiming to realize the method as a clinically available equipment, body surface scanner machinery was developed keeping good contact between a transducer and a human abdominal body surface. To-and-fro movement scanning equipment including the attachment of the elastic coupling gel hemisphere in front of the transducer surface was installed. The optimum pushing status was controlled by monitoring the variation of the received sound wave amplitude. They were in stable over the wide pushing distance region, regardless of the contact surface angle. It was shown that measured precision of the travel time were good enough to discriminate the difference of the sound speed between the fat and protein regions in the human abdominal region.

3aBAa7. Development of an ultrasound beamforming research platform based on SonixRP system. Xin Chen (School of Medicine, Shenzhen University, chenxin@szu.edu.cn), Ting Zhou, Siping Chen, and Tianfu Wang

The raw pre-beamforming data are necessary for the study of ultrasound imaging beamforming algorithm. However, the raw data are not accessible for most of the commercial ultrasound scanner. Therefore, most researches have to depend on software simulation. The purpose of this paper was to develop an open platform for the research of ultrasound beamforming. This platform utilized the SonixRP system to obtain the raw pre-beamforming data with great flexibility. The essential imaging parameters and scan sequence can be defined through a user-friendly GUI. Furthermore, a new adaptive beamforming algorithm was proposed and verified on the platform. The results showed that the algorithm can improve the image quality with better enhancement and lateral resolution while compared the conventional DAS algorithm. Acknowledgements: This work is supported by the National Natural Science Foundation of China (81000637, 61031003).

3aBAa8. Influence of temperature field produced by phase aberration in HIFU. Zhenbo Liu (Institute of Acoustics, Key Laboratory of Modern Acoustics (Ministry of Education), Nanjing University, Nanjing 210093; Nanjing Normal University, 210097, liuzb@njnu.edu.cn), Tingbo Fan, Xiasheng Guo, and Dong Zhang (Institute of Acoustics, Key Laboratory of Modern Acoustics (Ministry of Education), Nanjing University, Nanjing 210093)

High Intensity Focused Ultrasound (HIFU) is a noninvasive treatment in the field of cancer therapy. The principle of this technique is to raise the tissue temperature to relatively high values and cause thermal coagulation and ablation of cells by using tissue penetrable, strong directional and easily focused ultrasound. The phase aberration introduced by tissue inhomogeneity affects the tissue temperature evidently on the focus. This article studies the influence of temperature field debased by phase aberration theoretically using the 3D KZK (Khokhlov-Zabolotaskaya- Kuznetsov) combined phase aberration screen models and simulates the temperature field by solving Pennes equation. The peak and the size of the temperature with various phase aberration will be compare to homogeneity. In order to prove the theoretical results a series of in vitro experiments will be execute.

3aBAa9. The temperature dependent thermal properties of ex vivo porcine liver tissue heated from 20°C to 90°C. Min Joo Choi, Sitaramanjaneya Reddy Guntur (Jeju National University, mjchoi@jejunu.ac.kr), Kang IL Lee (Kangwon National University), Dong Guk Paeng (Jeju National University), and Andrew Coleman (Guy's & St Thomas'Hospital)

Thermotherapy uses a heat source which raises temperatures in a target tissue, and the temperature rise depends on the thermal properties of a tissue. Little is known about the temperature dependent thermal properties of a tissue, which prevents an accurate prediction of the temperature distribution of a target tissue that is undergoing thermotherapy. The present study reports the key thermal parameters (specific heat capacity, thermal conductivity and heat diffusivity) that was measured on ex-vivo porcine liver while being heated from 20° C to 90° C. The results show that all the thermal parameters resulted in the plots with asymmetrical quasi-parabolic curves with temperature, being convex downward with their minima at the turning temperature of 35-40° C. The largest change was observed for a thermal conductivity, which decreased by 9.6% from its initial value (at 20° C) at the turning temperature (35° C) and rose by 45% at 90° C from its minimum (at 35° C). The minima were 3.567 mJ/(m3.K) regarding the specific heat capacity, 0.520 W/(m.K) regarding the thermal conductivity, and 0.141 mm^2/s regarding the thermal diffusivity. The minimum at the turning temperature was unique and is suggested to be taken as a characteristic value of the thermal parameter of the tissue. The study indicates that the key thermal parameters varied largely with temperature, which resulted in having a substantial influence on the temperature distribution of the tissue undergoing thermotherapy. Keywords: thermal properties, ex-vivo porcine liver, temperature, specific heat capacity, thermal conductivity, thermal diffusivity, thermotherapy

3aBAa10. An optically transparent tissue mimicking phantom for monitoring the thermal lesion produced by high intensity focused ultrasound. Min Joo Choi, Sitaramanjaneya Reddy Guntur (Jeju National University, mjchoi@jejunu.ac.kr), Kang IL Lee (Kangwon National University), Dong Guk Paeng (Jeju National University), and Andrew Coleman (Guy's & St Thomas'Hospital)

An optically transparent tissue mimicking (TM) phantom whose acoustic properties are close to those of tissue was constructed for visualizing therapeutic effects by high intensity focused ultrasound (HIFU). The TM phantom was designed to improve a prevalent polyacrylamide hydrogel (PAG) which attenuates ultrasound far less than a tissue and does not scatter ultrasound unlike any other tissue. A modified recipe has been proposed in the study by adding scattering glass beads with diameters of 40 ~ 80 μm (0.002 % in w/v) and by raising the concentration of acrylamide (30% in v/v). The constructed TM PAG has an acoustic impedance of 1.66 Mrayls, a speed of sound of 1,573$\pm$5 m/s, an attenuation coefficient of 0.53$\pm$0.04 dB cm-1MHz-1, a backscattering coefficient of 0.23$\times$10-3 cm-1sr-1 MHz-1 and a nonlinear parameter (B/A) of 5.5$\pm$0.2. These parameters are close to those of liver. The thermal and optical properties are almost the same as the prevalent PAG. The TM PAG was tested to visualize the thermal lesions by HIFU and the characteristic features were contrasted with those of the prevalent PAG. In conclusion, the proposed TM PAG acoustically mimics tissue far better than the prevalent PAG and would be expected to be used in assuring if a clinical HIFU device could produce the thermal lesion as planned. Keywords: tissue mimicking phantom, high intensity focused ultrasound (HIFU), ultrasound, polyacrylamide hydrogel (PAG), monitoring, thermal lesion

3aBAa11. Ultrasonic standing wave patterns in a petri-dish. Min Joo Choi, Gwansuk Kang (Jeju National University, mjchoi@jejunu.ac.kr), Tetsuya Kodama (Tohoku University), and Andrew J Coleman (Guy's & St Thomas'Hospital)

A standing wave is well developed in a petri-dish used for cell culture, which is subject to the height of the culture fluid. A simple acoustic theory states that the pressure at the bottom of the petri-dish varies from 0 to its maximum while the height varies over a half wave length of the driving ultrasound. This suggests that the standing wave pattern should be taken into account when the cell line in a petri-dish is exposed to ultrasound. The present study has experimentally verified the theoretical prediction using the pressure sensitive film attached to the bottom of the petri-dish, when water, instead of the culture fluid, was contained in the dish. The driving field was made of 1 MHz transducer (V316, Panametrics, USA). Gradual destructuring of the standing wave patterns was observed as the driving power increased and, thus, the surface of water was becoming more fluctuating. Keywords, standing wave, petri-dish, cell, culture fluid, monitoring

3aBAa12. High-intensity focused ultrasound heating of large tissue region enhanced by cavitation bubbles at multiple focal spots. Shin Yoshizawa, Kotaro Nakamura, Ayumu Asai, Jun Yasuda, and Shin-ichiro Umemura (Tohoku University, 6-6-05 Aoba, Aramaki, Aoba-ku, Sendai 980-8579, Japan, syoshi@ecei.tohoku.ac.jp)

High-intensity focused ultrasound (HIFU) causes selective tissue necrosis through heating and is used for a noninvasive treatment of cancer therapy. However, it has a problem of a long treatment time for a large tumor. To improve the throughput of the treatment, the development of a highly efficient method is needed. It is known that cavitation bubbles enhance the heating effect of ultrasound during ultrasonic irradiation because the increase in the energy dissipation is caused by the volumetric oscillation of cavitation bubbles. In this study, cavitation bubbles were generated at multiple spots by changing focal position of high-intensity ultrasound. Immediately after generating the bubbles, the bubbles were exposed to a wide-focused ultrasound which covers all the cavitation sites for the cavitation-enhanced heating in a large region. The behavior of the cavitation bubbles at multiple spots in a tissue-mimicking gel was observed by high-speed photography, and the coagulation performance of the developed sequence was confirmed with an experiment using excised tissue. The results showed high efficacy of the proposed method for to coagulate a large tissue region.

3aBAa13. Efficient large-scale ultrasound simulation using the k-space pseudospectral method. Bradley E. Treeby (Research School of Engineering, College of Engineering and Computer Science, The Australian National University, Canberra ACT 0200, Australia, bradley.treeby@anu.edu.au), Jiri Jaros (Research School of Computer Science, College of Engineering and Computer Science, The Australian National University, Canberra ACT 0200, Australia), Ben T. Cox (Department of Medical Physics and Bioengineering, University College London, Gower Street, London WC1E 6BT, UK), and Alistair P. Rendell (Research School of Computer Science, College of Engineering and Computer Science, The Australian National University, Canberra ACT 0200, Australia)

Computational acoustics offers a powerful tool for investigating the interaction between ultrasound waves and the human body. However, in many common ultrasound settings, performing realistic simulations is computationally difficult due to the large size of the tissue volume compared to the size of the acoustic wavelength. This is particularly true in the case of high-intensity focused ultrasound where large diameter transducers are used to tightly focus ultrasound waves, often deep within the tissue. Here, an efficient model for large-scale ultrasound simulation is presented. The model is based on coupled first-order acoustic equations valid for nonlinear wave propagation in heterogeneous media with power law absorption. The equations are discretized using the k-space pseudospectral method and encoded using advanced programming techniques for parallel computer architectures. This allows the efficient simulation of nonlinear ultrasound propagation in three-dimensions over hundreds of wavelengths. Applications to both diagnostic and therapeutic ultrasound are discussed, and the results from several simulation examples are presented.

3aBAa14. Optically transparent gel for experimentally mimicking cavitation enhanced ultrasonic heating of tissue. Ayumu Asai, Tatsuya Moriyama, Shin Yoshizawa, and Shin-ichiro Umemura (Tohoku University, 6-6-05, Aramaki Aoba-ku Sendai, Miyagi, a.asai@ecei.tohoku.ac.jp)

High intensity focused ultrasound (HIFU) is a noninvasive method for cancer treatment. However, there is a problem of a long treatment time for treating a large volume. It is known that cavitation bubbles, generated by extremely high intensity ultrasound pulses, enhance the heating effect of HIFU. In order to investigate a cavitation-enhanced highly-efficient method of HIFU, an optically transparent gel with both ultrasonic absorption and cavitation threshold similar to biological tissue is being developed. Such a polyacrylamide (PAA) gel was successfully produced by controlling the concentrations of both acrylamide and albumin. The effect of cavitation bubbles enhancing the ultrasonic heating was measured using the gel by exposing it to HIFU. The effect was considered in the bio-heat transfer equation (BHTE) by increasing the ultrasonic absorption coefficient in the region of the cavitation, whose volume was determined by high-speed-camera observation. The absorption coefficient was calculated by fitting between the temperature rise curves at the focal point in the experiment and numerical simulation. The simulation using the obtained absorption coefficients of the gel with and without cavitation showed overall agreement with the experiment using the gel. The developed gel and method will be useful for further development of this HIFU method.

3aBAa15. Microbubble-enhanced high intensity focused ultrasound therapy: effect of exposure parameters on thermal lesion volume and temperature. Sonal Bhadane, Raffi Karshafian, and Jahan Tavakkoli (Department of Physics, Ryerson University, Toronto, ON M5B 2K3, Canada, sbhadance@ryerson.ca)

Microbubble agents have been shown to increase therapeutic effect of HIFU (high intensity focused ultrasound). Here, the effects of treatment parameters on lesion volume and temperature are investigated. Ex vivo tissue was treated with a 2 MHz HIFU beam in absence and presence of the Artenga^{TM} microbubbles at varying HIFU focal intensities (649-2316 W/cm^2), microbubble concentrations, and exposure durations (3-10 s). The temperature was measured at 1 mm from focus using a K-type thermocouple. Thermal lesion volume was measured based on an ellipsoid model. Microbubbles increased the lesion volume and peak temperature achieved with HIFU. At the intensity of 2316 W/cm^2, the lesion volume increased by 2-folds, and the peak temperature increased by 16° C with microbubbles. This effect depended on microbubble concentration, ultrasound intensity and exposure duration. Lower intensities and shorter time durations were required at higher microbubble concentrations to ablate the tissue. It was concluded that the efficacy of the HIFU therapy in combination with microbubbles can be controlled through ultrasound/microbubble exposure parameters

3aBAa16. Measurement of high intensity focused ultrasound fields using a combined measurement and SBE modeling approach. Tao Chen (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China; Jiangsu Province Medical Instrument Testing Institute, Nanjing 210012, China, 13584004956@139.com), Tingbo Fan (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China), Liyang Xia, Jimin Hu, Ru Liu (Jiangsu Province Medical Instrument Testing Institute, Nanjing 210012, China), and Dong Zhang (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China)

Acoustic characterization of high Intensity focused ultrasound (HIFU) is essential for its development in clinical treatment. In the present study, a combined measurement and modeling approach is proposed. At relative low amplitude excitation, acoustic measurement in water is performed to calibrate the transmitter parameters; then the acoustic fields of HIFU transmitter can be predicted based on the SBE model. To verify the validity of this approach, a 1 MHz HIFU transmitter with large aperture is utilized in the study, and the HIFU field is measured by a HFO-660 fiber optic hydrophone. This study is helpful for the accurate characterization of HIFU fields. [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 (10974093 and 11011130201), and the Fundamental Research Funds for the Central Universities (Grant Nos. 1103020402, 1116020410 and 1112020401)]

3aBAa17. Activities of the cavitation bubbles in the wake of a shock pressure pulse recorded by an optical fiber hydrophone. Min Joo Choi, Sung Chan Cho, Gwansuk Kang (Jeju National University, mjchoi@jejunu.ac.kr), and Andrew J. Coleman (Guy's & St Thomas' Hospital)

The shock pulse used in an extracorporeal shock wave treatment (ESWT or ESWL) has a large negative pressure ($<$ -5MPa) which can always produce acoustic cavitation. The resulting cavitation bubbles are known to play an important role in therapeutic effects, however, the bubble activities are not readily measurable yet. The present study considered a weird tail after the negative peak in the time history of pressure sensed by an optical fiber hydrophone which was usually abandoned in typical pressure field measurements. A shock pressure pulse in water causes change of mass density which modulates the optical refractive index. The change of the refractive index can be measured by the light reflection at the tip of the glass fiber submerged in water. The loss of water contact by cavitation bubbles at the fiber tip leads to an abnormal increase of high reflection which is clearly identified. This suggests that the weird tail of the hydrophone signal beyond the negative cycle of a shock pulse is closely related to the extent of the cavitation bubbles. This was experimentally validated in the shock wave field which was produced in water by a clinical ESWT system (ShineWave, HnT Medical, Republic of Korea) with an optical hydrophone (FOPH2000, RP Acoustics, Germany). Keywords: shock pressure pulse, ESWT, ESWL, cavitation, bubbles, optical fiber hydrophone

Session 3aBAb

Biomedical Acoustics: Ultrasound Enhanced Drug Delivery and Bone Quantification (Poster Session)

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Contributed Papers

All posters will be on display from 11:00 a.m. to 12:40 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 11:00 a.m. to 11:50 a.m. and contributors of even-numbered papers will be at their posters from 11:50 a.m. to 12:40 p.m.

3aBAb1. A novel formulation of lipid microbubbles coated with stearic acid modified polyethylenimine to enhance ultrasound-mediated gene delivery in vitro. Qiaofeng Jin, Zhiyong Wang, Fei Yan, Zhiting Deng, Lu Li, and Hairong Zheng (Shenzhen Institutes of Advanced Technology, Chinese Academy of Sciences, 1068 Xueyuan Avenue, Shenzhen University Town, Shenzhen, P.R. China, qf.jin@siat.ac.cn)

The aim of this study was to assess the properties of a new designed cationic microbubbles as gene carriers and the relative gene transfection efficacy with ultrasound triggered microbubble destruction. Polyethylenimine as a high efficient gene transfection agent has higher cell toxicity with molecular weight increasing. Stearic acid was used to modify branched polyethylenimine to change its hydrophilic properties so that it can be assembled onto the lipid shell of the microbubbles and simultaneously decrease its cell toxicity. Cationic microbubbles was prepared by encapsulating perfluoropropane into phospholipids and stearic acid modified polyethylenimine hybrid shell using mechanical vibration method. The mean, median size and zeta potential of the microbubbles were measured 1.84 ± 1.62um, 1.60um and 54mv respectively. Hoechst 33258 was used to stain the green fluorescent protein reporter plasmid which was charge-coupled to cationic microbubbles, and microbubbles was observed emitting blue light under a fluorescence microscope. About 4ug plasmid loaded by 10^8 microbubles that contain 5% mole ratio stearic acid modified polyethylenimine was measured by gel electrophoresis. A 1.25MHz single element transducer was used to mediate the gene transfection to MCF-7 cell by using the cationic microbubbles and enhancement of green fluorescent protein expression was observed.

3aBAb2. Paclitaxel-liposome loaded microbubbles for ultrasound-triggered drug delivery in vitro and in vivo. Fei Yan, Lu Li, Zhiting Deng, Qiaofeng Jin, and Hairong Zheng (Shenzhen Institute of Advanced Technology, Chinese Academy of Sciences, 1068 Xueyuan Avenue, Shenzhen University Town, Shenzhen, P.R. China, fei.yan@siat.ac.cn)

The aim of this work was to develop paclitaxel-liposome loaded microbubbles (PTX-loaded MBs) and to investigate the efficacy of chemotherapy through ultrasound-triggered drug delivery in vitro and in vivo. PTX-loaded liposomes were prepared by a minor modification of thin film hydration method and further conjugated to the surface of microbubbles through biotin-avidin linkage. The resulting payload-loaded MBs were characterized and applied to ultrasound assisted chemotherapy in breast cancer. Our results showed the MBs were able to achieve satisfactory drug encapsulation efficiency. Under ultrasound exposure, about 9.54-fold higher drug release and a significant improvement of cell uptake than that of loaded liposomes were observed. In addition, PTX-loaded MBs showed significantly greater tumor growth inhibition both in vitro and in vivo xenograft growth of breast tumor cells, compared with PTX-loaded liposomes and unloaded MBs. Drug distributions assay in various organs (heart, liver, spleen, lung, kidney and tumor) indicated about 3-fold higher PTX enriched in tumor. Histological examinations further demonstrated the tumor growth inhibition might contribute to increased apoptosis and reduced angiogenesis in tumor xenografts. In conclusion, the study indicated the PTX-loaded MBs significantly increased the anti-tumor efficacy and can be used as a potential chemotherapy approach for ultrasound assisted breast cancer treatment.

3aBAb3. Cytotoxicity enhancement of DOX-liposome-containing microbubbles combined with ultrasound to the multi-drug resistant cell. Zhiting Deng, Fei Yan, Lu Li, Qiaofeng Jin, Fei Li, and Hairong Zheng (Paul Lauterbur Center for Biomedical imaging, Institute of Biomedical and Health Engineering, Shenzhen Institutes of Advanced Technology, Chinese Academy of Sciences, Shenzhen, 518055, zt.deng@siat.ac.cn)

Multidrug resistance (MDR) in cancer cells is a significant obstacle to successful cancer therapy. Doxorubicin (DOX) is very active chemotherapeutic agent for the treatment of breast cancer and the efficacy of DOX is also restricted by multidrug resistance.Ultrasound (US) targeted destruction of drug loaded microbubbles (MBs) is gaining more and more attention as a promising strategy for locally drug delivery. In this article, through avidin–biotin binding of DOX-containing liposomes to the microbubbles, we developed DOX-liposome-containing microbubbles in order to investigate its effect of enhancing cellular uptake and cytotoxicity of DOX against drug-resistant cancer cells. The results demonstrated that treatment of cells with ultrasound and DOX-liposome-containing microbubbles caused a significant higher drug uptake in DOX-resistant MCF-7 cells, compared with control (DOX-liposome). More importantly, a significant enhancement of tumor growth inhibition against DOX-resistant MCF-7 cells was found when using the drug-liposome-containing microbubbles combined with ultrasound.-Compared with DOX-liposome treatment, the cytotoxicity effect was greatly enhanced from 21% to 60%. By further mechanism study, DOX-loaded microbubbles plus ultrasound induced significant apoptosis in MDR line of MCF-7 cells.

3aBAb4. The estimation of bone microstructure using characteristic refraction of fast wave. Katsunori Mizuno, Keisuke Yamashita, Bunri Fujita, and Mami Matsukawa (Doshisha University, 1-3 Tatara Miyakodani Kyotanabe City Kyoto, kmizuno0417@gmail.com)

Two-wave phenomenon, known as the wave propagation of fast and slow waves, has been observed in the cancellous bones when the ultrasonic wave propagates along the trabecular alignment. It is also reported that the acoustical parameters of the fast and slow waves related to not only bone

mass but also bone microstructure. In this study, the characteristic refraction of the fast wave was focused, and a method was proposed to estimate the bone microstructure. Cylindrical specimens of cancellous bone were obtained from a bovine femur. Using a conventional ultrasound pulse technique, the receiver was moved to investigate the ultrasonic fields after the propagation in the specimen. In addition, the structure of the specimen was estimated by using the X-ray micro CT. The fast wave showed the "apparent refraction" following the trabecular alignment in the specimen, which was not observed from the slow wave behavior. The ultrasonic field after the sample propagation changed reflecting the trabecular alignment, which gives us the information of microstructure of cancellous bone. Especially, the evaluation of fast wave propagation can be useful for the estimation of bone microstructure and will support the ultrasound bone assessment in the in vivo situation.

3aBAb5. Ultrasound propagation in trabecular bone: a numerical study of the influence of micro-cracks. Samuel Callé, Hélène Moreschi, and Marielle Defontaine (Université François-Rabelais - INSERM U930 - 10 bd Tonnellé - 37032 Tours cedex - France, samuel.calle@univ-tours.fr)

The accumulation of microdamage in trabecular bone tissue is suspected of being a predictive indicator of osteoporosis diagnosis. To quantify this microdamage, the Dynamic AcoustoElastic Testing (DAET) method measures the time of flight (TOF) and amplitude variations of transmitted ultrasound (US) pulses, while the bone sample is submitted to a low frequency pressure (opening/closing of microcracks). However, DAET is both sensitive to viscoelastic properties changes and microcracks density. To estimate the microcraks density contribution, a numerical approach is proposed, allowing simulations of different levels of microdamaged media. A 2D pseudo-spectral time domain numerical model was then developed to simulate linear wave propagation in heterogeneous solids including thermo-viscous attenuation. The influence of the microcracks number, size and orientation on the US TOF and amplitude was particularly investigated. Results are discussed and compared with experimental data extracted from DAET measurements in trabecular bone samples.

3aBAb6. Application of dual frequency ultrasound method in through transmission measurements. Janne Petri Karjalainen (University of Eastern Finland, P.O. Box 1627, FI-70211 Kuopio, Finland, janne.p.karjalainen@iki.fi), Michal Pakula (Institute of Environmental Mechanics and Applied Computer Science, Kazimierz Wielki University in Bydgoszcz, ul. Chodkiewicza 30 85064 Bydgoszcz Poland), Quentin Grimal (Universite Paris 6 - CNRS, 15 rue de l'Ecole de Medecine 75006 Paris, France), Juha Töyräs (Department of Clinical Neurophysiology, Kuopio University Hospital, and Department of Applied Physics, University of Eastern Finland, P.O. Box 1627, FI-70211 Kuopio, Finland), Jukka Sakari Jurvelin (University of Eastern Finland, P.O. Box 1627, FI-70211 Kuopio, Finland), and Pascal Laugier (Universite Paris 6 - CNRS, 15 rue de l'Ecole de Medecine 75006 Paris, France)

Soft tissue layers overlying bones, with unknown thickness, can produce significant errors to bone quantitative ultrasound measurements. In this study dual frequency ultrasound (DFUS) technique, developed originally for pulse-echo measurements, is applied and evaluated in a configuration typical to clinical through-transmission measurement. A mathematical algorithm is presented for determination of soft tissue composition, i.e. amount of lean and fat tissue, and correction of typical clinical parameters such as broadband ultrasound attenuation (BUA) and speed of sound (SOS). Ultrasound soft tissue phantoms mimicking lean and fat tissues, were tested in five different configurations by varying the composition (0-100% of fat). Different configurations were built using 10mm of fat or lean and 30mm of fat or lean, and total thickness of soft tissue constructs varied from 20mm to 40mm. Using through-transmission measurements at 0.5 MHz center frequency the thickness of soft tissue layers could be determined using DFUS technique (low and high frequency band 0.4-0.45 and 0.55-0.6 MHz, respectively). The average absolute error of fat/muscle layer thickness was 2.2mm (SD=1.3mm). The results suggest that DFUS technique may be used in through-transmission to assess the soft tissue content over/underlying the bone, and subsequently to reduce soft tissue induced errors in quantitative ultrasound measurements.

3aBAb7. Age-dependence and variation of elastic properties and cortical porosity in human femoral neck and shaft. Markus Kalle Henrik Malo (Department of Applied Physics, University of Eastern Finland, Kuopio, Finland, markus.malo@uef.fi), Daniel Rohrbach (Julius Wolff Institut & Berlin-Brandenburg School for Regenerative Therapies, Charité - Universitätsmedizin Berlin Augustenburger Platz 1, 13353 Berlin, Germany), Hanna Isaksson (Division of Solid Mechanics, Lund University, Lund, Sweden), Juha Töyräs (Department of Applied Physics, University of Eastern Finland, Kuopio, Finland), Jukka Sakari Jurvelin (Department of Applied Physics, University of Eastern Finland, Kuopio, Finland), and Kay Raum (Julius Wolff Institut & Berlin-Brandenburg School for Regenerative Therapies, Charité - Universitätsmedizin Berlin Augustenburger Platz 1, 13353 Berlin, Germany)

Tissue level structure and mechanical properties are important determinants of macroscopic bone strength. Scanning acoustic microscopy (SAM) provides maps of local elasticity in bone. The present aims were to evaluate spatial variations of elastic and structural properties in human femoral neck and shaft, and their variations with age. A total of 48 transverse cross-sectional bone samples were obtained from femoral neck (Fn) and proximal shaft (Ps) of 24 cadavers (21 men, 4 women; age 49.2 ± 16.3 years). Samples were measured with a custom SAM using a 50-MHz ultrasound transducer. Distributions of the elastic coefficient c_{33} of cortical (Ct) and trabecular (Tr) tissues and microstructure of cortex (cortical thickness Ct.Th and porosity Ct.Po) were investigated (ANOVA). Significant variations in c_{33} were observed with respect to tissue type (trabecular < cortical), location (Ct.Ps=32.3GPa > Ct.Fn=29.9GPa > Tr.Ps=28.9GPa > Tr.Fn=26.9GPa, and cadaver age. Regional variations in Ct.Po were found in the neck (inferior 6.7%; superior 13.2%; anterior 11.3%; posterior 9.1%) and in the shaft (medial 9.3%; lateral 8.0%; anterior 8.3%; posterior 12.9%). This study provides a comprehensive database of age-dependent spatial distributions of microstructural and microelastic properties in the femoral neck and shaft.

3aBAb8. Numerical investigation of ultrasound reflection properties in cancellous bone. Atsushi Hosokawa (Akashi National College of Technology, hosokawa@akashi.ac.jp)

Bone is composed of two components of cancellous and cortical bones, and cancellous bone is generally surrounded by cortical bone. Therefore, ultrasound waves propagating in bone can reflect at the boundary between cancellous and cortical bones. In this study, the ultrasound reflection properties in cancellous bone were numerically simulated using a finite-difference time-domain (FDTD) method with microcomputed tomographic (μCT) models of the bone. From the simulated results, it was investigated how the ultrasound waves propagating in cancellous bone could reflect at the boundary. The reflected waveform at normal incidence to the boundary was calculated using the numerical model comprised of two layers of cancellous and cortical bones, and only the reflection properties were derived by subtracting the waveform calculated using the model adopting the absorbing boundary condition instead of the cortical bone layer. In the case of the strong trabecular orientation parallel to the ultrasound propagation, the reflections of both fast and slow longitudinal waves could be observed.

3aBAb9. Analytical model of Dynamic AcoustoElastic Testing measurements in trabecular bone tissue: a rheological approach. Hélène Moreschi, Samuel Callé, Chloé Trarieux, and Marielle Defontaine (Université François-Rabelais INSERM U930 - 10 bd Tonnellé - 37032 Tours - France, helenemoreschi@gmail.com)

A dynamic acoustoelastic testing (DAET) method based on the interaction of two acoustical waves was developed to measure viscoelastic properties of trabecular bone tissue. While a sinusoidal low-frequency (LF) acoustic wave successively compresses and expands the medium, ultrasonic (US) pulses are generated to probe the uniformly stressed medium (quasi-static pressure, pLF). The US pulses velocity and amplitude (c, A) vary with respect to the medium stress level. These modulations (c(pLF)-c(0); A(pLF)-A(0)) are plotted as a function of the instantaneous LF pressure. From these rheograms, nonlinear elastic and viscous (dissipative) parameters, β' and β" respectively, are extracted. Since the US velocity is directly related to the medium viscoelastic modulus (M*), considering either the celerity variations or the modulus variations as a function of LF pressure is equivalent. In this perspective, we developed an analytical model of the variations of M*, derived from the Kelvin-Voigt approach (M* as a function of

3a WED. AM

pLF), introducing linear (M', M'') and nonlinear (β', β'') viscoelastic moduli. This model was first validated in water and Newtonian oils. In this paper we present preliminary results derived from measurements in the calcaneum, a highly inhomogeneous and porous trabecular bone tissue exhibiting hysteretic loops and asymmetry in rheograms.

3aBAb10. Influence of healing time on the echographic response of the bone-implant interface. Vincent Mathieu, Romain Vayron (CNRS, Université Paris-Est, Laboratoire Modélisation et Simulation Multi Echelle, UMR 8208 CNRS, 61 avenue du Général de Gaulle, 94010 Créteil, France, vincent.mathieu@u-pec.fr), Emmanuel Soffer, Fani Anagnostou (CNRS, Université Paris-Diderot, Laboratoire Bioingénierie et Biomécanique Ostéo Articulaire, UMR 7052 CNRS, 10 avenue de Verdun, 75010 Paris, France), and Guillaume Haiat (CNRS, Université Paris-Est, Laboratoire Modélisation et Simulation Multi Echelle, UMR 8208 CNRS, 61 avenue du Général de Gaulle, 94010 Créteil, France)

The study aims at investigating the effect of bone healing on the ultrasonic response of coin-shaped titanium implants. Sixteen implants were inserted on the tibiae of rabbits. Two groups of eight samples were considered, each group corresponding to a healing duration (7 or 13 weeks). After the sacrifice, the ultrasonic response of the bone-implant interface was measured in vitro at 15 MHz with a 2-D scanning device. The average value of the ratio r between the amplitudes of the echo of the bone-implant interface and of the water-implant interface was determined. The fraction of implant surface in contact with bone was measured by histomorphometry. The ultrasonic quantitative indicator r decreases significantly with healing time ($p = 2.10^{-4}$, from $r = 0.53$ to $r = 0.49$). Two phenomena are responsible for the decrease of the gap of acoustical impedance at the bone-implant interface: i) the increase of mineralization of newly formed bone tissue and ii) the increase of the bone-implant contact fraction (from 27 % to 69 %).

3aBAb11. Experimental investigation of a bone-conducted ultrasonic hearing system based on a DSP platform. Ziying Yu (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, yuziying@mail.ioa.ac.cn), Junxian Shen (State Key Laboratory of Brain and Cognitive Science, Institute of Biophysics, Chinese Academy of Sciences, Beijing, China), and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China)

Ultrasonic signals can be conducted and perceived by bones when a certain part of human skull is pressed. Experiments show that hearing-impaired people may be able to perceive ultrasound, distinguish different frequencies, or even recognize words after training. This finding is very promising for hearing-aid studies. In this paper, a novel bone-conducted ultrasonic (BCU) hearing system is developed. Both the software simulations and experiments on human subjects are carried out. The BCU hearing system is implemented on a DSP platform to achieve an appropriate modulation strategy. An ultrasonic vibrator is attached to the system, allowing the audible sound signals to be demodulated from the ultrasonic region via bone conduction. Different carrier frequencies and modulation algorithms are examined and validated in this platform. To evaluate system performance, perception tests are also conducted on the deaf and normal-hearing subjects. Experimental results show that the proposed system can operate as a flexible experimental platform suitable for BCU hearing studies.

3aBAb12. Estimation of the radiation force on implanted medical devices: a theoretical study. Samuel Callé (Université François-Rabelais INSERM U930 - 10 bd Tonnellé - 37032 Tours - France, samuel.calle@univ-tours.fr), Guillaume Ferin (Vermon - 180 rue du général Renault - BP 93813 - 37038 Tours cedex 1 - France), and Jean-Pierre Remenieras (Université François-Rabelais INSERM U930 - 10 bd Tonnellé - 37032 Tours - France)

Implantable medical devices, such as pacemaker or insulin pump, are more and more used to extend or improve the quality of life. The radiation force could actually be a solution to remotely control these devices in different applications (energy supply, drug delivery control...). The aim of this work is to quantify the amplitude and spatial repartition of the ultrasound radiation force which could be applied on such a system. Reflection of the incident acoustic beam at the interface (surface force) and attenuation of the US beam along the wave path (body force) both contribute to this force. In this study, these force contributions have been calculated in the case of a device implanted one centimeter under the skin. First, a heterogeneous axisymetric pseudo-spectral time domain method is used to estimate the ultrasound particle velocity and acoustic pressure (amplitude, spatial repartition) inside the device and at the interface. These parameters are then inserted in analytical expressions to precisely calculate the force applied on the implantable device. Surface and body forces contributions have been analyzed and the global force estimated for two configurations of 5 MHz transducers (plane and focused) and two types of device mechanical characteristics (semi-rigid and rigid).

Session 3aEAa

Engineering Acoustics: Acoustic Well Logging and Borehole Acoustics II

Xiuming Wang, Cochair
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Hailan Zhang, Cochair
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Invited Papers

8:20

3aEAa1. Low frequency broadband monopole transducer based on trilaminar bender bar structures. Dai Yuyu, Wang Xiuming, Xin Penglai, and Cong Jiansheng (Institute of Acoustics, Chinese Academy of Sciences, daiyuyu001@126.com)

A transducer which is produced by a skeleton and four Trilaminar bender bar vibration elements is presented in the paper. Four vibration elements formed a square array are excited by the same signal to realize monopole radiation. The influence of the skeleton is taken into account: flexural mode of vibration of the skeleton is used to extend the bandwidth. The transmitting voltage response of a modeling result shows that there are two peaks existed close to 1.8 kHz, which correspond to the 2nd order flexural mode of the Trilaminar bender bar and the 1st order flexural mode of the skeleton. The -3dB operational frequency range from 16.6kHz to 20.2kHz. The directivity patterns reflect that transducer can realize a monopole radiation in the horizontal plane before 13kHz.

8:40

3aEAa2. Design of multi-functional ultrasonic imaging logging tool. Zhifeng Sun (COSL, P.O. Box 232, Bejing, China, sunzf@cosl.com.cn), Ao Qiu, Wenliang Wang, Aihua Tao, Honghai Chen, and Xien Liu

By analyzing the measurement principle of the ultrasonic impulse method, a kind of Multi-functional ultrasonic imaging logging tool was designed and the implementation of its measurement technique was given. mainly contains the components of the tool, the realization of electronic sections and the measurement specifications. It can conduct borehole surface imaging in high resolution in the opened-hole. It can also conduct casing inspection and cement bonding evaluation in the cased-hole. The proposed tool was tested in the field of opened-hole and cased-hole. It shows that this tool can evaluate the geologic feature's informations, such as fracture, vug and bedding structure in the opened-hole. The casing thickness imaging, inner and outer diameter imaging and cement impedance imaging also can be measured in the cased-hole.

Contributed Papers

9:00

3aEAa3. Study on the discrimination of the second interface bonding conditions using characteristics of the first arrivals in cased boreholes. Xiumei Zhang (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, No. 21, 4th Northwestern Ring RD, Haidian District, Beijing 100190, P.R. China, zhangxiumei@mail.ioa.ac.cn), Xien Liu, Honghai Chen (Well-Tech R&D Institutes, COSL, Hebei 101149, P.R. China), and Weijun Lin (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, No. 21, 4th Northwestern Ring RD, Haidian District, Beijing 100190, P.R. China)

The evaluation of cementing bonding conditions influence the production of oil or gas in cased boreholes. Many researchers have focused their studies on identifying the cementing quality of the first and second interface, and provide lots of methods in evaluating the bonding conditions of the first interface, while for the second interface, remain a difficult problem till now. In this paper, the propagation velocity and amplitude of the first arrivals with different cement qualities, together with different channeling for the same cased hole are studied. The results show that for poor first interface bonding conditions, the velocity of the first arrival almost doesn't change with different channeling, while for poor second interface bonding conditions, the velocity varies greatly with different channeling. Specially, Time-Frequency Analysis on different cementing conditions are conducted, the analysis indicate that there is a relative large energy peak of the first arrivals for all poor first interface, while for poor second interface, which is not the case. As a result, the propagation velocity and the time frequency distributions rather than the amplitude of the first arrivals can evaluate the second interface bonding conditions.

9:20

3aEAa4. Radial detection depths of borehole Stoneley modes. Xiao He and Xiuming Wang (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, No. 21, 4th Northwestern Ring Rd, Beijing 100190, China, hex@mail.ioa.ac.cn)

Propagation features of Stoneley modes are widely used for measurements of reservoir permeability and rock anisotropy in acoustic logging. Through the finite difference algorithm, the radial detection depth of the Stoneley wave along a fluid-filled borehole is investigated. The radial variations of the displacement components Ur for Stoneley wavefronts are presented as the particle motions in that direction are controlled by the permeability and transverse shear modulus of the medium surrounding the

wellbore. It is shown that the Stoneley-wave energy reaches a peak value at the borehole wall and attenuates exponentially with the increasing radial depth in the formation. By comparison of reflected Stoneley-wave amplitudes from reflectors with various distances to the borehole wall, the prospecting depths of Stoneley modes can be confirmed. It is generally no more than 0.2 meters with the frequency range of acoustic logging. This result reveals that only a very shallow region around the wellbore can be detected from the Stoneley-wave responses.

9:40

3aEAa5. Processing dipole acoustic logging data to image fracture network in shale gas reservoirs. Zhuang Chunxi, Su Yuanda, and Tang Xiaoming (China University of Petroleum, Qingdao, Shandong 266555, zhuangchunxi@sina.com)

A recent advance in borehole remote acoustic reflection imaging is the utilization of a dipole acoustic system in a borehole to emit and receive elastic waves to and from a remote geologic reflector in formation. An important application of this new technique is the delineation of fracture network in shale gas reservoirs, as interest and activities in shale gas exploration increase in China. We develop a data processing procedure and implement it to handle routine processing of dipole acoustic logging data. The procedure takes into account the characteristics of the dipole data, such as frequency, dispersion, attenuation, recording length, and dipole source orientation, etc., to obtain an image of reflectors within 20~30 meters around the borehole. We have applied the technique to process dipole acoustic data from several wells drilled into gas reservoirs in China. The obtained images clearly identify major fracture network in the gas producing intervals of the reservoir, demonstrating the effectiveness of the imaging technique.

10:00

3aEAa6. Acoustic wave propagation in cracked porous rocks and application to interpreting acoustic log data in tight formations. Xiaoming Tang and Xuelian Chen (China University of Petroleum, Qingdao, Shandong 266555, tangxiam@yahoo.cn)

Rocks in earth's crust usually contain both pores and cracks. Typical examples include tight sandstone and shale rocks that have low porosity but contain abundant microcracks. By extending the classic Biot's poroelastic wave theory to include the effects of cracks, we obtain a unified elastic wave theory for porous rocks containing cracks, adding crack density and aspect ratio as two important parameters to the original theory. The new theory is applied to interpret acoustic velocity log data from tight sand and shale gas formation, whereas the classic Biot theory has difficulty in explaining such data. Because the flat- or narrow-shaped cracks can easily deform under acoustic wave excitation, the acoustic property of a cracked porous rock is quite different for different saturation conditions. This allows the new theory to correctly predict the trend of velocity variation with gas saturation in low-porosity rocks, providing a useful interpretation tool for acoustic logging in tight formations.

10:20

3aEAa7. Effects of eccentric acoustic source on the amplitude and arriving time of the first arrival in cased boreholes. Dehua Chen, Xiuming Wang, Hailan Zhang, and Weijun Lin (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-Huan-Xi Rd., Haidian District, Beijing 100190, China, chendh@mail.ioa.ac.cn)

The acoustic fields excited by eccentric acoustic source in cased boreholes bonded by cements with different densities are simulated numerically using 2.5-dimension (2.5D) finite-difference method (FDM). The effects of the source eccentricity on the amplitude and arriving time of the first arrival (FA) in the full waveform are investigated. The numerical results show that the amplitude of the FA will decrease quickly and its arriving time will go ahead as the source eccentricity increases. The change of the arriving time can be estimated approximatively using geometrical acoustics theory. As the eccentric distance of the acoustic source reaches 1/4 of the borehole radius, the amplitude of the FA will reduce to below 20% of the centered case. The quantitative varying trends of the amplitude and arriving time of the FA are the same both in the free pipe and in boned pipe. Therefore, during the cementing evaluation, the tool's eccentricity should be estimated through the arriving time of the first arrival, and then its amplitude should be corrected according the eccentricity. Otherwise, when the interface between the steel pipe and the cement is not boned well, the over-optimistic cementing evaluation could be led to because of the large reduction of the FA's amplitude aroused by eccentric acoustic source.

10:40–11:00 Break

11:00

3aEAa8. Phase spectrum correlation method to extracting anisotropic parameters from four-component dipole waveforms. Xiao-Hua Che, Rui-Jia Wang, and Wen-Xiao Qiao (State Key Laboratory of Petroleum Resources and Prospecting, China University of Petroleum, Beijing 102249, China, aclab@cup.edu.cn)

The anisotropy parameters extracted from 4-component dipole waveforms are always affected by dispersion of mode waves, the distribution of stress, the invasion zone and so on. It is impossible to extract the real anisotropy parameters in time domain process. In this paper, we develop a phase spectrum correlation method (PCM) to extract the anisotropic parameters in frequency domain. The PCM searches the real slowness difference of fast and slow flexural waves by comparing the coherence of phase spectra of the fast wave and slow wave. Contrast to the traditional time domain method, this method offers not only the normal shear anisotropy parameters but also an anisotropic map, an anisotropic monitor curve and so on, which can make the interpretation conclusion more reliable. The PCM is verified by both synthesized multi-pole array waveforms and experiment data. The results show that this method is fast and stable for anisotropy analysis.

11:20

3aEAa9. Dispersion measurement of acoustic field excited by dipole sources in fluid-filled borehole. Rui-Jia Wang, Wen-Xiao Qiao, and Xiao-Hua Che (State Key Laboratory of Petroleum Resources and Prospecting, China University of Petroleum, Beijing 102249, China, wruijia@foxmail.com)

The flexural wave in fluid-filled borehole excited by dipole sources is dispersive wave. To obtain the phase slowness of dispersive waves, firstly we made several small-scaled model wells, and then measured the acoustical field excited by dipole sources. We researched three kinds of formation, including hard formation (aluminum well models), soft formation (nylon well models) and anisotropy formation (bakelite well models). For hard formation, the flexural mode is the main propagation wave mode in the measured waveforms, and the extracted dispersion curves are well agreed with theory curves. For soft formation, the measured dispersion curves do not well agree with theory curves due to the existence of stoneley wave mode. We drilled seven holes in the anisotropic media with different dip angles. This paper only studied the 7th borehole, whose axis is perpendicular to the symmetry's in this media, which is also called HTI formation model. For the HTI formation model, we observed that the dispersion curve of the slow flexural wave is obviously higher than that of fast flexural wave at the frequency plotted, which is consistent with the theory that the dispersion curves of slow and fast flexural wave arrays for intrinsic anisotropy formation are uncrossed.

11:40

3aEAa10. Decreasing casing deformation by using cross-dipole acoustic logging data. Baohua Huang, Hao Chen, and Jianqiang Han (Institute of Acoustics, Chinese Academy of Science, huangbh@mail.ioa.ac.cn)

For increasing oil production, it is necessary to inject water into formations continuously during the oil field development. If the injection pressures are not controlled in a proper range according to in-situ stresses, it can lead to a serious imbalance of horizontal geostresses around the borehole and a higher reservoir pressure. And this will therefore cause casing deformations after a long-term exploitation. The paper presents an approach to acquire anisotropy parameters of the formation and evaluate the distribution and magnitude of in-situ stresses by using cross-dipole acoustic logging data (XMAC). Together with electronic imaging results, we can conclude an appropriate design of injection wells and thus decrease casing deformation. This approach conducted on Daqing Oilfield shows efficiency and greatly increases the economic profit.

12:00

3aEAa11. Research on sensitivity of dipole receiving transducers. Chengxuan Che, Xiuming Wang, Dehua Chen, and Jiansheng Cong (Institute of Acoustics, Chinese Academy of Sciences, chechengxuan@mail.ioa.ac.cn)

The dipole receiving transducers play an important part in Cross-dipole array acoustic well logging tools. It is necessary to study the properties of the dipole receiving transducer, especially the sensitivity of it. In this paper, transducers in two ways of wiring, including parallel connection and series connection, were considered numerically and experimentally. The sensitivities obtained by numerical and experimental method respectively were contrasted. The results show that as a receiving transducer, the receiving sensitivity in series connection is higher, which can guide us in the dipole receiving transducer design. Keywords: dipole, receiving sensitivity, receiving transducer

12:20

3aEAa12. Numerical study on the acoustic field in noncircular elastic pipes filled with fluid. Hailan Zhang, Chengxuan Che, Xiao He, and Xiuming Wang (Institute of Acoustics, Chinese Academy of Sciences, State Key Laboratory of Acoustics, 21 Beisihuanxilu, Beijing 100190, China, zhanghl@mail.ioa.ac.cn)

Due to the high cost of real field test, the testing and calibration of acoustical logging tools are usually conducted in a steel pipe filled with fluid. The information provided by the calibration in a circular pipe is not proper for nonaxisymmetric dipole tools. So the possibility of using a noncircular pipe filled with fluid for the testing and calibration of acoustic dipole logging tools is considered. In our previous report the acoustic fields generated by a dipole source in the noncircular fluid waveguide with the rigid wall were numerically studied using the finite-element software package of Comsol Multiphysics. So called 2.5D method has been used. The transformed acoustic field in a frequency wave-number domain is calculated first and from which the acoustic field in the time space domain was obtained by a 2D Fourier transformation. In this report the method is extended to include an elastic wall instead of the rigid boundary of the fluid waveguide. The results show that the contribution of the elastic wall is significant to the whole acoustic field and the propagating modes in the waveguide are coupling of the modes in the fluid part with that in the elastic wall.

WEDNESDAY MORNING, 16 MAY 2012 S221, 8:20 A.M. TO 12:00 NOON

Session 3aEAb

Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials III (Lecture/Poster Session)

Michael Haberman, Chair
haberman@arlut.utexas.edu

Contributed Papers

8:20

3aEAb1. Numerical simulation of noise induced by contact between two rough surfaces. Dang Viet Hung and Le Bot Alain (Laboratoire de Tribologie et Dynamique des Systèmes, 69134 Ecully Cedex, France, viet-hung.dang2@ec-lyon.fr)

The roughness noise generated during the sliding of two rough surfaces under light load is a complex phenomenon which is relevant to mechanics of multi contact interfaces. The objective of this study is to estimate the statistical properties of the local dynamic inaccessible to measurement (shock rate, shock duration, probability density function of local forces). A numerical approach based on a modal development has been proposed with two algorithms, penalty or Lagrange multipliers, to compute the contact force and six time integration schemas. The validity and the efficiency of this approach is discussed. In terms of CPU time, the comparisons show that the proposed approach is better than the classical finite element method. The analysis also shows that the source mechanism of roughness noise is the presence of shocks occurring between antagonist asperities transforming a part of kinetic energy into acoustical energy. It is also shown that roughness noise level is simultaneously an increasing function of the logarithm of the surface roughness and the sliding speed. Microscopically, the shock rate, the shock duration, the power being injected by individual shocks are correlated with these two macroscopic parameters.

8:40

3aEAb2. Complex-modulus measurement by longitudinal vibration testing using pulse wave. Hong Hou, Liang Sun, and Jianhua Yang (Northwestern Polytechnical University, houhong@nwpu.edu.cn)

Paper presents a method to measure the complex modulus of rigid viscoelastic materials by the longitudinal vibration testing. In the measurement, one end of a rod of the material is driven by a transducer using a pulse wave and the other end is allowed to move freely. The Complex-modulus is calculated from the displacement ratio between the driven end and the free end. The displacement of the rod end is measured by an accelerometer whose mass load should be considered since the accelerometer is attached to the rod. The displacement ratio can be obtained over a wide frequency range because the driven signal of the pulse wave is a broadband signal. The Complex-modulus can then be determined at resonant frequencies. Some rigid viscoelastic rods are measured and the measured results agree well with that of commercial viscoelastic instrument.

9:00

3aEAb3. Effective mass density for periodic solid-fluid compostics. Jun Mei (South China University of Technology, phjunmei@scut.edu.cn), Ying Wu (King Abdullah University of Science and Technology), Zhengyou Liu (Wuhan University), and Ping Sheng (Hong Kong University of Science and Technology)

Effective mass density is one of the basic parameters in the study of acoustic wave propagating in fluid-solid composites. Based on the multiple-scattering theory, an analytic solution of the dynamic effective mass density for composites with solid inclusions immersed in fluids periodically in two dimensions is obtained in the low frequency limit. When the concentration of solid is small, the dynamic mass density can be described by an angle-dependent dipole solution and the angle-dependence vanishes if the structure has a four- or six-fold symmetry. When the solid concentration is getting large, the dynamic mass density differs from the dipole solution and also becomes structure-dependent even for square and hexagonal lattices. The Wood's formula is accurately valid, independent of the structures, at any solid concentrations. Numerical evaluations from the analytic solution are shown to be in excellent agreement with finite-element simulations. In the vicinity of the tight-packing limit, the dynamic mass density exhibits a universal behavior independent of the lattice symmetry. Support of this work comes from KAUST Start-up Package, National Natural Science Foundation of China (Grant No. 10804086), the Ph.D. Programs Foundation of Ministry of Education of China (Grant No. 200804861018) and Hong Kong RGC grant HKUST 604207.

9:20

3aEAb4. Experimental investigation of elastic modes localized within a defect in a phononic slab. Bernard Bonello, Olga Boyko, Mathieu Rénier, and Rémi Marchal (INSP - University Paris 6; 4 place Jussieu 75252 Paris cedex 05, bernard.bonello@insp.jussieu.fr)

Heterostructures with periodic variations of both their optical refractive index and their elastic properties may induce band gaps for both electromagnetic and acoustic waves. As a consequence of the expected enhancement of the acousto-optic interactions, these artificial materials, called "PhoXonic crystals", are of primary interest for new sensing applications. In this context, we have experimentally studied the localization of elastic energy within a defect. The heterostructures consist in arrays of voids periodically drilled throughout silicon plates (graphite symmetry) and featuring a vacancy, or a line of vacancies. An all-optical experimental technique allows for both the generation and the detection of the elastic guided waves. The non-contact probing allows one to monitor the displacements field inside the defects. First, we have measured the dispersion of broad band elastic waves guided in between the free surfaces of the sample. Then narrow band elastic guided waves, whose central frequency corresponds to resonance modes of the cavity are generated. The optical probe allows for the measurement of the out-of-plane displacements associated to the elastic modes localized within the cavity or transmitted through the phononic structure. The spatial distribution of elastic energy inside the cavity is measured and compared with numerical predictions. This work is supported by the European Community through the FET-Open project "TAILPHOX" (Grant No. 233883).

9:40

3aEAb5. Elastic metamaterials with negative bands. Ying Wu (King Abdullah University of Science and Technology, ying.wu@kaust.edu.sa), Yun Lai (Soochow University), Ping Sheng, and Zhaoqing Zhang (Hong Kong University of Science and Technology)

The unusual properties of a metamaterial come from special resonances supported by its resonating structure units. Guided by a previously developed effective medium theory, which links the resonances of the microstructures and the unusual properties, two types of elastic metamaterials in two dimensions were designed with different resonant structures in their building blocks that exhibit multiple negative dispersion bands with special characteristics. The first type possesses negative mass density and negative shear modulus simultaneously over a large frequency regime, which leads to a negative band for shear waves only. Mode conversion takes place at the interface of the metamaterial and the common solids. The second type is able to produce negative effective moduli in different frequency regimes within a large frequency regime of negative effective mass density. This results in a super anisotropic negative band and a negative band that only compressional is allowed. All of these unusual properties are demonstrated by simulations. This work was supported by Hong Kong RGC Grant No. 605008, HKUST604207 and KAUST start-up package.

10:00

3aEAb6. Negative stiffness metamaterials and periodic composites. Michael R. Haberman, Timothy D. Klatt, Preston S. Wilson (Applied Research Laboratories & The Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX, haberman@arlut.utexas.edu), and Carolyn Seepersad (The Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX)

The creation of negative stiffness (NS) metamaterials is of interest for damping treatments, vibration isolation, and even more exotic applications such as acoustic lenses displaying negative refraction. This work will present ongoing efforts to produce NS metamaterial elements that rely on a bistable microscale geometry that leads to NS under quasistatic loading conditions. A candidate bistable microstructure employing thermal mismatch will be introduced along with finite element simulations predicting its full stiffness tensor. Effective medium modeling will show the broadband utility of these elements to enhance acoustic absorption for low volume fraction bi-material composites. The special case of a bi-material periodic composite containing NS inclusions will then be explored. It will be shown that a periodic composite containing NS inclusions permits the simultaneous elimination of the acoustic branch and reduction of the lower frequency of the optical branch passband to arbitrarily low frequencies. Physical interpretations of these results and potential applications will be discussed. *This material is based upon work supported by the U. S. Army Research Office under grant number W911NF-11-1-0032.*

10:20

3aEAb7. Vibroacoustic characteristic of membrane-type acoustic metamaterials. Yuguang Zhang and Jihong Wen (Key Laboratory of Photonic and Phononic Crystals of Ministry of Educationand Institute of Mechatronical Engineering, National University of Defense Technology, Changsha 410073, China, zyg00192@163.com)

Acoustic barriers with effective sound insulation performance can find many useful applications in the area of aerospace, automotive vehicles and environmental noise control. However, traditional acoustic barriers are always not effective at low frequencies due to the mass law. Membrane-type acoustic metamaterials have been shown to exhibit unique sound insulating performance at low frequencies (100-1000Hz). The structure of membrane-type acoustic metamaterials can be simplified as a stretched, rim fixed membrane carrying a distributed mass. There are publications on beams, rods and plate with uniformly distributed mass, but the vibration equation of membrane is different from the plate, and the presence of air around the membrane needs to be properly included. In this paper, we present an analytical model of this type acoustic metamaterials. The transmission loss and vibration characteristic are then described using the coupled structural-acoustic model. The validity of the model is confirmed by comparing our analytical results with the FEA calculations and experimental measurements in existing publications.

10:40–11:00 Break

The following abstracts will be presented in poster format. The posters will be on display and the authors will be at their posters from 11:00 a.m. to 12:00 noon

3aEAb8. An investigation on polymer modulus test using laser-based finite element method. Yao Yin, Bi-Long Liu, Guo-Feng Bai, Cheng-Guang Zhou, and Ke Liu (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, yinyao@mail.ioa.ac.cn)

A laser-based finite element(FEM) method for determination of the complex modulus of visco-elastic material is re-examined in this paper. The parametric sensitivity on FEM inverse algorithm and the applicable scope on material damping are investigated. Measurement of time-temperature superposition (TTS) has been achieved at the developed test systems. The discrepancy is within 10% in comparison with the measured date from a commercial Dynamic Mechanical Analysis (DMA), and this validates the accuracy of the developed test system.

3aEAb9. An effective medium approach and elastic metamaterials. Pai Peng and Ying Wu (King Abdullah University of Science and Technology, pai.peng@kause.edu.sa)

Metamaterials are artificial materials that are designed to manipulate waves as desired. The unusual properties of a metamaterial are derived from its complex building blocks which make the development of effective medium theory challenging. In this work, effective medium parameters, such as anisotropic mass density and moment of inertia, are obtained for metamaterials based on mechanical models consisting of masses and springs. The effective moment of inertia is capable of providing a clear understanding of the mechanism of rotational modes which are widely observed in both two- and three-dimensional phononic crystals and metamaterials. The effective medium descriptions are served as guidance in the engineering of the building block to achieve rich resonances that lead to intriguing properties of the metamaterial. (Support of this work comes from KAUST Start-up Package)

3aEAb10. Elastic wave scattering by periodic axisymmetric cavities in viscoelastic materials : theory and experiment. Guofeng Bai (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, bgf@mail.ioa.ac.cn), Bilong Liu, Ke Liu, and Fusheng Sui

The reflection, transmission and absorption performance of periodic axisymmetric cavity in viscoelastic materials are analyzed by multiple scattering(MS) method. Based on MS method of elastic wave, the transition matrix between the incident wave and scattered wave is obtained by the numerical integrals for cavity surface. Meanwhile, the complex modulus of viscoelastic materials is measured by dynamic mechanical thermal analysis and time-temerature equivalent theory. With transfer function method in water tube, the absorption coefficient curves of different specimens of rubber materials containing cylindrical cavities are obtained. The measuring results are compared with that of MS method and also verified experimentally. The results indicate energy attenuation in viscoelastic materials depends on cavity scattering properties.

3aEAb11. A micro-machined high-Q film bulk acoustic resonator for chip-scale atomic clock. Liang Tang, Quan Sun, Min Qi, and Donghai Qiao (Institute of Acoustics, Chinese Academy of Sciences, Beijing, P.R. China, tangliang@mail.ioa.ac.cn)

Taking advantage of its low power consumption and small volume, Chip-Scale Atomic Clock (CSAC) will play a critical role in portable applications, underwater communication systems, unmanned aerial vehicles, underwater sensor systems and geophysical equipments. It has aroused great research interest in these years. Traditionally, the 3.4GHz Voltage Controlled Oscillator (VCO) applied to 87Rb atomic clock is based on the frequency multiplier method of a high stable crystal oscillator. It's power hungry and can not satisfy the requirement of CSAC. Based on the fact that SiO2 film can both compensate the Temperature Coefficient of Frequency (TCF) of Film Bulk Acoustic Resonator (FBAR) and improve the Q-factor of FBAR's high-order resonance, a film bulk acoustic resonator with resonant frequency 3.4GHz, Q-factor better than 500 and TCF better than 20ppm per K was designed and fabricated with a micro-machined method, which satisfied the requirement of CSAC. The authors would thank National Natural Science Foundation of China for the support under contract number 11104313.

Session 3aHT

Hot Topics: Noise Around Airport

Yamada Yamada, Cochair
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C. K. Lee, Cochair
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Chair's Introduction—8:15

Invited Papers

8:20

3aHT1. Noise improvement measures of Hong Kong International Airport. Tim Wan Choi Hung and Chee Kwan Lee (Environmental Protection Department, Hong Kong SAR, 26th Floor Southorn Centre, Hong Kong, timhung@epd.gov.hk)

Hong Kong International Airport (HKIA) at Chek Lap Kok was designed in the early 90s and commenced operation in 1998 to meet Noise Exposure Contour (NEF) 25 standard under the Hong Kong Planning Standards and Guidelines. No new noise sensitive receiver is to be located within the NEF 25 contour. The NEF 25 contour of the airport covers largely sea areas. Only a small number of acoustically insulated village houses close to the airport marginally lie within the NEF25 contour. Despite all these planning efforts, aircraft noise from the airport operation still triggered considerable complaints from public about aircraft noise nuisance upon the airport came into operation in 1998. To address the nuisance, the Civil Aviation Department and the Environmental Protection Department of the Government of The Hong Kong Special Administrative Region implemented a number of improvement measures to reduce noise disturbance caused by these aircrafts on areas near the flight paths. This paper gives a retrospective view of aircraft noise problem in Hong Kong and experience gained after its operation.

8:40

3aHT2. Community response to aircraft noise around three airports in Vietnam. Thu Lan Nguyen, Huy Quang Nguyen, Khanh Tuyen Nguyen, Hiroaki Hukushima, Keiji Kawai, Takashi Yano (Kumamoto University, Kurokami 2-39-1, Kumamoto 860-8555, Japan, linh2lan@gmail.com), Tsuyoshi Nishimura (Sojo University, Ikeda 4-22-1, Kumamoto 860-0086, Japan), and Tetsumi Sato (Hokkai Gakuen University, Chuo Minami 26 Nishi 11-1-1, Sapporo 064-0926, Japan)

To assess the effect of aircraft noise on people in Vietnam, socio-acoustic surveys on community response to aircraft noise were conducted in residential areas around three airports in Ho Chi Minh City, Hanoi, and Da Nang. The community response was obtained by face-to-face interviews during the daytime on weekends. The aircraft noise was measured for seven successive days. Aircraft noise exposures ranged from 53 to 71 dB, from 48 to 61 dB, and from 52 to 64 dB Lden in Ho Chi Minh, Hanoi, and Danang, respectively. A representative dose-response relationship for aircraft annoyance in Vietnam has been proposed based on 3487 responses and noise data obtained by field measurements at 25 sites. It has been found that the curve for Vietnam is slightly higher than but rather almost consistent to the EU's curve. The respondents in each surveyed city have different levels of annoyance for the same aircraft noise exposure.

9:00

3aHT3. Noise around Suvarnabhumi Airport. Krittika Lertsawat (Project on the Development on Draft Law of Environmental Judicial Process, Thailand, krittikanonoise@gmail.com), Lalin Kovudhikulrungsri (International Institute of Air and Space Law, Leiden University, the Netherlands), Surocha Phoolsawat (Air Quality and Noise Management Bureau, Pollution Control Department, Thailand), and Tanaphan Suksaard (Environmental Research and Training Center, Thailand)

The noise levels in the vicinity of the Suvarnabhumi Airport (NBIA) were reported by the relevant authorities, including the noise levels before and after the opening of it since 2006. The overview of noise around NBIA will be illustrated in this paper. Meanwhile, the new guidelines on the airport noise management are under the consideration and development by the Pollution Control Department in corporation with other relevant agencies in compatible with the international recommendations on ISO 20906:2009, ISO 1996-1:2003, and ISO1996-2:2007. It should be into active within the next two years for applying to the airport projects in Thailand.

9:20

3aHT4. Treatment of auxiliary power unit as a ground noise source in airport noise modeling. Naoaki Shinohara (Narita International Airport Promotion Foundation, shino@napf.or.jp), Katsuji Iwasawa (Narita International Airport Corporation), and Ichiro Yamada (Airport Environment Improvement Foundation)

In Japan, noise index for evaluating airport noise was changed from WECPNL to Lden, which will be enforced from April, 2013. It was also decided to take aircraft ground noise within the airport when necessary. As a part of these ground noise components, it is necessary to take account of noise contributions due to APU operation on the apron before take-off, after landing, or maintenance during the midnight. This presentation explains a brief summary of an investigation of sound source characteristics of APU noise and compares noise calculations using the result with measurements observed by unattended noise monitoring devices.

9:40

3aHT5. Change of noise index and guideline values for airport noise in Japan. Ichiro Yamada (Airport Environment Improvement Foundation, K5 Bldg., 1-6-5, Haneda Kuhkou, Ohta-ku, Tokyo 144-0041, Japan, i-yamada@center.aeif.or.jp)

In Japan, mitigation of noise impact around airport has been promoted within the frame work of environmental noise measures, based on a noise guideline for aircraft noise enacted in 1974. Prompted by a subtle contradiction on noise index, the guideline was revised in 2007 and is planned to be enforced from April, 2013. The new guideline uses Lden as the cumulative noise evaluation index, instead of the conventional WECPNL. This paper explains details of revision of the guideline and the way to have specified new standard values. It also discusses the relationship of Lden with WECPNL at various airports in Japan.

Contributed Paper

10:00

3aHT6. Discussion on measurement method and standard of airport environment noise in China. Guang Yang, Jianghua Wang, Dandan Guo, Xiangdong Zhu, and Xiang Yan (School of Architecture, Tsinghua University, Beijing, China, yg@abcd.edu.cn)

With the blooming of economy, aircraft industry is growing rapidly in China. As result, the problem of disturbance from airport noise becomes more and more serious. However, airport noise control in China is still in the beginning stage. Noise testing is an essential part to hand the problem and collecting exact data can show what extent that the noise has impacted on the sensitive areas around the airport. But the Chinese standard GB 9661-88 Measurement of Aircraft noise around airport was promulgated in 1988. It has been over twenty years up to now. So there are some problems, such as, long measuring time, complicated data progressing and other noise interference including construction noise and so on in the actual measurements, which not only make the testing hard, but also difficult to evaluate the real effects accurately. This study will introduce approaches of testing points setting, data collection, filtration and progressing, and point out the practical deficiencies based on site measurements and obtained data. So the paper will be an important reference to revise measurement standard of aircraft noise.

Session 3aMU

Musical Acoustics and Speech Communication: Singing Voice in Asian Cultures

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Invited Papers

8:20

3aMU1. The tuning and vocal formant features of Chinese folk song singing: a case study of Hua'er music. Yang Yang (Institute of Education, UoL 20 Bedford Way London, WC1H 0AL, UK, yangyang.ioe@gmail.com), Johan Sundberg (KTH - Royal Institute of Technology, Drottning Kristinas v. 31 SE-100 44 Stockholm, Sweden), Graham Welch, and Evangelos Himonides (Institute of Education, UoL 20 Bedford Way, London, WC1H 0AL, UK)

Hua'er music is one of the representative folk music traditions in China today, designated as part of the world intangible cultural heritage by UNESCO in 2009. Whilst folk music traditions like Hua'er have been promoted in the education of young musicians due to their musical and cultural significance, no in-depth research has analyzed the acoustic characteristics of this vocal style. In this study, studio recordings were made of eighteen folk song examples sung by four four traditional and one formally educated singers. Analyses showed that both the traditional and the formally trained singers used a tuning pattern containing four main anchor-pitches approximating a Pythagorean tuning. The voice source in two different vocal styles ('Zhensheng' and 'Jiasheng'), reportedly used by these singers in performances, were examined by inverse filtering. LTAS of songs performed by the traditional singers were similar and showed a 'Speaker's formant' cluster near 3.5kHz. Implications are drawn for the education of folk music singers in higher education.

8:40

3aMU2. Acoustical study of classical Peking opera singing. Johan Sundberg (KTH, TMH/KTH, SE 10044 Stockholm, pjohan@speech.kth.se), Lide Gu, Qiang Huang, and Ping Huang (Voice Research Institute of China Conservatory, Peking, China)

Acoustic characteristics of classical opera singing differ considerably between the Western and the Chinese cultures. Singers in the classical Peking opera tradition are specializing on one out of a limited number of standard roles. Audio and electroglottograph signals were recorded of four performers of the Old Man roel and four performers of the Colorful Face role. Recordings were made of the singers' speech and when they sang recitatives and songs from their repertoires. Sound pressure level, fundamental frequency and spectrum characteristics were analyzed. Histograms showing the distribution of fundamental frequency showed marked peaks for the songs, suggesting a scale tone structure. Some of the intervals between these peaks were similar to those used in Western music. Vibrato rate was about 3.5 Hz, i.e., considerably slower than in Western classical singing. Spectra of vibrato-free tones contained unbroken series of harmonic partials sometimes reaching up to 17000 Hz. LTAS curves showed no trace of a singer's formant cluster. However, the Colourful Face role singers' LTAS showed a marked peak near 3300 Hz, somewhat similar to that found in Western pop music singers. The mean LTAS slope between 700 and 6000 Hz decreased by about 3 dB/octave per dB of equivalent sound level.

9:00

3aMU3. Vocal fold vibratory and acoustic features in fatigued Karaoke singers. Gaowu Wang, Andy Lo, Karen Chan (the University of Hong Kong, gwwang@hku.hk), Jiangping Kong (Peking University), and Edwin Yiu (the University of Hong Kong)

Karaoke is a popular singing entertainment particularly in Asia and is gaining more popularity in the rest of world. In Karaoke, an amateur singer sings with the background music and video (usually guided by the lyric captions on the video screen) played by Karaoke machine, using a microphone and an amplification system. As the Karaoke singers usually have no formal training, they may be more vulnerable to vocal fatigue as they may overuse and/or misuse their voices in the intensive and extensive singing activities. It is unclear whether vocal fatigue is accompanied by any vibration pattern or physiological changes of vocal folds. In this study, 20 participants aged from 18 to 23 years with normal voice were recruited to participate in an prolonged singing task, which induced vocal fatigue. High speed laryngscopic imaging and acoustic signals were recorded before and after the singing task. Images of /i/ phonation were quantitatively analyzed using the High Speed Video Processing (HSVP) program (Yiu, et al. 2010). It was found that the glottis became relatively narrower following fatigue, while the acoustic signals were not sensitive to measure change following fatigue. [Supported in part by HKRGC-GRF#757811]

9:20

3aMU4. Abrupt register changing technique "Atari" in traditional Japanese singing. Takeshi Saitou (Kanazawa University, Kakuma, Kanazawa 920-1192, Japan, t-saitou@ec.t.kanazawa-u.ac.jp), and Ken-Ichi Sakakibara (Health Sciences University of Hokkaido, 2-5, Ainosato, Kita-ku, Sapporo 002-8072, Japan)

A rapid modal-falsetto-modal register changing technique "Atari" is widely found in various Japanese traditional singing styles, such as Nagauta, Okinawan traditional singing, and Japanese traditional folk singing "Min-Yo". In other Asian singing styles, such as Urtin doo in Mongolia and similar styles in Tyva, and Sakha, consecutive rapid modal-falsetto register change is also widely used. In this study, a vocal fold vibratory pattern of "Atari" was observed by high-speed digital imaging. In "Atari", the vocal fold vibration did not break and was very smoothly carried out modal-falsetto and falsetto-modal register changing. Acoustical characteristics along with modal-falsetto-modal register changing were also analyzed. In "Atari", rapid changings of F0 and spectral tilt were observed. Based on these results, synthesis methods of "Atari" using STRAIGHT were proposed. By a listening test, it was clarified that "Atari" was effectively synthesized by changing only F0 without changing any spectral parameters.

9:40

3aMU5. Excitation source structural analysis of Japanese traditional singing voices. Hideki Kawahara (Faculty of Systems Engineering, Wakayama University, 930 Sakaedani, Wakayama 640-8510, Japan, kawahara@sys.wakayama-u.ac.jp)

New set of voice excitation source analysis methods are applied to study Japanese traditional singing voices, especially Noh. The first method, XSX (excitation Structure extractor) is capable of visualize detailed structure of subharmonic periodicity, by using multiple dedicated periodicity detectors. The second one analyzes symmetry of the fundamental component waveform, cycle by cycle. It enables to discriminate voice onset and offset details.

Contributed Papers

10:00

3aMU6. Timbral and melodic characteristics of Persian and Kurdish singing. Hama Jino Biglari and Johan Sundberg (Royal Institute of Technology, Department of Speech, Music and Hearing, TMH&KTH, SE/10044 Stockholm, Sweden, biglari@kth.se)

The floridly ornamented vocal technique in the courtly heritage of the Persian singing style called Avaz was studied along with excerpts from the flamboyant variety of the vivid Kurdish tradition. Audio and EGG signals were recorded from professional male tenor singers singing stylistically typical song excerpts from each tradition. Voice source parameters and formant settings (F1 & F2) were measured from inverse filtering of the audio signal, using the custom made DeCap (Svante Granqvist) and the commercial Soundswell softwares. Fundamental frequency F0 was measured from the EGG signal using the Soundswell CORR tool. In all melismatic embellishments, melody tones were preceded by short falsetto episodes whereby F0 quickly jumped up to a peak. For example, rapid tone repetitions as well as drill-like alternations between two neighbouring scale tones were interleaved with short falsetto segments. Moreover, for most vowels, the singers tuned F1 and sometimes also F2 to a spectrum harmonic in the higher part of their voice range, i.e. above about Bb4 (235 Hz, approximately). These findings will be discussed in relation to other singing styles, such as western operatic singing.

10:20

3aMU7. Easy confusion issues for non-Chinese scholar studying China traditional singing voices (CTSVs). Lide Gu (GU, Voice And Medical Institute, Sweden, gulide_sweden@hotmail.com)

In recent years, more and more non-Chinese scientists interested in investigation China Traditional Singing Voices (CTSVs). Misunderstandings then emerged out owing to cultural background differences. In order to make research more effective and to be carried out smoothly, some of the easy confusion issues are mentioned as reference. **A. Clearly indicates background of investigating voice sample** It is important to give clear indications of your investigating object(s) rather than using a general substantive expression, e.g. "China Native Folk Singing Voice", "Peking Opera Singing Voice", etc. because China traditional Singing Voice is varied. **B. Avoid misleading by accustomed terms** With Peking Opera example, she obviously missing some important elements of western opera must have. Her exact name is "Jing Ju", which means "Drama of Capital". **C. Concerned about the different aesthetic standards** Chinese native folk singers and local dramas performers usually have their own aesthetic standards, which is quite different from the western vocal music. **D. Try to know habitual used words of singing voices and techniques** For example, big voice or big natural voice vs chest voice, small voice vs falsetto, position of the larynx, opening of the mouth, support of the breath, etc. **E. Others**

10:40–11:00 Break

11:00

3aMU8. Analysis of Chinese singing voices and its application to singing voice synthesis. Kenko Ota (Tokyo University of Science, Suwa, 5000-1 Toyohira, Chino, Nagano 391-0292, Japan, otakenko@rs.suwa.tus.ac.jp), and Terumasa Ehara (Yamanashi Eiwa College, 888 Yokone-cho, Kofu, Yamanashi 400-8555, Japan)

Currently, many researchers work on singing voice synthesis in Japanese or English etc. However, there are few researches on singing voice synthesis in Chinese. Thus, this research tackles development of a fundamental frequency (F0) controlling method for realizing a natural vocal conversion system from a Chinese speaking voice to a singing voice. Firstly, Chinese singing voices are analyzed in order to clarify the characteristics of F0 contour. From the analysis result of Chinese singing voices, it has been clarified that the F0 of Chinese singing voices is varied in accordance with not only the acoustic characteristics affecting the singing voice perception, e.g. overshoot, but also the four tones. Then, vocal conversion system is developed based on findings. In order to confirm the validity of the developed F0 controlling model, the following synthesized singing voices are subjectively evaluated by native Chinese evaluators. One is synthesized by controlling F0 contour according to the musical note, the second is synthesized by considering the acoustic characteristics affecting the singing voice perception and the third is synthesized by the proposed F0 controlling method. As the result, the singing voices synthesized by the proposed method realize high naturalness.

11:20

3aMU9. A system for developing a series of interactive tests of vocal production requiring on-line audiovisual recording. Bing-Yi Pan (Department of Psychology, University of Prince Edward Island, bpan@upei.ca), Ding Liu (Department of Information Engineering, Hubei University for Nationalities), and Annabel J. Cohen (Department of Psychology, University of Prince Edward Island)

AIRS-TEST, an online system supporting a major collaborative research initiative, Advancing Interdisciplinary Research in Singing (AIRS), was developed. AIRS-TEST administers a sequence of interactive tests and organizes the results for analysis. The tests can present text and audiovisual information to prompt the participant's response (e.g., key presses, mouse clicks, touch-screen or audiovideo input). Researchers can design and create a sequence of related tests with auditory and/or visual stimuli via a management interface delivered by a web browser. Audiovideo recording modules can be embedded into the tests in many flexible ways. Participants need an invitation code to access a test collection. Experimental results can be explored online or downloaded for further analysis. An authority module is associated with collected data to control user's right of retrieval, considering both confidentiality and collaborative sharing. The software technologies supporting the various modules of AIRS-TEST are MySQL, Java EE, Flex and Red5. Whereas AIRS-TEST will be used worldwide to promote the AIRS study of cultural, universal, and individual influences on the acquisition of singing (Cohen *et al*, 2009, *Annals NYAS, 1169*, 112-115), AIRS-TEST can potentially support other experiments requiring on-line audio or audiovisual recording, as will be demonstrated. [Work supported by SSHRC MCRI]

11:40

3aMU10. Body radiation patterns of singing voices. Orie Takada and Rolf Bader (University of Hamburg, Institut of Musicology, Neue Rabenstr. 13, 20354, Hamburg, Germany, orie_deutschland@hotmail.com)

Most musical instruments exhibit complex patterns of sound radiation, which change with direction, played pitch, and many other factors. The same holds true for the body of a singer, regarded as an instrument, singing with her or his voice but activating also parts of the chest, neck, face, etc. The study examines differences in sound radiation of different body parts between various techniques and singing styles. Radiation patterns of the classical singing voice as well as non-classical singing styles, the Musical style, overtone singing, and Pop music, all produced by professional singers, were investigated using a microphone array comprising 128 microphones. The results are visualized displaying the strength of voice radiations and marking areas of radiation of the singer's upper body. The experiment shows that radiation patterns of the singing voice depend on vocal techniques, the vowel employed, and pitch. Additionally accelerometer measurements at lower body parts like the legs or foots show the transmission of the singing vibrations even to these remote body parts.

12:00

3aMU11. Changes in the vocal tract shape of sopranos at high pitch. Hironori Takemoto (NICT, c/o ATR 2-2-2 Hikaridai Seika-chou Soraku-gun, Kyoto 619-0288, Japan, takemoto@nict.go.jp), Kiyoshi Honda (CNRS), Takeshi Saitou (Kanazawa Univ.), Yosuke Tanabe (Hitachi America Ltd.), Hiroko Kishimoto (Showa Univ. of Music), and Tatsuya Kitamura (Konan Univ.)

As sopranos increase their fundamental frequency (F0) to sing at higher pitches, they also increase the first resonance frequency (R1) of their vocal tract. This is probably to avoid sudden F0 changes when F0 and R1 cross. It is unclear, however, how sopranos change vocal tract shape to increase R1. Therefore, the vocal tract shapes of two Japanese sopranos during production of the sung vowel /a/ in the modal register (A4 and D5) and in the falsetto register (G5) were measured by MRI. The measured vocal tract shapes were compared with each other and their area functions were extracted to calculate acoustic characteristics. Results showed that changes in the vocal tract shape were small between A4 and D5, while changes were large between D5 and G5. At G5, it was observed in both subjects that the lower jaw opened, the pharyngeal wall and tongue root advanced, and the larynx retracted. In addition, one subject shortened the laryngeal cavity length. All these changes achieved R1 increase, in agreement with the acoustic sensitivity function. Thus, in conclusion, sopranos selectively modified parts of the vocal tract with high sensitivity to R1. This research was partly supported by Kakenhi (Grant Nos. 21500184, 21300071, 22520156).

12:20

3aMU12. Analysis of high-frequency energy in singing and speech. Brian B. Monson (National Center for Voice and Speech, 136 S. Main, Ste 320, Salt Lake City, UT 84101, bbmonson@email.arizona.edu), Brad Story, and Andrew Lotto (University of Arizona, Speech and Hearing Science Dept., P.O. Box 210071, Tucson, AZ 85721)

The human singing and speech spectrum includes energy above 5 kHz, but this portion of the spectrum is typically ignored in speech and voice science. Generally it has been assumed that this high-frequency energy (HFE) contributes to only qualitative percepts of singing and speech, but prior work shows HFE contributes to several non-qualitative percepts, including speech intelligibility. To begin an in-depth exploration of HFE, a database of multi-channel anechoic high-fidelity recordings of singers and talkers was created and analyzed. Third-octave band analysis from the long-term average spectra (LTAS) showed that production level (soft vs. normal vs. loud), production mode (singing vs. speech), and phoneme (for voiceless fricatives) all significantly affected HFE characteristics. Female HFE levels were significantly greater than male levels only above 11 kHz. As expected, HFE was found to be highly directional toward the front of the singer/talker. While this information resulted from a study initially focused on singing voice aesthetic, it is pertinent to various areas of acoustics, including vocal tract modeling, voice synthesis, augmentative hearing technology (hearing aids and cochlear implants), cell phone technology, and training/therapy for singing and speech. [Work supported by NIH-NIDCD.]

Session 3aNSa

Noise, Engineering Acoustics, Physical Acoustics, and Signal Processing in Acoustics: Active Noise Control I

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Invited Papers

9:20

3aNSa1. Lessons learned for implementing near-field active control systems to achieve global control of fan noise. Scott D. Sommerfeldt and Kent L. Gee (Brigham Young University, Provo, UT 84602, scott_sommerfeldt@byu.edu)

Work has been ongoing for a number of years to develop compact active noise control systems that can effectively achieve global attenuation of radiated fan noise associated with information technology (IT) equipment. Both axial and centrifugal fans are used in IT applications, and both types of fans have been studied. An approach has been developed that can guide the user to properly implement near-field sources and sensors to achieve significant global attenuation of the fan noise. These concepts have been investigated experimentally in numerous configurations to verify the effectiveness of the approach. Both narrowband and broadband noise components have been targeted, and limitations on the attenuation that can be achieved have been established. The sensitivity of various configurations has also been studied, and it has been found that some implementations may be able to achieve greater attenuation, but are also much more sensitive to placement errors of the sources and sensors. This paper will overview the research to date and provide an overview of lessons that have been learned for effective implementation of near-field control techniques in order to achieve global attenuation.

9:40

3aNSa2. An active vibration isolation system for an air-compressor in marine applications. Tiejun Yang, Minggang Zhu, Xueguang Liu, Jingtao Du, and Zhigang Liu (Power and Energy Engineering College, Harbin Engineering University, Harbin 150001, China, yangtiejun@yahoo.cn)

Similar to a diesel engine, the main vibration sources of an air-compressor are to and fro inertial forces of its piston, rod, and crankshaft. An active vibration isolation system is developed for an air-compressor in a tug boat. This system consists of four inertial actuators and a DSP processor. A four-input and four-output adaptive control strategy is applied and the reference input signal comes from a laser tachometer on the shaft of the motor. The active vibration isolation experiment is conducted in a tug boat when only the air-compressor is working. The experimental results demonstrate that good vibration reductions are obtained at not only error sensor's locations but also the hull under the compressor. Some discussion and conclusions are given at last.

10:00

3aNSa3. Active control of noise in vehicle cabins. Jie Pan (School of Mechanical Engineering, University of Western Australia, Crawley WA 6009, Australia, pan@mech.uwa.edu.au), Yulan Liu, Jing Lu, and Xiaojun Qiu (The Institute of Acoustics, Nanjing University, Nanjing 210093, China)

Active noise control (ANC) technology has been applied to reduce low frequency noise in vehicle cabins. Different from ANC in laboratory conditions, installation of the ANC system in vehicles for a long term use requires the solution of a number of practical issues. This paper reports the progress of a couple of practical ANC installations, and the implication and solution of practical issues such as magnitude and frequency variations (corresponding to different vehicle operating conditions) of the primary and reference signals, and poor coherence between primary and reference signals.

10:20

3aNSa4. Head-mounted active noise control system and its application to reducing MRI noise. Nobuhiro Miyazaki and Yoshinobu Kajikawa (Kansai University, miya.nbhr@gmail.com)

We propose an active noise control (ANC) system for reducing periodic noise generated in a high magnetic field such as noise generated from magnetic resonance imaging (MRI) devices (MRI noise). The proposed ANC system utilizes optical microphones and piezoelectric loudspeakers and consists of a head-mounted structure to control noise near the user's ears. Moreover, internal model control (IMC)-based feedback ANC is employed because the MRI noise includes some periodic components and is predictable. Our experimental results demonstrate that the proposed ANC system (head-mounted structure) can significantly reduce MRI noise by approximately 30 dB in a high field in an actual MRI room even if the imaging mode changes frequently.

10:40–11:00 Break

11:00

3aNSa5. Difference potential based active noise shielding in three dimensional settings. Yiu Wai Lam (Acoustics Research Centre, University of Salford, Salford M5 4WT, UK, y.w.lam@salford.ac.uk), Sergey Utyuzhnikov (School of Mechanical, Aerospace and Civil Engineering, University of Manchester, M13 9PL, UK), and Liam Kelly (Acoustics Research Centre, University of Salford, Salford M5 4WT, UK)

Active noise shielding based on the Difference Potential Method has been developed and validated in previous publications. The key ability of the method, to provide global noise cancellation while automatically preserving wanted sound in multi-connected domains, has been demonstrated clearly in 1D experiments. In this paper, the tests are extended to 3D. The shielded domain is a 1mx1mx1m box with 1-3 sides open. Up to 4 control sources were used on each of the open sides to shield the domain from external noise. Noise attenuation of around 10dB was achieved up to about 350 Hz. The attenuation is lower than the >20dB attenuation achieved in previous 1D experiments. A reason for this is the sensitivity of the cancellation to the positioning of sensors and controls and the physical size of the control sources. Furthermore, up till now the method relies on separate measurements of the total sound field without the control sources. In realistic applications where noise and wanted sound are non-stationary, the total sound field has to be measured in real time with the control sources operating. In such cases a new mathematical formulation is needed, and this paper will discuss how such a solution can be achieved.

11:20

3aNSa6. Fixed point realization of partial updating adaptive algorithm for active noise control. Jing Lu, Kai Chen, and Haishan Zou (Institute of Acoustics, Nanjing University, Nanjing 210093, China, lujing@nju.edu.cn)

An efficient low cost realization of the partial updating FXLMS algorithm is proposed in this paper. Unlike the conventional partial updating adaptive filters, the modified sign-LMS algorithm is used to make it easier to be implemented in a fixed point DSP. Moreover, the calculation of the filtered reference signal is divided into several steps, which further reduces the whole computational burden. An active noise controller using TMS320C5502 DSP chip is designed based on the proposed algorithm. The experiment results demonstrate that at a reasonable updating rate, the proposed algorithm has exactly the same performance with the normal float-point FXLMS algorithm while the computational burden is much lower.

11:40

3aNSa7. A novel frequency-domain periodic active noise control algorithm with nonlinear secondary path. Ming Wu and Jun Yang (Institute of Acoustics, Chinese Academy of Sciences, 100190, mingwu@mail.ioa.ac.cn)

Frequency-domain periodic active noise control (FDPANC) algorithm is usually adopted for harmonic noise reduction. However it may fail to converge when the secondary path includes a nonlinear element. In this paper, the convergence behavior of FDPANC algorithm with nonlinear secondary path is analyzed and a sufficient condition for convergence is derived. Unlike the conventional FDPANC algorithm with linear secondary path, this sufficient condition for FDPANC algorithm with nonlinear secondary path is time-varying and depending on the coefficients of the adaptive filter. If system has a steady-solution, the sufficient condition can be satisfied for all iterations by selecting the proper initial values. In this situation, the adaptive filter will convergent to the optimum value and the disturbance signal can be eliminated totally. If system has no steady-solution, the sufficient condition cannot be satisfied for all iterations whatever the initial values are. Based on the sufficient condition for convergence, a modified FDPANC algorithm integrated with convergence detector is also proposed in this paper.

12:00

3aNSa8. Varying auxiliary-noise-power for improved noise reduction in active noise control systems with online secondary path modeling. Shakeel Ahmed (Department of Communication Engineering and Informatics, The University of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan, s1196001@edu.cc.uec.ac.jp), Muhammad Tahir Akhtar (The Center for Frontier Science and Engineering (CFSE), The University of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan), and Wataru Mitsuhashi (Department of Communication Engineering and Informatics, The University of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan)

Active noise control (ANC) systems with online secondary path modeling (SPM) comprise two adaptive filters: 1) The FXLMS algorithm-based ANC filter to cancel the primary noise, and 2) an LMS algorithm-based online SPM filter. The online SPM filter is a supporting filter and is needed for a stable operation of the FXLMS algorithm-based ANC filter. The online SPM is achieved by injecting an auxiliary noise, which however degrades the noise reduction performance of the ANC system. In this paper, a new method is proposed for varying the power of the injected auxiliary noise in accordance with the convergence status of the power of the error signal of the SPM filter. Furthermore, in order to deal with the non-stationary noise sources, normalized step-sizes are employed for both the ANC and SPM filters. The proposed method for varying the auxiliary-noise-power achieves a good online SPM, and an improved noise reduction performance as compared with the existing methods. Furthermore, it has a reduced computational complexity as compared with the method proposed by Carini et. al. In order to verify the effectiveness of the proposed method, various scenarios of the input noise are considered in the simulation results presented in this paper.

12:20

3aNSa9. Active noise control systems integrated with infant cry detection and classification for infant incubators. Lichuan Liu and Kevin Kuo (Northern Illinois University, Dekalb, IL 60115, liu@niu.edu)

Infant incubators are widely used in neonatal intensive care units (NICU) for the pre-term or sick babies. In this paper, we propose an active noise control (ANC) system integrated with infant cry detection and classification function for infant incubators. The incubator ANC system reduces the high noise level which results in numerous adverse health effects for infants, and the cry detector and analyzer monitors the infants' physical conditions using the same ANC system hardware. The infant crying signals picked up by the error microphone of ANC inside the incubator are detected, the signals' features are extracted by using linear prediction coding (LPC), and Mel-frequency cepstral coefficients (MFCC) methods. The database of different signals and their associated infant health features will be established through training phase. The signal classification algorithm is based on shortest distance and Bayesian classifier. Simulation is conducted to evaluate the performance of proposed system using a Giraffe incubator from GE Healthcare. The experiments show that the proposed ANC system can reduce the harmful noise and the cry classification algorithm can recognize infant cries in order to develop the innovative, low cost methods to monitor the infant's health conditions.

WEDNESDAY MORNING, 16 MAY 2012 HALL C, 9:20 A.M. TO 11:40 A.M.

Session 3aNSb

Noise, ASA Committee on Standards, and Animal Bioacoustics: Assessment and Measurement of Park Soundscapes

Paul Schomer, Cochair
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Invited Papers

9:20

3aNSb1. The ISO 12913 series on soundscape: An update, May 2012. Östen Axelsson (Department of Psychology, Stockholm University, SE-106 91 Stockholm, Sweden, oan@psychology.su.se), and on behalf of ISO/TC 43/SC 1/WG 54

In February 2009 the working group ISO/TC 43/SC 1/WG 54 "Perceptual assessment of soundscape quality", of the International Organization for Standardization (ISO), begun preparing the first International Standard on soundscape "ISO 12913-1 Acoustics –- Soundscape –- Part 1: Definition and conceptual framework". This paper presents the latest version of the definition of "soundscape" and its conceptual framework. At its current state of development the framework highlights seven general concepts and their relationships: (1) sound sources, (2) acoustic environment, (3) auditory sensations, (4) interpretation of auditory sensations, (5) responses, (6) context, and (7) outcomes. By providing a standard reference, the working group aims at international consensus in order to avoid ambiguity, and to enable conceptual progress in soundscape research. ISO 12913-1 is expected to be published as an International Standard in 2015. Subsequent parts of the ISO 12913 series will deal with minimum reporting requirements in soundscape research, and methods for measuring soundscape quality.

9:40

3aNSb2. Protecting soundscapes in U.S. National Parks: lessons learned and tools developed. Peter Newman (Colorado State University, peter.newman@colostate.edu), Kurt Fristrup, Karen Trevino (National Park Service), Steve Lawson (Resource Systems Group), Derrick Taff, Dave Weinzimmer, and Tim Archie (Colorado State University)

Researchers and protected area managers' are working together to protect natural soundscapes in U.S. National Parks. In this paper, soundscapes have been defined as the total acoustics environment and includes the sounds of nature and as well as anthropogenic noise (unwanted sounds). In particular, human-caused noise can mask the sounds of nature and detract from the quality of the visitor experience and have negative impacts on wildlife in parks and protected areas. Over the past decade, researchers at Colorado State University have teamed up with the United States National Park Service (USNPS) to explore, build simulation models of, and derive management actions in National Parks in order to protect natural quiet and the soundscapes of national parks. This paper will provide an overview of challenges and successes of these efforts in order to create a list of lessons learned. In particular, results (maps, models and experiments) of studies in Denali National Park, Rocky Mountain National Park Yosemite National Park and Sequoia Kings National Park will be shown and presented in order to show how these data can lead to informed management decision making. This research was funded by the USNPS Natural Sounds Program as well as support from aforementioned parks.

10:00

3aNSb3. Further research on separating anthropogenic from natural sounds in a park setting. Jack Gillette, Jeremy Kemball, and Paul Schomer (Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821, gillett1@uni.illinois.edu)

This paper is a continuation of work presented last year on this subject. Last year's work involved separating nearly all anthropogenic sound from natural sound except for jet aircraft. This year a method to detect jet aircraft has been developed, so that now one is able to separate all anthropogenic sound from natural sound. Testing of the detection of non-aircraft anthropogenic sound has been continued with a wide body of data. For the detection of anthropogenic sound (other than jet aircraft) we rely on the assumption that nearly all of this sound involves the use of motors and, hence, except for aircraft, creates fundamental tones that are well below 1000 Hz, where as natural sounds (e.g., insects, birds, frogs) nearly always will create tones with fundamental frequencies above 1000 Hz. This paper present the results of the testing for aircraft noise detection and the additional testing of the detection of non-aircraft anthropogenic sound.

10:20

3aNSb4. A pretest of several versions of a survey questionnaire for use in Rocky Mountain National Park, USA. Paul Schomer and James Boyle (Schomer and Associates, Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

This paper presents the results of a pretest of visitors' perception of the soundscape on hiking trails in the Bear Lake area of Rocky Mountain National Park, Colorado, USA. The pretest involved 5 versions of a procedure for administering the questionnaire. This pretest includes concurrent noise monitoring with the noise monitor generally spaced 1 to 2 km apart. For those versions of the pretest survey that included questions during the hike, the test booklet included a position and time recording GPS. Although the majority of visitors to the Bear Lake area go on 0.5 to 3 km long hikes, most hikers that we selected went on hikes that were in the length range of 5 to 10 km. The Bear Lake area is the busiest hiking area in Rocky Mountain National Park; it receives about 10 to 20,000 visitors per day during nice summer days. As a result, one source of anthropogenic sound that contributed negatively to the soundscape was the sound of other hikers; beyond about 1.5 km from the transit area, hikers on the trails thinned markedly. Based on the results of the pretest analysis, a full-scale test plan is recommended.

10:40–11:00 Break

11:00

3aNSb5. Assessment of tranquility in an urban church garden. Jin Yong Jeon, Inhwan Hwang, and Jooyoung Hong (Hanyang University, Seongdong-gu, Seoul, Korea, jyjeon@hanyang.ac.kr)

Tranquility in view of urban church soundscape was assessed by field measurements and soundwalks. The Cathedral Myeong-dong located in the heart of Seoul was selected as a measurement site. Five sites adjacent to the church buildings were selected for field measurements and audio recordings were conducted at the sites. From the field measurements, the temporal and frequency characteristics, and several acoustic parameters of the sound environment at each site were analyzed. In addition, a general questionnaire survey was conducted to identify people's perception on the current church soundscape and design elements to enhance the spaces. Individual soundwalks were also performed in order to evaluate soundscape perception in the church garden. From the results, the indicators representing tranquility sensation in urban church garden were investigated.

11:20

3aNSb6. The role of paying attention to sounds in soundscape perception. Dick Botteldooren, Michiel Boes, Damiano Oldoni, and Bert De Coensel (Ghent University, Gent, Belgium, dick.botteldooren@intec.ugent.be)

It has been stated frequently that the soundscape as perceived and appraised by the user of a space, extends beyond the physical stimulus. We argue that, when introducing to human-factor in analyzing a sonic environment, the sounds that people hear play an important role. This holds in particular for rather quiet and infrequent disturbance of park soundscapes. Auditory attention mechanisms are essential in the process. Attention can be drawn by saliency elements such as changes in time and frequency, but it can also be outward oriented and voluntary. These mechanisms could explain the special role of natural sounds in distracting attention from mechanical background hum in a park environment. These theoretical concepts have now been implemented in measuring equipment that allows estimating how often particular sounds will be heard by a human listener. The methodology includes biologically inspired feature extraction, learning based on co-occurrence of features and saliency, attention focusing, and inhibition of return. Extension to binaural measurements increasing the unmasking effect is also discussed.

Session 3aPA

Physical Acoustics and Biomedical Acoustics: Metrics and Objectivity in Cavitation Research

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Invited Papers

8:00

3aPA1. Cavitation thresholds and why to be wary! Victoria Bull (Physics Department, Institute of Cancer research, Royal Marsden Hospital, Sutton, Surrey SM2 5PT UK, victoria.bull@icr.ac.uk), Ian Rivens (Physics Department, Institute of Cancer Research, Sutton, Surrey, SM2 5PT, UK), and Gail ter Haar (Physics Department, Institute of Cancer Research, Royal Marsden Hospital, Sutton, Surrey, SM2 5PT, UK)

Many biological studies involving ultrasound exposure report acoustic cavitation "thresholds". The purpose of these is often to convince the reader either that cavitation activity was avoided, or that it definitely occurred as the exposure was "above the threshold". Formally, the cavitation threshold is the minimum negative pressure amplitude at which pre-existing bubbles begin to oscillate (non-inertial cavitation) or collapse (inertial cavitation). The cavitation nucleation threshold is that for which bubbles can be drawn out of solution and driven to oscillate. Since the true physical cavitation threshold of a medium can only be measured if it is possible to detect single bubble activity, quoted thresholds are more likely to represent a threshold for detection than for cavitation activity. Furthermore, they may not be relevant beyond the specific experimental conditions tested. In interpreting a negative pressure threshold for cavitation in tissue it is important to know, amongst other things: the tissue status and sample geometry (in order that the in situ pressure may be calculated); details of the ultrasound exposure; method of cavitation detection and detector geometry; data gathering, processing and interpretation methods; threshold definition; sample statistics. Without these details, results may be misleading.

8:20

3aPA2. The influence of thresholding passive bubble-acoustic signals on the quantification of physical effects. Richard Manasseh (Engineering & Industrial Sciences, Swinburne University of Technology, P.O. Box 218, Hawthorn, VIC 3122, Melbourne, Australia, rmanasseh@swin.edu.au), Yonggang Zhu (Fluid Dynamics, CSIRO, P.O. Box 56, Highett, VIC 3190, Melbourne, Australia), Hubert Chanson (Civil Engineering, University of Queensland, QLD 4072, Brisbane, Australia), Andrew Ooi (Mechanical Engineering, University of Melbourne, VIC 3010, Melbourne, Australia), Alexander Babanin (Engineering & Industrial Sciences, Swinburne University of Technology, P.O. Box 218, Hawthorn, VIC 3122, Melbourne, Australia), and Irena Bobevski (Psychology and Psychiatry, Monash University, Monash Medical Centre, Clayton, VIC 3168, Melbourne, Australia)

Passive emissions of sound by bubbles provide potentially useful data in many natural and engineered systems. Bubble-acoustic data are used to predict the severity of volcanic eruptions, identify undersea gas seeps, and measure ocean wave breaking. In these natural cases, and in many chemical engineering and metallurgical processes, statistics on the bubble size are sought; the bubble size controls the rate of gas to liquid mass transfer as well as fluid dynamical aspects such as mixing. However, owing to uncertainty in the physics generating the acoustic amplitude, bubble-size determination remain essentially empirical, requiring an threshold to select data. The impact of various thresholding and data-windowing techniques is discussed. Examples are given on the data from a chemical engineering aerator, a plunging jet, and a case in which diagnostic statistics can be used to identify the threshold giving the optimum sensitivity and specificity for detecting the breaking of wind-waves. Once found, the optimum threshold can be used to infer relevant physics from the acoustic signal alone.

8:40

3aPA3. Cavitation micro-bubble behavior in ultrasound field for HIFU therapy enhancement. Yoichiro Matsumoto (The University of Tokyo, Hongo, Bunkyo-ku, Tokyo, Japan, ymats@fel.t.u-tokyo.ac.jp), Kohei Okita, and Shu Takagi

The incepted cavitation micro-bubbles at the target tissue are used to enhance tissue heating in High-Intensity Focused Ultrasound therapy. The control of the inertial cavitation is required to achieve the efficient tissue ablation. The high-intensity ultrasound is used to incept a cavitation, so that the duration time is short enough to prevent the tissue coagulation. The low-intensity ultrasound follows just after the high-intensity ultrasound to continuously drive the cavitating micro-bubbles and coagulate the tissue. Following the experiment,

we simulate the focusing of ultrasounds through a mixture containing micro-bubbles with considering the size and number density distributions in space. The numerical simulation shows that the heating region coincides with the cavitation incepted region by high-intensity ultrasound where the micro-bubbles are remaining as cavitation nuclei. The numerical results elucidate well the experimental ones.

9:00

3aPA4. Postexcitation collapse as a characteristic of single ultrasound contrast agent destruction. Daniel King and William OBrien Jr (University of Illinois Urbana-Champaign, IL, daking3@illinois.edu)

Measurement of the response of single, unconstrained ultrasound contrast agents (UCAs) is useful for facilitating experiment interpretation and theoretical comparison. An experimental setup has been developed to characterize the acoustic large amplitude response of microbubbles called double passive cavitation detection (PCD) which consists of three confocally aligned transducers. The symmetric single bubble responses from within the confocal region are analyzed for the presence or absence of a postexcitation signal (PES), a rebound characteristic response to large amplitude pressures that may follow the initial harmonic UCA response. Experimental sensitivity to the PES is explored by receiving with transducers of different frequencies. Theoretical models indicate postexcitation rebound occurs following shell rupture and inertial cavitation of the UCA. The postexcitation response curves as a cavitation metric are useful for characterizing distinct collapse thresholds among UCAs which arise due to material differences; additionally, the thresholds may be considered as destruction thresholds and compared to a variety of in-vitro and ¬in-vivo studies to aid in understanding the resultant bioeffects in these studies. (NIH Grant R37EB002641.)

9:20

3aPA5. Spatiotemporal quantification of therapeutically relevant cavitation for ablation and drug delivery by ultrasound. Constantin-C. Coussios, Carl Jensen (Institute of Biomedical Engineering, Department of Engineering Science, University of Oxford, Oxford OX3 7DQ, UK, constantin.coussios@eng.ox.ac.uk), Rob Ritchie (Clinical HIFU Unit, Churchill Hospital, Oxford OX3 7LJ, UK), Miklos Gyongy (Faculty of Information Engineering, Pázmány Péter Catholic University, Budapest, Hungary), Natalie Hockham, Bassel Rifai, Eleonora Mylonopoulou, James Choi, and Jamie R.T. Collin (Institute of Biomedical Engineering, Department of Engineering Science, University of Oxford, Oxford OX3 7DQ, UK)

Stable and inertial cavitation have been shown to play a key role in several therapeutic ultrasound applications, ranging from non-invasive ablation to drug delivery. In order to achieve meaningful quantification of therapeutically relevant cavitation, a hypothesis as to the underlying mechanism by which bubbles enhance energy delivery and momentum transfer to their surroundings must first be formulated. Based on this hypothesis, correlations can be sought between time-based or frequency-based metrics of a relevant cavitation 'dose' and the desirable associated bioeffects, enabling the establishment of thresholds for therapeutically relevant processes to occur. However, the problem is complicated further by the fact that a single type of cavitation activity rarely occurs throughout an ultrasound exposure, or throughout the acoustic field of a therapeutic transducer. A recently developed method, known as passive acoustic mapping, makes it possible to obtain real-time maps of inertial and stable cavitation activity and could thus enable improved dosimetry of therapeutically relevant cavitational processes. The use of passive acoustic mapping for spatiotemporal classification and quantification of cavitation will be illustrated in the context of HIFU ablation, drug release from thermosensitive liposomes and drug delivery.

9:40

3aPA6. Effect of microbubble-induced microstreaming on the sonoporation. Dong Zhang and Juan Tu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, dzhang@nju.edu.cn)

In the present work, MCF-7 cells mixed with polyethylenimine: Deoxyribonucleic acid (DNA) complex and microbubbles were exposed to 1-MHz ultrasound with low acoustic driving pressures (0.05-0.3 MPa). The sonoporation pores generated on the cell membrane were examined with scanning electron microscopy. The results show that larger sonoporation pores would be generated with the increasing acoustic pressure or longer treatment time, which could be useful for the enhancement of DNA transfection efficiency, the mean diameters of pores ranged from 100nm to 1.25 μm. The calculated results based on Marmottant model indicate that the microstreaming-induced shear stress might be involved in the mechanisms of the low-intensity ultrasound induced sonoporation. [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)]

10:00

3aPA7. Quantification of bubble population in bubbly liquids. Won-Suk Ohm (Yonsei University, Seoul 120-749, Korea, ohm@yonsei.ac.kr)

Estimation of bubble spectra is usually performed through the inversion of attenuation and/or dispersion measurements of bubbly liquids [see, for example, Commander and Prosperetti, J. Acoust. Soc. Am. 85, 732-746 (1989)]. However, the inversion method often fails to yield accurate bubble spectra, as the bubble resonance starts to play a significant role. An alternative approach is to infer the bubble population from the sound of Minnaert-like bubble oscillations within the bubble cloud [Leighton and Walton, Eur. J. Phys. 8, 98-104 (1987)]. In this talk, a survey of the existing methods for bubble spectrum estimation is given with a particular emphasis on the pros and cons of each method under different circumstances.

10:20

3aPA8. Measurement of density and size distribution of oscillating cavitation bubbles using optical spectrometer. Takanobu Kuroyama, Tadashi Ebihara (University of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8577, kuroyama@aclab.esys.tsukuba.ac.jp), Koichi Mizutani (Unviersity of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8577), and Takeshi Ohbuchi (National Defense Academy, 1-10-20 Hashirimizu, Yokosuka, Kanagawa 239-8686)

Measuring of properties of acoustic cavitation bubbles, such as size distribution and density distribution, is important to estimate the physical and chemical effects of the cavitation bubbles. In this paper, the optical spectrometer is applied to measure the bubble density in the cavitation field. In previous study, it has been proposed that a size distribution measurement method of the acoustic cavitation bubbles using the optical spectrometer by the authors. The size distribution measurement of the oscillating bubbles was successfully achieved by measuring the far-field (Fraunhofer) diffraction pattern of the bubbles, which was obtained by introducing the laser light to the cavitation bubbles. In addition to the previous study that focused on the diffracted light pattern, the bubble density is measured from the diffracted light intensity in this paper. The obtained results suggest that the proposed method can measure the bubble densities successfully.

10:40–11:00 Break

11:00

3aPA9. Evolution of acoustic cavitation structures induced by a liquid jet. Lixin Bai, Jingjun Deng, and Chao Li (Ultrasonic Physics and Exploration Laboratory, Institute of Acoustic, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, lixin.bai@gmail.com)

Temporal evolution and spatial distribution of acoustic cavitation structures induced by a liquid jet in a 20 kHz ultrasonic field were investigated experimentally with high-speed photography. It is revealed that a large-scale flare structure is produced by a liquid jet shooting towards a transducer radiating surface. A few large bubbles and numerous small bubbles are formed on the interface of liquid jet. The size of large bubbles differs almost by an order of magnitude from that of small bubbles. The flare structure hits the radiating surface, spread and adhere. The liquid jet is arrested and a thin bubbly layer lying at the radiating surface is formed. The cavitation cloud will evolve from one structure to another without the interference of liquid jet. (a) Most bubbles move towards the center, and the bubbly layer will become thicker at the centre than at the periphery like a lens. (b) Large bubbles accumulate at the center and skip off the surface, and the lens structure becomes a conical structure. (c) Large bubbles align themselves along the axial line, and the conical structure becomes a fishbone structure. Many other interesting phenomena were observed. The liquid-jet method is useful for the investigation of cavitation structure. This work is supported by the National Natural Science Foundation of China (Grant No. 11174315) and the National Science and Technology Major Project of China (Grant No. 2011ZX05032-003).

11:20

3aPA10. High-speed observation of bubble dynamics influenced by surfactant molecules. Pak-Kon Choi and Shota Deno (Meiji University, 1-1-1 Higashimita, Tama-ku, Kawasaki, Japan, pkchoi@isc.meiji.ac.jp)

Surfactant molecules affect the dynamics of acoustic bubbles such as the oscillation of expansion and contraction and a bubble coalescence when adsorbed with bubble/liquid interface. A high-speed shadowgraph of acoustic bubbles was observed in surfactant SDS solutions using a high-speed camera with a speed of a million fps. The experimental results of time evolution of bubble diameter showed that a surfactant-adorbed bubble favors a spherical oscillation. Histograms of maximum bubble diameters obtained at several frequencies indicated that the number of large-size bubble considerably decreased compared to the case of pure water and this tendency was marked at higher frequencies. The histograms measured at various SDS concentrations at 87 kHz showed that the decrease in the number is significant at 5 mM. The results are elucidated in terms of electrostatic repulsion between charged bubbles, which inhibits the bubble coalescence.

11:40

3aPA11. An axisymmetric model of ultrasound contrast agent collapse following shell rupture. Daniel A. King, Jonathan B. Freund, and William D. O'Brien, Jr. (University of Illinois, Urbana-Champaign, IL, daking3@illinois.edu)

Widely used spherically symmetric models of ultrasound contrast agents (UCA) are unable to capture complicated collapse behaviors when the UCA shell ruptures. Observations from acoustic passive cavitation detection analysis of collapsing UCAs as well as images from high speed videos of UCA destruction suggest that including spatial asymmetry in the shell interface conditions of models may be useful for studying collapse, onset of fragmentation, and initiation of postexcitation rebound. The concept of lipid shell rupture is incorporated into an axisymmetric boundary element method formulation as a circular hole developing during the growth phase of an initially spherical bubble. The different material interfaces are represented by a spatially varying surface tension due to different pressure discontinuity conditions from the inner gas region to the outer fluid region. Results from the simplified geometrical model simulations demonstrate that shell fragments may influence the evolution of collapsing UCA and indicate the possibility of microbubble jetting following lipid shell rupture. (NIH Grant R37EB002641.)

12:00

3aPA12. Theoretical microbubble dynamics in a homogeneous viscoelastic medium at capillary breaching thresholds. Brandon Patterson, Douglas Miller, and Eric Johnsen (University of Michigan, 1231 Beal Ave., Walter E. Lay Automotive Engineering Laboratory, 2016 Ann Arbor, MI 48109-2133, awesome@umich.edu)

In order to predict ultrasound-induced bioeffects in diagnostic and therapeutic procedures, the dynamics of cavitation microbubbles in viscoelastic media must be determined. For this theoretical study, measured 1.5-7.5 MHz pulse pressure waveforms, which were used in experimental determinations of capillary breaching thresholds for contrast enhanced diagnostic ultrasound in rat kidney (Miller et al. Ultras. Med. Biol. 2008;34:1678), were used to calculate cavitation nucleated from contrast agent microbubbles. A numerical model for cavitation in tissue was developed using the Keller-Miksis equation (compressible Rayleigh-Plesset equation), with various viscoelastic models (Kelvin-Voigt, Maxwell and Standard Linear Solid). From this model, the bubble dynamics corresponding to the experimentally obtained capillary breaching thresholds were determined. For example, the particular value of the maximum radius and temperature corresponding to this threshold was determined for a range of ultrasound pulses and bubble sizes, and was shown to be frequency-dependent. It was observed that the bioeffects thresholds do not correspond exactly to inertial cavitation thresholds, suggesting the possibility of a more complex mechanism of injury. We plan to identify local, secondary bioeffect mechanisms, which might be constant for threshold conditions at different frequencies.

12:20

3aPA13. Theoretical cavitation thresholds in vivo and their relationship to the mechanical index. Charles C. Church (University of Mississippi, University, MS 38677, cchurch@olemiss.edu)

The mechanical index (MI) displayed on the screens of most modern diagnostic ultrasound machines is a relative indication of bioeffect risk. The current definition of the MI, the ratio of the rarefactional pressure in situ to the square root of the center frequency of the acoustic wave, is based on an analytic evaluation of the threshold for violent cavitation of pre-existing gas bubbles in biologically relevant fluids, i.e., water and blood. An alternative approach assuming spontaneous nucleation of bubbles when these liquids are under tension produced by an acoustic field indicated that the threshold was nearly independent of frequency in water but directly proportional to frequency in blood. More recently, it has been suggested that the lipid

3a WED. AM

bilayers surrounding and within individual cells may rupture when subjected to a rarefactional pressure, thereby producing cavitation nuclei that may be further acted upon by the acoustic field. By combining theoretical thresholds for bilayer-nucleation and cavitation in tissue, it is shown that the maximum acoustic pressure permitted by the US FDA for the safe use of diagnostic ultrasound is lower than necessary in most circumstances. An evidence-based safety threshold for non-thermal effects of diagnostic ultrasound will be proposed. (NIH Grant R21EB013763-01)

WEDNESDAY MORNING, 16 MAY 2012 S423, 8:20 A.M. TO 12:40 P.M.

Session 3aPP

Psychological and Physiological Acoustics: Cortical Neuroimaging Techniques in Auditory Perception and Cognition

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Invited Papers

8:20

3aPP1. Mapping the human cortex associated with auditory processing in space, time and frequency using different neuroimaging techniques. Adrian KC Lee (Dept. of Speech and Hearing and Inst. For Learning and Brain Sci. (I-LABS), Portage Bay Bldg. Rm. 204, Box 357988, Univ. of Washington, Seattle, WA 98195, akclee@uw.edu)

Functional magnetic resonance imaging (fMRI), Magneto- and Electro-encephalography (M-EEG) are neuroimaging techniques that have been used extensively to study human auditory perception and cognition. Due to the different spatial and temporal resolutions associated with these methods, each technique offers a different unique window into how our cortex participates in auditory tasks. In this talk, a summary of how each presentation in this session leverages the strengths of these neuroimaging techniques to map how our cortex dynamically responds to different sound features will be presented. Other methodological advances that are particularly relevant to experiments in auditory perception and cognition will also be discussed. Funded by USA-NIH R00DC010196.

8:40

3aPP2. Using functional magnetic resonance imaging to explore the representation of binaural cues in human auditory cortex. G Christopher Stecker (University of Washington, 1417 NE 42nd St, Seattle, WA 98105, cstecker@uw.edu), and Susan A McLaughlin (University of Washington, 1417 NE 42nd St, Seattle, WA 98105)

Functional magnetic resonance imaging (fMRI) was used to examine representations of interaural differences of level (ILD) and time (ITD) in human auditory cortex (AC). In one experiment, ILD of amplitude-modulated sounds varied parametrically across 12-s blocks, with a single image acquired at the end of each block (i.e., using a “sparse” imaging protocol). In another experiment, ILD or ITD varied parametrically across brief (1-s) presentations, and responses measured from continuously-acquired images using an “event-related” paradigm. Whole-head echo-planar imaging (~3x3x3 mm resolution) was conducted at 3T (Philips Achieva), with sounds presented via insert earphones (Sensimetrics). Blood-oxygenation-level-dependent (BOLD) signals were analyzed on each individual’s cortical surface using FSL, Freesurfer, and MATLAB. In this presentation, results will be discussed in terms of (1) the tuning of BOLD responses to ILD and ITD; (2) the relationship between tuning to ILD and tuning to monaural intensity; (3) ILD-related information as assessed using multi-voxel pattern analysis; (4) adaptation of the BOLD response as a function of trial-to-trial variation in binaural cues; and (5) whether the BOLD responses reflect physical versus perceptual (e.g., perceived location or loudness) aspects of auditory experience. [Supported by NSF DBI-0107567, NIH R03-DC009482-02S1, R01-DC011548, T32-DC005361]

9:00

3aPP3. Decoding feature information in human auditory cortex—A comparison of auditory perception, short-term memory and imagery. Annika Carola Linke and Rhodri Cusack (The University of Western Ontario, The Brain and Mind Institute, London, ON N6A SB7, Canada, alinke2@uwo.ca)

The flexible nature of auditory cortex, the complexity of real-world sounds, and limitations of the methods for neural measurement in humans have made it difficult to investigate auditory feature information processing in the human brain. Which precise features are encoded in auditory cortex and the roles they play in different cognitive tasks remained unclear. New methods for functional magnetic resonance (fMRI) provide solutions to these limitations. New acquisition sequences make less noise, improving the suitability of fMRI for auditory research. Real-time adaptive fMRI and multivariate pattern analysis methods are robust to individual differences in anatomy and exploit information in distributed neural networks. They allow assessment of which acoustic and abstracted features of simple and natural sounds are represented in human auditory regions during perception, and cognitive tasks such as change detection and imagery. The results indicate that auditory cortex is recruited for processes beyond analyzing simple feature information, playing an important role in maintaining sounds in short-term memory and encoding abstracted information during imagery. This research was supported by the Medical Research Council (UK) and The Brain and Mind Institute, University of Western Ontario.

9:20

3aPP4. Hemispheric specialization in auditory processing of Chinese lexical tones: a study using whole-head recordings of EEG.
Xiao-Dong Wang, Feng Gu, and Lin Chen (University of Science and Technology of China, Hefei 230027, China, wxd@mail.ustc.edu.cn)

It has long been established that the left hemisphere is specialized for speech whereas the right for music. However, it remains elusive whether the labor division between the two hemispheres is determined by function or acoustic property of stimuli. The confusion in the literature raises a possibility that the two factors are involved at different levels of auditory perception, respectively. In the present study, we demonstrate the dependence of hemisphere specialization on acoustic properties of stimulus in early auditory processing. We frequently presented to Mandarin Chinese speakers a meaningful consonant-vowel syllable and infrequently varied either its lexical tone or initial consonant to result in changes in word meaning. The lexical tone contrasts evoked a stronger pre-attentive electric response, as revealed by whole-head recordings of the mismatch negativity, in the right hemisphere than in the left but the consonant contrasts produced an opposite pattern. This hemisphere dominance was acoustically dependent since lexical tones and consonants in Chinese have an equal function in defining word meaning but have distinct spectral and temporal features. Our finding suggests that dominant involvement of functional cues in hemisphere specialization is only possible at a higher level of auditory processing.

9:40

3aPP5. Benefit and limitation of combining MEG and fMRI to study correlates of perception in the auditory cortex. Alexander Gutschalk (Department of Neurology, University of Heidelberg, Im Neuenheimer Feld 400, 69120 Heidelberg, Germany, Alexander.Gutschalk@med.uni-heidelberg.de), Iris Steinmann, Katrin Wiegand, and Andrew Dykstra

MEG is a strong tool for cognitive hearing research, because it allows for studying the brain's function in silence and with high temporal precision. However, MEG's ability to clearly separate neighboring sources, e.g. primary and secondary auditory cortex, is limited. While fMRI is more precise with respect to the latter requirement, it lacks the previous two advantages. Here we will demonstrate two examples of combined MEG and fMRI: The first example is perceptual awareness under informational masking. MEG revealed a distinct negative response that is evoked when listeners indicated to be aware of targets in presence of an informational masker, but not when targets were successfully masked. Earlier activity from primary auditory cortex was evoked by all targets and was independent of perceptual awareness. Nevertheless, fMRI shows that activity related to awareness is, at least in part, also generated in the primary auditory cortex. The second example is periodicity pitch; while MEG has consistently revealed responses related to periodicity and supposedly pitch in lateral Heschl's gyrus, fMRI results have been inconsistent. In a direct comparison, we could show that fMRI activity is not predominantly related to the evoked pitch response in MEG, demonstrating limitations for the combination of the techniques.

10:00

3aPP6. Hijacking gamma oscillations during auditory attention. Barbara Shinn-Cunningham (Center for Computational Neuroscience and Neural Technology, Boston University, 677 Beacon St., Boston, MA 02215, shinn@cns.bu.edu), Hari Bharadwaj (Biomedical Engineering, Boston University, 677 Beacon St., Boston, MA 02215), and Adrian KC Lee (Speech and Hearing Sciences, Portage Bay Bldg. Room 206, University of Washington, Box 357988, Seattle, WA 98195)

Neuroelectric imaging methods (electroencephalography and magnetoencephalography) can reveal cortical activity phase locked to amplitude modulation in sensory inputs for frequencies up to about 100 Hz. The neural response at the input modulation frequency is known as the visual or auditory steady-state responses (VSSR or ASSR, depending on input modality). In vision, VSSR strength is modulated by attention: energy in the frequency modulating an attended object is enhanced, while the VSSR to a distracting object is suppressed. However, in the literature, attention causes less consistent effects on the auditory steady-state response. We combined M/EEG to study how the ASSR is modulated when listeners focus spatial attention on one of two speech streams. We find that attention enhances the ASSR power at the frequency of an attended stream in auditory cortex contralateral to the attended direction. The attended-stream modulation frequency also drives phase-locked responses in left, but not right precentral sulcus (which is associated with control eye gaze and spatial attention). This asymmetric activation of the attentional network helps explain seemingly contradictory results of past auditory studies, most of which used dichotic rather than binaural simuli and analyzed results in sensor space or assumed particular dipole solutions rather than doing whole-brain analysis.

Contributed Papers

10:20

3aPP7. Cortical activation during the perception of intelligible and unintelligible speech as measured via high- density electroencephalography. Rene Utianski (Arizona State University, P.O. Box 870102, Tempe, AZ 85287-0102, rutiansk@asu.edu), John Caviness (Mayo Clinic Scottsdale, 5777 East Mayo Boulevard, Phoenix, AZ 85054), Julie Liss (Arizona State University, P.O. Box 870102, Tempe, AZ 85287-0102), Andrew Lotto (University of Arizona, P.O. Box 210071, Tucson, AZ 85721-0071), and Kaitlin Lansford (Arizona State University, P.O. Box 870102, Tempe, AZ 85287-0102)

High-density electroencephalography (EEG) was used to evaluate cortical activity patterns during the auditory processing of speech presented at various levels of degradation in a sentence verification task. 25 healthy participants listened to true-false sentences produced with one of 3 channel levels (1, 6, 16), ranging from unintelligible, moderately intelligible, and highly intelligible. Behavioral data were collected via button press (reaction time and accuracy) for each sentence. The analysis of cortical activation patterns includes 1) the identification of independent components that account for variations of activity associated with degradation levels and 2) the examination of associated event-related potentials. Statistical analyses will be performed on individuals' cortical activity, in tandem with behavioral data. Preliminary results reveal differences in the timing and magnitude of frontal and temporal lobe activation that coincide with task difficulty. Results of the present investigation will inform hypotheses about regions of interest for further investigation and will bear on models of connected speech perception, including the neurobiological underpinnings of individual differences and manifestations in a listener's understanding of degraded speech.

10:40–11:00 Break

11:00

3aPP8. Cortical activation associated with the unmasking effect of perceived separation on speech perception. Huahui Li (Department of Psychology, Speech and Hearing Research Center, Peking University, Beijing 100871, huahui.li.2319@gmail.com), Lihua Mao (Department of Psychology, Peking University, Beijing 100871), Xihong Wu, and Liang Li (Department of Psychology, Speech and Hearing Research Center, Peking University, Beijing 100871)

Spatial separation between the sound images of the target speech and the masker can improve recognition of the target speech, especially when the masker is irrelevant speech. In the present study, functional magnetic resonance imaging (fMRI) was used to investigate the neural basis of the unmasking effect of the perceived spatial separation. Target sentence was presented along with either steady-state speech-spectrum noise or irrelevant 2-talker speech at the signal-to-noise ratio of -8 dB through headphones. Position of the target image and that of the masker image were manipulated separately by modulating the time interval between the left and the right headphones, thus the two images were either co-located or separated. Sparse temporal sampling was used to avoid the influence of scanner noise. The results show that bilateral superior temporal gyrus (STG) activation was larger under the speech-masking condition relative to the noise-masking condition. When the masker is speech but not noise, the target-masker co-location was associated with more activation in left post parietal lobe and bilateral precentral/postcentral gyrus relative to the target-masker separation. The results suggest that more demand of both attention resource and central processing are required when the target speech is perceived as co-located with the speech masker.

11:20

3aPP9. Behavioral and physiological measure for pitch matching between electrical and acoustical stimulation in cochlear implant patients. Chin-Tuan Tan, Benjamin Guo (New York University, School of Medicine, Department of Otolaryngology, 550 First Avenue NBV 5E5, New York, NY 10016, Chin-Tuan.Tan@nyumc.org), Brett Martin (City University of New York, Speech and Hearing Sciences, 365 Fifth Avenue, Room 7107, New York, NY 10016), and Mario Svirsky (New York University, School of Medicine, Department of Otolaryngology, 550 First Avenue NBV 5E5, New York, NY 10016)

This study examines behavioral and physiological measures of pitch matching in cochlear implant (CI) users who have residual hearing in the contralateral ear. Subjects adjusted the frequency of an acoustic tone to match the pitch percept elicited by electrical stimulation in the other ear, when stimulation was alternating across two ears. In general, the selected acoustic frequencies did not line up perfectly with the center frequencies of the analysis bands corresponding to each stimulation electrode. Similar alternating electro-acoustic stimuli were used to record Auditory Evoked Potentials on 8 NH subjects and 3 CI patients. NH subjects were presented with a fixed tone in one ear, while tones in the other ear varied within a few octaves from the fixed tone. CI patients were stimulated with six different audible tones including their pitch-matched tones, while receiving electrical stimulation in the electrode. N1 latency for NH subjects was minimized when the same frequency was presented to both ears. Similarly, N1 latency for CI patients who are able to pitch match was minimized when the tone was at the pitch matched frequency of the stimulated electrode. These results suggest that N1 latency can be a possible objective measure of pitch matching. (Work supported by NIH/NIDCD 1K25DC010834-01;PI:Tan, PSC-CUNY;PI:Martin, and NIH/NIDCD R01-DC03937;PI:Svirsky.)

11:40

3aPP10. Cortical and brainstem differentiation of neural responses to hits versus misses. Scott Bressler (Boston University, Auditory Neuroscience Laboratory, 677 Beacon Street Boston, MA 02215, scbressler72@gmail.com), Vincent Lin, Diana Aksyonova, and Barbara Shinn-Cunningham (Boston University)

Near threshold, the same physical stimuli sometimes are perceived, but sometimes go unheard. Past visual studies have shown cortical evoked response potentials (ERPs) recorded via electroencephalography (EEG) are greater when an input is perceived than when it is not detected. Here, this approach was extended to ask whether there are detectable differences in neural activity in response to detected and undetected sounds, and whether these differences are present not only cortically, but also even earlier in the auditory system. Listeners are asked to detect the presence of a threshold-level tone complex presented in noise, presented amidst "no tone complex" catch trials. Tone-present trials are binned based on perceptual outcome ("hits" versus "misses"). Cortical ERPs from "hits" trials were found to be more robust than ERPs from "misses." Moreover, post-stimulus alpha activity (8-12 Hz) differs; specifically, "misses" showed sustained alpha activity beginning 200 ms post-stimulus, while "hits" showed an amplitude decrease in alpha around 250-300 ms. FFRs from the same EEG data for "hits" and "misses" are being compared to determine whether perceptual differences are observable early in the auditory processing stream.

12:00

3aPP11. Feature-specific cortical plasticity after rapid perceptual learning during speech segregation: a MEG study. Yi Du (Department of Psychology, Speech and Hearing Research Center, Key Laboratory on Machine Perception (Ministry of Education), Peking University, Beijing 100871, China, duyi23@126.com), Yu He (Rotman Research Institute, Baycrest Centre for Geriatric Care, Toronto, Ontario, M6A 2E1, Canada), Xihong Wu (Speech and Hearing Research Center, Key Laboratory on Machine Perception (Ministry of Education), Peking University, Beijing 100871, China), Liang Li (Department of Psychology, Speech and Hearing Research Center, Key Laboratory on Machine Perception (Ministry of Education), Peking University, Beijing 100871, China), and Claude Alain (Rotman Research Institute, Baycrest Centre for Geriatric Care, Toronto, Ontario, M6A 2E1, Canada; Department of Psychology, University of Toronto, Ontario, M8V 2S4, Canada)

This study examined whether 1-hour perceptual training could elicit feature-specific improvement of performance and corresponding cortical plasticity in humans during speech segregation by using magnetoencephalography (MEG). One group of participants learned to segregate concurrent vowels by using difference in fundamental frequency (f_0) while the other group learned to use difference in sound location. MEG recordings were conducted after the training and required participants to identify the two different vowels, which may have the same f_0 and location or differ in f_0 only, location only or both f_0 and location. Compared to Control Group who didn't receive pre-scan training, the trained groups showed behavioral improvements specific to the trained cues which were paralleled by feature-specific changes on brain activities. That is, f_0-difference-induced changes in dipole source-waveforms in auditory cortex were only modulated in Frequency Group, while location-difference-induced changes were only modulated in Location Group. Furthermore, Frequency Group showed stronger activations in auditory "what" pathway than Location Group when processing f_0-difference, while Location Group revealed stronger activation in auditory "where" pathway than Frequency Group when processing location-difference. The double-disassociation in both behaviors and neuromagnetic responses indicates that rapid perceptual learning could elicit highly feature-specific plasticity in human cortex during speech segregation.

12:20–12:40 Panel Discussion

Session 3aSA

Structural Acoustics and Vibration and Noise: Aircraft Cabin Noise

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Invited Papers

8:20

3aSA1. Prediction method of aircraft interior noise and validation testing. Haoming Qin and Kai Pan (Aircraft Strength Research Institute, Xian City, 710065, China, qinhming@163.com)

VA-one software is based on Statistical Energy Analysis theory. In this paper, acoustic modeling and prediction of interior noise of Fuselage Acoustic Research Facility (FARF) are accomplished with this software. With its support, the subsystem partition method and modeling method of airframe are proposed; further validation by testing method is also studied. With several different kinds of sidewall structures, the FARF is excited by speaker arrays and shakers while acoustic and vibration energy were measured and added into model as input. The interior damping loss factor was obtained by reverberant time measurement. Then the sound pressure level was predicted. Good coherent between prediction and measurement was obtained while methods of testing and data processing are validated. All these are very important to improve the capability of prediction of interior noise of aircraft.

8:40

3aSA2. Sound random incidence absorption characteristics of a thin plate with PZT shunted with passive electrical circuits. Chen Xiang (IACAS, chenxiang@mail.ioa.ac.cn)

Sound absorption of a thin plate with PZT shunted with passive electrical circuits in diffuse field Sound absorbing properties of a plate with piezoelectric shunted circuit in diffuse field is investigated in this paper. A theoretical model of a thin plate with piezoelectric patch and electrical circuit under random sound incidence is proposed and numerically analyzed. Measurement is also carried out in a reverberation room and found to be in good agreements with the theoretical model. Theoretical and measured results confirmed that this technology has a potential to improve sound absorption at low frequency

9:00

3aSA3. Cabin noise controlling on MA600. Feng Hou and Qun Yan (Aircraft Strength Research Institute, 710065, China, carsper623@gmail.com)

MA600 is one turbo-propeller driven regional aircraft modified from MA60, which has considerable high interior noise level. Project on MA600 cabin noise reduction involve numbers flight test, huge laboratory measurement campaign as well as numerical simulations. Based on the understanding of noise sources and their transmission path learned from both test and simulation, an integrated acoustic treatment plan was drawn and implemented on the MA600 to reduce the interior sound pressure level while weight reduction was also archived.

9:20

3aSA4. Near-field effect as sound transmission through a panel-cavity system. Bilong Liu (The Key Laboratory of Noise and Vibration, Institute of Acoustics, Chinese Academy of Sciences, 100190, Liubl@mail.ioa.ac.cn)

Evidence from measurement indicates that sound pressure level inside cabin is dominant for the seats close to the windows. It could be 3~6 dB(A) higher than that of the seats close to the corridor for a commercial aircraft during cruising. This near-field phenomenon of transmitted sound is modeled and investigated by the sound transmission through a panel-cavity system. The distribution of transmitted sound pressure level behind the flexural panel is calculated and the effect of local absorption layer behind the flexural panel is specially included for a comparison. Simple measurement is also carried out to show that the size and thickness of local absorption layer is sensitive to the near-field of transmitted sound.

9:40

3aSA5. The experimental research of acoustic property of anti-ice shield for propeller aircrafts. Kai Pan and Haoming Qin (Aircraft Strength Research Institute, Xian City, 710065, China, pankai.er@126.com)

For propeller driven aircraft, the structural noise caused by periodic aerodynamic load on the fuselage hence transmitted into cabin is dominating, especially on the propeller rotating plane. The anti-ice shield is effective against such. Through experimental research, the sound isolation property of the anti-ice shield from one particular aircraft was improved, and the routine of acoustic property optimization was summarized.

10:00

3aSA6. The aircraft acoustics research and achievements of AVIC. Wenchao Huang and Hai Yang (Aircraft Strength Research Institute, Xian City, 710065, China, va-623@163.com)

Passing through decades of research, development, applications and improvements, the AVIC has sent their ARJ-21 and MA-600 for the first flight and drawing greater prospect for future. The acoustic research for aircraft in AVIC mainly focus on three aspects: interior noise environment for passenger comfort, exterior noise emission for mandatory airworthiness regulation and sonic fatigue for safety issue, while integrated design strategy, optimization routine and validation procedure were developed, supporting aircrafts development and aiming the future.

Contributed Papers

10:20

3aSA7. Measurement of the noise radiation of the panels subjected to turbulent boundary layer pressures. Daoqing Chang (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, changdq@mail.ioa.ac.cn), Xiang Chen (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China), Zhongchang Qian (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China), and Bilong Liu (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China)

The noise radiation of the panels with different treatments under a turbulent boundary layer (TBL) excitation is tested in an anechoic wind tunnel. The averaged dimensionless spectrum for the response and the radiated sound power of panels with different size, stiffening and damping treatments are collected at the condition of various flow speeds. Numerical analysis of panel parametric sensitivity on the radiated noise due to TBL excitation is also presented for a comparison.

10:40–11:00 Break

11:00

3aSA8. Prediction and measurement of acoustical property for natural fiber reinforced composite laminates. Weidong Yang, Yan Li, and Yongdong Pan (Tongji University, 200092, kevin_yangwd@hotmail.com)

Owing to the advantages of low density, low cost, easy availability and biodegradability, natural fibers are promising for industrial use in fiber-reinforced composite materials. Porous structures of natural fibers greatly contribute to the excellent sound absorption properties and multilayered composite panels also possess good sound insulation. Natural fiber reinforced composite (NFRC) laminates were manufactured in view of the process requirements of industrial production. Acoustic properties of NFRC, such as absorption coefficient, transmission coefficient and transmission loss, were measured by transfer function method in four-microphone impedance tube. Moreover, a theoretical model of multilayer material system of NFRC was proposed and the predicted results calculated by transfer matrix method are consistent with experimental measurement. This model can be used to predict acoustic properties of multilayered laminate system of NFRC.

11:20

3aSA9. Sound transmission through a double panel structure jointed with rubber isolators. Zhongchang Qian, Daoqing Chang, and Bilong Liu (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, qianzhongchang@mail.ioa.ac.cn)

Sound transmission through an aircraft sidewall structure is investigated theoretically and experimentally in this paper. The studied configuration is composed of double-leaf lightweight partitions jointed with discrete placed rubber isolators. The effects of isolator stiffness, damping and spacing on sound transmission loss are calculated and tested. The influence of acoustic absorption treatment filled in the cavity is also included for a comparison. Furthermore, two methods which refers to space-harmonic and modal superposition expansions are employed for infinite and finite dimensions. At last, some approaches for sound transmission loss optimization are proposed.

11:40

3aSA10. Civil aircraft cabin noise resource acoustic characteristic and transmission path analysis. Qian Shi (The Olympic Building, No. 267, North 4th Ring Road, Haidian Disctrict, Beijing 10083, P.R. China, shiqian@comac.cc)

Typically, there are mainly 4 different noise resources critical to cabin +F1184noise: auxiliary power unit (APU) noise, environment control system (ECS) noise, engine noise and turbulent boundary layer (TBL) noise. Because of the different acoustic characteristic and transmission path for each resource, their impacts to cabin noise level are not the similar. A vibration and noise test under ground and flight status of an in-service civil aircraft was conducted. Based on the test results, comparing the data under different test status, the acoustic characteristic and transmission path are analyzed for the 4 noise resources in this paper, including distribution characteristic, spectrum characteristic and transmission path. APU noise mainly affects the rear fuselage, ECS noise transmits by ducts, engine noise and TBL noise transmit through side panel.

12:00

3aSA11. The study on sonic response analysis method for civil aircraft structure. Yu Wang, Jiazhen Zhang, Xiaoling Zheng (Beijing Aeronautical Science & Technology Research Institute of COMAC, Beijing 100083, China, wangyu2@comac.cc), and Jun Liang (Center for Composite Materials and Structure, Harbin Institute of Technology, Harbin 150001, China)

Sonic fatigue problem is one of the important terms of civil aircraft airworthiness examination. In this paper, two different engineering analysis methods are employed to study the sonic response characteristics of a kind of typical commercial aircraft structure (skin-string-frame structure). Whereafter the sonic response experiment for this structure is carried out, and the resonance frequency and root mean square (RMS) stress are measured. According to the comparison of engineering analysis results and experimental results, the analysis error and effectiveness of these two methods are studied. The resonance frequency analysis results for both methods are in good agreement with experimental results, and the prediction results of RMS stress are also acceptable in engineering.

Session 3aUWa

Underwater Acoustics: Propagation

Megan Ballard, Cochair
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Zhaolu Peng, Cochair
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Contributed Papers

8:20

3aUWa1. Acoustic wavefront tracing in range-dependent ocean. Oleg A. Godin (CIRES, University of Colorado at Boulder and NOAA/Earth System Research Laboratory/Physical Sciences Division, Mail Code R/PSD99, Boulder, CO 80305-3328, Oleg.Godin@noaa.gov), and Nikolay A. Zabotin (CIRES, University of Colorado at Boulder, Boulder, CO 80309)

It has been established experimentally and confirmed by numerical simulations that early arrivals of acoustic waves at long-range propagation in a deep ocean are stable and identifiable despite strong perturbations of the ray paths due to sound-speed fluctuations primarily induced by internal gravity waves. It is wavefronts rather than rays that are typically observed in underwater acoustic experiments. Wavefronts are much more stable with respect to environmental perturbations than individual rays, which form the wavefronts. The relative stability of the wavefronts takes place because scattering of the end points of rays resulting from weak environmental perturbations occurs primarily along wavefronts of the unperturbed wave with the same travel time [O. A. Godin, J. Acoust. Soc. Am. **122**, 3353–3363 (2007)]. When wavefronts are much more stable than rays, the traditional approach, which relies on ray tracing to determine wavefronts' position, may be counterproductive and sometimes misleading, especially for highly structured environments such as the ocean with internal waves and "spice." This paper presents an efficient numerical technique for modeling acoustic wavefronts and timefronts in range-dependent ocean without solving ray equations. The acoustic wavefront tracing code has been benchmarked using analytic solutions of the eikonal equation. [Work supported by ONR.]

8:40

3aUWa2. Spatiotemporal variability of underwater sound fields near steep slopes. Timothy F. Duda (Woods Hole Oceanographic Institution, AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), Ying-Tsong Lin (Woods Hole Oceanographic Institution, AOPE Dept. MS 11, Woods Hole, MA 02543), Weifeng Gordon Zhang (Woods Hole Oceanographic Institution, AOPE Dept. MS 12, Woods Hole, MA 02543), Aurelien Ponte, and Bruce D. Cornuelle (Scripps Institution of Oceanography, University of California, San Diego, 9500 Gilman Drive, La Jolla, CA 92093-0209)

Studies using idealized bathymetry have shown that steep slopes can have important effects on the geometric structure of ocean sound fields. Other studies have provided insight into the temporal variability of sound fields in the moving ocean. There is a general understanding that both effects occur together in the ocean, but their relative importance is not fully known, nor is the way they interact. The processes are often linked because slopes can generate strong internal tides and waves. Here, the coupled spatial and temporal variability is investigated using time-stepped three-dimensional (3.5-D) sound propagation modeling. The needed ocean sound-speed information is generated using modern regional ocean modeling. Solutions of the 3D parabolic acoustic wave equation are generated for a few hundred hertz. The spatial patterns of sound reveal that temporal intensity variation (scintillation) caused by internal tides can be enhanced at areas where sound encounters steep slopes. The magnitude of the effect is measured by repeating the simulations with a smoothed seafloor substituted for the realistic seafloor. The effect of internal tide strength on the interaction is quantified similarly. The effects of tide strength and seafloor steepness on spatial patterns of the sound field horizontal correlation length are also investigated.

9:00

3aUWa3. Acoustic propagation modeling in environments which induce horizontal refraction and mode coupling. Megan Ballard (Applied Research Laboratories at the University of Texas at Austin, P.O. Box 8029, Austin, Texas 78713-8029, meganb@arlut.utexas.edu)

In typical applications of modeling underwater sound propagation, three-dimensional effects are assumed to be relatively weak and two-dimensional models are applied on a vertical plane to predict acoustic signals. However, this assumption breaks down for many shallow-water environments. For example, evidence of horizontal refraction has been documented in the context of nonlinear internal waves and sloping bathymetry. In this work, three-dimensional effects are modeled using a coupled-mode technique which includes the effects of out-of-plane scattering. Several examples of propagation in three-dimensional environments will be presented. A decomposition of the field into modal amplitudes will be used to identify features in the environment responsible for the observed effects on the acoustic field. [Work supported by ONR]

9:20

3aUWa4. The effect of random bottom bathymetry on mode structure and coherence in shallow water propagation. Jennifer Wylie and Harry DeFerrari (University of Miami—RSMAS Division of Applied Marine Physics, 4600 Rickenbacker Causeway, Miami, Fl 33149, jennie.wylie@gmail.com)

Ideal flat bottoms in shallow water propagation channels produce predictable surface-reflected-bottom-reflected (SRBR) mode structures. Further, modes have predictable group velocities so that arrivals patterns for pulse transmissions can be identified by travel time. But bathymetry is rarely truly flat to a fraction of an acoustic wavelength over ranges of a few km. Here, we examine the mode pattern deviation from the flat bottom case as random variations in bathymetry are introduces. A PE propagation model with range dependent bottom reveals that mode structures become distorted with increasing fluctuation in bathymetry. Higher order modes deteriorate first and the fundamental mode propagates longest without distortion. For higher order modes, RMS fluctuations greater than $\frac{1}{2}$ the acoustic wavelength destroy recognizable mode structure as compared to the smooth bottom case. If the fluctuations are further increased to 1 wavelength, discreet modes give way to continuous ones and energy is smeared in space and arrival time and mode arrivals become incoherent and undetectable with the commonly used phase coherent signal processing methods. The coherence of modes has implications on both temporal and spatial coherence. The

models are used here to predict the frequency/depth/range limits on coherence for simple statistical description of bottom bathymetry fluctuations.

9:40

3aUWa5. Analysis of acoustic fluctuation of the different tracks by the internal waves. Fan Li (No. 21, Beisihuanxi Road, Institute of Acoustics, Chinese Academy of Science, Beijing 100190, China, lifanyuxin@sohu.com), Xinyi Guo, Tao Hu, Li Ma, and Yaoming Chen

Internal waves in shallow-water cause sound speed profiles variations and acoustic transmission abnormity. The acoustic data show arrival time variations and mode coupling phenomena. The 2009 experiment collected high quality environmental and acoustic data and were used to analyse the effect of internal waves on the acoustic transmission. One notable feature of the experiment is that internal crossed two tracks at different incidence angles. The paper compare acoustic fluctuation character at two different tracks and use mode filter method to investigate the mode coupling caused by internal wave. The correspondence between data and simulation results show that the alternate of mode 2 and mode 1 were observed at the track which is vertical with the internal wave direction. The alternate of Mode 3 and first two modes occur at the another track.

10:00

3aUWa6. The mode coupling and effect on ambient noise vertical directionality caused by internal waves in shallow water. PengFei Jiang, Jian-Heng Lin, XueJuan Yi, and Guojian Jiang (Qingdao Lab of Institute of Acoustics, Chinese Academy of Sciences, No. 8, Shangqing Road, Shibei District, Qingdao 266023, Shandong Province, China, joyouc@126.com)

Experiments show that internal waves can cause the fluctuation of the depth of thermocline in shallow water, which may bring about two results: one is the change of hit number of sound rays and sea bottom, the other is mode coupling. The former changes the sound intensity, and the latter changes the sound energy distribution. The previous studies show that internal waves cause mode coupling by changing the sound propagation paths, which can fill the noise notch. In the calculation, it is found that the notch is deepened when some solitary internal waves are present. The reason is that the sound wave excited by the distant sources which are located on the palne near the sea surface propagates through the soliton which is a single downward undulation. The mode coupling from low-order modes to higher-order modes happens first, then the reverse conversion occurs. However, the former coupling is stronger, the lower-order modes fall off and the filling ability is weakened, so the notch is deeper.

10:20

3aUWa7. Passive measurement of lengthways transfer function in ocean waveguide using ambient noise. Xinyi Guo (Key Laboratory of Underwater Acoustics Environment, Institute of Acoustics, Chinese Academy of Science, Beijing 100190, China, guoxinyi@mail.ioa.ac.cn), Fan Li, Li Ma, and Yaoming Chen

This paper introduces a function of correlation between two hydrophones, basing on the Kuperman-Ingenito ocean ambient noise model. There is a similarity in form between the correlation function and the transfer function in ocean waveguide from a point source to a receiver. Thus, the noise correlation function between two hydrophones in vertical location can extract actual transfer function, then the acoustics line arrival structure of propagation in lengthways waveguide can be analyzed. In this paper, the transfer function in lengthways ocean waveguide can be obtained from broadband ambient noise correlation function of vertical line array. There are some analyses about physical significance of noise interference basing on compared simulation and experience. This method can be used to research layered sea floor considering the arrival time structure of each propagation route.

10:40–11:00 Break

11:00

3aUWa8. Long range propagation modeling of offshore wind turbine noise using finite element and parabolic equation models. Huikwan Kim, Gopu R. Potty (Department of Ocean Engineering, University of Rhode Island, Narragansett, RI 02882, hkkim524@my.uri.edu), James H. Miller (NATO Undersea Research Centre, La Spezia, Italy), Kevin B. Smith (Department of Physics, Naval Postgraduate School, Arlington, VA 22203), and Georges Dossot (Department of Ocean Engineering, University of Rhode Island, Narragansett, RI 02882)

Noise generated by offshore wind turbines and support structures radiates and propagates through the air, water and sediment. Predicting noise levels around wind turbine structures at sea is required to estimate the effects of the noise on marine life. We used Finite Element (FE) and Parabolic Equation (PE) models to predict long range propagation of noise from the construction and operation of offshore wind turbines. FE analysis produced pressure outputs at short ranges were used as a starting field for a modified PE propagation model. Furthermore, we investigated the optimum range for the transition to PE modeling. The effects of various sediment types were also considered determining the pressure starting field. In FE analysis models, we implemented the axisymmetric elements and implicit dynamic analysis with pressure impact loading and vertical acceleration boundary conditions to simulate pile driving and operational noise radiation. We will present the PE long range pressure field outputs from the offshore pile driving and operation for a shallow water environment around Block Island, Rhode Island.

11:20

3aUWa9. Quantitative predictions of impact pile driving noise in water using a hybrid finite-element propagation model. Mario Zampolli, Marten J.J. Nijhof, Michael A. Ainslie, Christ A.F. De Jong, Erwin H.W. Jansen (TNO Acoustics and Sonar Group, 2597 AK The Hague, The Netherlands, mario.zampolli@tno.nl), and Ahmad T. Abawi (Heat Light and Sound Research Inc., La Jolla, CA 92307, U.S.A.)

The quantitative prediction of noise radiated into the underwater environment during offshore impact pile driving is a topic of major interest to the underwater acoustics community. This problem is addressed by using a hybrid model consisting of a high fidelity local model describing the vibrating pile and its vicinity, and an efficient model for the propagation of the sound to large distances from the pile in the underwater waveguide. The local model is an axisymmetric linear frequency domain structural acoustic finite-element model, and the propagation model is based on ring-source Green's functions in the water and in the sediment surrounding the pile, obtained by horizontal wavenumber integration. The coupling between the two models is achieved via the Helmholtz integral, in which the pressure and normal displacement in the media surrounding the pile are obtained by sampling the finite-element model results and the Green's functions are obtained from the propagation model. The simulation results are compared to measurements conducted during pile driving operations, showing that the quantitative prediction of the sound exposure levels with such a tool is possible.

11:40

3aUWa10. Air-to-water sound propagation at low frequency in shallow water. Zhaohui Peng and Guangxu Wang (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, pzh@mail.ioa.ac.cn)

Owing to great difference of acoustic characteristic impedance between air and water, sound transmission loss from an airborne source into underwater is very high. So, it is very difficult to do experimental research on air-to-water sound propagation. High power of air source is needed to receive signals with high signal-noise-ratio at long range. A fog whistle was used as the source during an experiment conducted for air-to-water sound propagation in the South China Sea in July 2011. A HLA was laid on the sea bottom

at the depth of 90m. The high-power fog whistle was hung on a research ship floating near the HLA. The maximum distance between the whistle and the HLA was 7.5km. The transmission at low frequencies were analyzed and compared with the theoretical predictions.

12:00

3aUWa11. Environmentally robust time reversal processing based on multiple constraint method. Jae Hoon Joo (LIG Nex1, 702, Sampyeong-dong, Bundang-gu, Seongnam-city, Gyeonggi-do, Korea, joojaehoon@lignex1.com), Jea Soo Kim, Jung Hong Cho (Korea Maritime Univ., Dept. of Ocean Engineering, Korea Maritime Univ., 727, Yeong-do), and Sang Taeck Moon (LIG Nex1, 702, Sampyeong-dong, Bundang-gu, Seongnam-City, Gyeonggi-do, Korea)

Time-reversal processing (TRP) uses the signals from a probe source to refocus the signal at the probe source location by back-propagating the time-reversed version of the received signal. However, the performance of TRP can be degraded by the mismatches due to the non-static environment where propagation condition changes during the time between receiving a probe signal and back-propagating the time reversed signal. In this presentation, two methods for the environmentally robust TRP based on the multiple signal vectors with singular value decomposition and minimum variance with multiple constraint method has been numerically simulated and compared under the condition of mismatches. Multiple sound speed profile, probe source location and tilted array are used to make the TRP robust. The results show that the focal point using robust TRP is well maintained when compared to the conventional TRP. Both methods are efficient for the robustness of time varying environment.

WEDNESDAY MORNING, 16 MAY 2012 S426 + S427, 8:20 A.M. TO 11:20 A.M.

Session 3aUWb

Underwater Acoustics: Target Scattering

Aubrey Espana, Cochair
aespana@apl.washington.edu

Mario Zampolli, Cochair
mario.zampolli@gmail.com

Contributed Papers

8:20

3aUWb1. Research on recognizing a cylindrical object using the 3-D formation from highlight and shadow of side scan sonar image. Jieun Lee (Soongsil Univ., 525 Hyoungnam Engineering Dept., Soongsil Univ., Sangdo-dong, Dongjak-gu, Seoul, oldfashion04@naver.com), Taebo Shim, Taejung Kim (Soongsil Univ., 525 Hyoungnam Engineering Dept., Soongsil Univ., Sangdo-dong, Dongjak-gu, Seoul), and Jooyoung Hahn (ADD, P.O. Box 18, Jinhae, Kyuongnam 645-600, Korea)

There are many researches on recognizing and classifying a cylindrical object using side scan sonar images. In this paper we segment object, shadow of object and background of the images in order to extract the cylindrical object from the segmented object information and figure out whether it corresponds to the cylindrical object what we want to find. Shadow of cylindrical object estimated from 3-dimensional segmented space is compared with the shape of shadow got from morphological method. Many objects of different shape such as cylindrical, cylindrical object-likes and non- cylindrical objects are used as a background objects to simulate the objects are mixed with similar objects. Result shows that more than 90 % of the objects were recognized as objects in a comparably short time of order of 1 second.

8:40

3aUWb2. Submerged target scattering: comparison of combined finite element/simplified acoustics models to data. Kevin Williams, Aubrey Espana, Steven Kargl (Applied Physics Laboratory, University of Washington, Seattle, WA 98105, williams@apl.washington.edu), and Mario Zampolli (TNO Defense, Security and Safety, Oude Waalsdorperweg 63, P.O. Box 96864, 2509 JG The Hague, Netherlands)

The environment and the location of the target within that environment affect the scattering from elastic targets in ocean waveguides. Computational power is now realizable to compute the target scattering, in-situ, via finite elements. However, these calculations still require high cost computer facilities and in the end do not offer physical insight into processes involved. Here we compare two models, with different levels of simplification, to data acquired from an Aluminum target machined to replicate an Unexploded Ordnance (UXO). The first model treats the scattering using two-fluid Green's function propagators in combination with finite element calculations of the target scattering as placed within the waveguide. The second model uses free field, plane wave incidence, finite element results for the target scattering in conjunction with simple ray based propagation to account for the waveguide environment. The data/model comparisons are discussed in light of the physical insight they can help provide, the speed of the calculation and the level of fidelity they achieve. [Work supported by ONR and SERDP, USA].

9:00

3aUWb3. Low- to mid-frequency scattering from submerged targets partially buried in the sediment at an oblique angle. Mario Zampolli (TNO Acoustics and Sonar Group, 2597 AK The Hague, The Netherlands, mario.zampolli@tno.nl), Aubrey L. Espana, Kevin L. Williams (Applied Physics Laboratory, University of Washington, Seattle, WA 98015, U.S.A.), and Philip L. Marston (Washington State University, Pullman, WA 99164, U.S.A.)

The scattering from elastic targets in the low- to mid-frequency regime is affected by the environment surrounding the target. For axisymmetric targets with the axis of symmetry parallel to the water-sediment boundary, previous work has dealt with the change in the target strength as a function of frequency and aspect angle in relation to the burial depth in the sediment. The present work deals with the extension of a finite element model, based on the decomposition of the acoustic and elastic fields into azimuthal Fourier modes, to the case of a target buried at a tilt angle. The interaction between the target and the sediment is represented by the model up to the first order of the scattering series, which means that the scattering of the incident field and of the target reflected field is taken into account, but the rescattering of the boundary reflected echo from the target is neglected.

Model results up to 30 kHz are compared to experimental data for a 2 foot long aluminum cylinder of 1 foot diameter buried in sand at a tilt angle.

9:20

3aUWb4. Study on inverse problem of acoustical resonance scattering by elastic shells. Wang Fangyong (NHangzhou Applied Acoustics Research Institute, No. 96, HuaXing Road, Hangzhou, China, sklwfy@gmail.com), Cao Zhengliang, and Du Shuanping (NHangzhou Applied Acoustics Research Institute, No. 96, HuaXing Road, Hangzhou, China)

For the complexity of the analytical expressions for acoustical scattering by even the most simplistic elastic bodies such as spheres and cylinders, inversion of the exact direct problem in analytical way is almost infeasible. So, this paper mainly studies the inverse problem in a numerous way. Low-frequency(ka$<$50) resonance spectra of targets with varied parameters including longitudinal wave velocity, shear wave velocity, material density and shell thickness are theoretically calculated using Resonance Scattering Theory(RST), and in what way the elastic parameters affect the resonance signature is analyzed and concluded. The results obtained have great significance on active classification research based on acoustical resonance scattering.

9:40

3aUWb5. Predicting the acoustic response of targets in an ocean environment based on modal analysis of finite element calculations. Aubrey España, Kevin Williams (APL-UW, 1013 NE 40th St., Box 355640, Seattle, WA 98125-6698, aespana@apl.washington.edu), Mario Zampolli (TNO Defense, Security and Safety, Oude Waalsdorperweg 63, P.O. Box 96864, 2509 JG The Hague, Netherlands), and Philip Marston (Washington State University, Department of Physics and Astronomy, P.O. Box 642814, Pullman, WA 99164-2814)

Low frequency sound is a viable means for the detection of elastic targets in contact with the ocean floor. The incoming sound, with wavelengths on the order of the target dimensions, can excite resonant modes of the target leading to enhancements in the scattered field. A hybrid model has been developed to predict the acoustic scattering from cylinders, pipes and unexploded ordnance (UXO) in proud or buried configurations in the ocean sediment. The model exploits the symmetry by decomposing the 3-D problem into a sum of 2-D independent Fourier modal sub-problems. This hybrid modeling technique has been shown to agree well with experimental measurements conducted in a pond [A.L. España et al., J. Acoust. Soc. Am. 130, 2330 (2011)]. Presently, these hybrid model results are used to examine the target response on a mode-by-mode basis. A modal map is generated by keeping track of the number of dominant modes contributing to the bright features observed in the acoustic template. For features that are predominantly due to one or two modes, simple analytical models can be used to predict their evolution as a function of target/sensor geometry within the ocean waveguide. [Work supported by ONR and SERDP.]

10:00

3aUWb6. The effects of lumped mass attachments on vibro-acoustic behavior of a fluid-loaded plate. Danzhu Yu, Sheng Li (School of Naval Architecture at Dalian University of Technology, No. 2, Linggong Road, Ganjingzi District, Dalian City, Liaoning Province, 116024, P.R. China, summerydz@163.com), and Yunfei Chen (School of Naval Architecture at Dalian University of Technology & Underwater Test and Control Laboratory)

The effects of attachments on the dynamics of a master structure are of fundamental significance. In this study, the changes in the vibro-acoustic behavior of a fluid-loaded plate due to variations in lumped mass attachments are examined. The finite element for modeling the structure with lumped mass attachments is coupled with the Rayleigh integral for the acoustic fluid to solve the structure – fluid interaction problem and to obtain the response of the coupled system. The changes in the modal parameters due to the variations in attachments are determining from a model reduction method. The Monte Carlo simulation is used for the uncertainty analysis of the vibro-acoustic behavior. Both the mean and the standard deviation of the changes attributable to the attachments are discussed. A baffled plate with water loading/air loading is involved in the study. The numerical results show that the effects and the changes-in-impedance on the master plate due to the attachments vary with the frequency spectrum.

10:20

3aUWb7. Study on an underwater focused acoustic phased array and the sound fields characteristics. Xinwu Zeng (College of Opto-Electric Science and Engineering, National University of Defense Technology, Changsha 410073, China, x.w.ZENG@139.com)

Regarding to the fact that the hydrodynamic characteristics of supercavitating torpedoes strongly depend on the supercavity envelops, this paper describes a high intensity underwater focused acoustic phased array used to remotely disrupt this envelope surface. First, the demand of designing an underwater focused acoustic phased array was discussed, and then an efficient sound field calculation formula for the convex spherical-section phased array was obtained. Furthermore, the effects of element size and array aperture on the valid focusing region of above array were analyzed based on pseudo-inverse method. It is demonstrated, via simulation, that the arrays composed of minimum element size with large beam angle perform well on the intensity of focus pressure, and the valid focusing region under the similar size of emission area. However, the change of valid focusing region is small when merely enlarge the aperture of array. And the possible application of multiple-focus pattern was investigated.

10:40–11:00 Break

11:00

3aUWb8. A theoretical simulation to the arc expansion process in underwater discharge. Yibo Wang (College of Opto-Electric Science and Engineering, National University of Defense Technology, Changsha 410073, China, yibowang.nudt@gmail.com)

A one-dimensional simulation to the arc expansion process in underwater discharge is carried out in this paper towards its final destination—the estimation of the acoustic output in the arc expansion process. Theoretical models for these four sub-processes which are strongly coupled in the arc expansion process, are built up. Based on these theoretical models, a simulation code is developed and the time-evolution data of some key parameters in the arc expansion process are simulated, from which the acoustic output is finally estimated.

Awards Ceremony

Jing Tian
President, Acoustical Society of China

Mardi C. Hastings
President, Acoustical Society of America

Acoustical Society of China

Presentation of Acoustical Society of China Awards

Dah-You Maa Acoustical Award of the Acoustical Society of China

Honorary Membership in the Acoustical Society of China to Leo L. Beranek

Acoustical Society of America

Presentation of ASA Fellowship Certificates

Joel A. Lewitz
William Shofner
Annemarie Surlykke
Sten Ternstrom
Christopher O. Tiemann

Presentation of Acoustical Society of America Awards

R. Bruce Lindsay Award to Constantin-C. Coussios

Silver Medal in Engineering Acoustics to Gary W. Elko

Honorary Fellowship to Dah-You Maa

Gold Medal to William A. Kuperman

Keynote Lecture

Invited Paper

8:20

Acoustics of traditional Chinese theatres. Ji-qing Wang (Institute of Acoustics, Tongji University, Shanghai, Shanghai 200092, China)

The traditional Chinese theatre is a unique architectural form. Chinese opera is a form of imaginary performing art; therefore, it does not require large stage and realistic stage settings. A pavilion stage open on three sides and thrusting into the audience area is its commonly applied characteristic feature. A comparatively low ceiling with elegant dome-like caisson acts as a sound shell, providing beneficial reflections to the audience, and to actors on the stage as well. The older generation Chinese opera goers used the term ?]going to listen opera??which well explains how they placed great demands on vocal performance. In Chinese theatrical history, there were different types of theatre from open-air theatre to hall theatre, built both in cities and rural areas all over the country. Nevertheless, the courtyard theatre was the most popular. Up to the present day, thousands of ancient traditional theatres still exist in China, and many of them are well preserved. Interesting results are reported in this paper after acoustical surveys of these theatres. Acoustical issues are raised from these studies, such as, does the classical parameter of reverberation time still adequate for qualifying the acoustics for a roofless courtyard theatre, or for an amphitheatre as well? A primary subjective survey conducted in our laboratory recently presents the negative conclusion. Another presentation is involved in this paper: the puzzle of vase resonators beneath the traditional stage which was long believed to be effective for sound enhancement as recorded in the Chinese historical accounts. The author also gives other acoustic analysis of the kind with pictorial presentations.

THURSDAY MORNING, 17 MAY 2012 HALL A, 9:15 A.M. TO 12:40 P.M.

Session 4aAAa

Architectural Acoustics and Noise: Healthcare Acoustics

Kenneth Roy, Cochair
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Erica Ryherd, Cochair
erica.ryherd@me.gatech.edu

Jerry Li, Cochair
Jli1@armstrong.com

Chair's Introduction—9:15

Invited Papers

9:20

4aAAa1. Soundscape study for the improvement of neonatal intensive care units. Jennifer Nelson (UF School of Architecture, P.O. Box 115702, Gainesville, FL 32611-5702, jennifer.nelson@ufl.edu), and Gary Siebein (Siebein Associates, Inc., 625 NW 60th Street, Gainesville, FL 32607)

Guidelines for healthcare spaces address day and nighttime Leq and peak levels. However, there are many complex and transient sounds that make up the overall sound levels in healthcare environments. Many of these sounds contribute to the background level, while others are transient noises and alerts to professionals who must hear them to care for their patients. Unfortunately, these noisy environments are also where the patient is placed to heal. Three different Neonatal Intensive Care Units (NICUs) built in different years in Florida were observed and categories of sounds in each were documented. Overall level vs. time measurements made over a one week time period in each NICU were compared with WHO guidelines. Spectral level measurements of individual and combined sounds are also documented in each NICU. The individual sounds were classified into necessary and unnecessary criteria that orchestrate at all times of the day by observing and documenting them. The results of this study show how changes being made in the design and operation of contemporary NICU's are reflected in the measured sound levels, and what future changes can be made to further decrease unwanted noise.

9:40

4aAAa2. Evidence based design for hospital corridor noise control—Center for Health Design. Kenneth Roy (Armstrong World Industries, 2500 Columbia Ave, Lancaster, Pennsylvania 17604, kproy@armstrong.com), and Sean Browne (Armstrong World Industries, 2500 Columbia Ave, Lancaster, Pennsylvania 17604)

Armstrong participated as a member of a joint research group including the Center for Health Design and Palomar Pomerado Health in San Diego, California. The goal of this research was to evaluate the effects of flooring and ceiling choices on Corridor Activity Noise, and its perception by both patients and healthcare professionals for 2 material choices: 1. corridors with carpet and standard acoustical ceilings, and 2. corridors with hard flooring and high performance acoustical ceilings. This work was managed jointly by the PPH and CHD, the acoustic measurements were taken by CMSalter Associates, and materials and some data analyses were provided by Armstrong. Test results showed that substitution of a hard surface flooring material for carpeting resulted in a net increase in corridor noise levels with the expected patient and medical professional perceptions of increased annoyance and distraction. However, if the hard surface flooring is combined with a high performance acoustical ceiling, then the rise in noise due to the floor surface is negated with the added absorption of the improved ceiling, such that the result is equivalent in level and perception by both patients and staff.

10:00

4aAAa3. Emerging findings from the Healthcare Acoustics Research Team (HART). Erica E Ryherd (Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405, erica.ryherd@me.gatech.edu), Kerstin Persson Waye (Occupational & Environmental Medicine, Gothenburg Univ., 405 30 Gothenburg, Sweden), James E West (Electrical Engineering, Johns Hopkins Univ., Baltimore, MD 21218), Craig Zimring (College of Architecture, Georgia Institute of Technology, Atlanta, GA 30332), and Jeremy Ackerman (Dept. of Emergency Medicine, Emory Univ. School of Medicine, Atlanta, GA 30322)

Hospital patients, staff, and visitors need healthy soundscapes: patients need to sleep and heal without stress; staff, patients and family need to communicate accurately but privately; staff need to hear alarms and calls for help. Unfortunately, many hospitals are noisy and stressful places. Although there is growing and strong evidence that the hospital soundscape is problematic, there are many remaining questions and obstacles. This presentation will discuss recent case studies and findings from the Healthcare Acoustics Research Team (HART), an international, interdisciplinary collaboration of specialists in architecture, engineering, medicine, nursing, and psychology. HART is actively engaged in research in the United States and Sweden, having worked in a dozen hospitals and a broad range of unit types including intensive care, emergency, operating, long-term patient care, mother-baby, and others. HART seeks to advance the understanding of how various aspects of the hospital soundscape impact occupants, how to best measure and quantify these aspects, and how to translate results into evidence-based-design. Taken as a whole, these studies provide new insight into how to create healthier hospital acoustic climates.

10:20

4aAAa4. Experimentally investigated sleep disturbance of intensive care unit sounds. Kerstin Persson Waye (Occupational and Environmental Medicine, The Sahlgrenska Academy, Gothenburg University, Sweden, kerstin.persson.waye@amm.gu.se), Richard Wallenius (Gothenburg University, Sweden), Eva Maria Elmenhorst (German Aerospace Centre, DLR, Institute of Aerospace Medicine, Cologne, Germany), and Eja Pedersen (Department of Environmental Psychology, Lund University, Sweden)

Patients at intensive care often report fragmented sleep from noise due to care activities from personal, from other patients and alarms. The aim of this study was to explore the effects of original and modified intensive care noises on sleep in 15 healthy subjects. Their sleep was registered with polysomnography during four nights, one adaptation night, one control night and two exposed nights with similar equivalent sound levels of 47 dB LpAeq, but with either a maximum sound pressure level of 64dB LpAFmax or 56 dB LpAFmax. The subjects also answered questionnaires and saliva cortisol was sampled in the morning. The results showed that during exposure nights, subjects had less slow wave sleep and spend more time awake. No relation was found between arousals and maximum sound levels. Apart from an unexpected reduction of time in the REM-stage for the exposure with lower maximum level, there was no impact of the reduction of maximal levels for the sleep parameters recorded. The subjective data supported the polysomnographical findings while cortisol levels were not affected by the conditions. For healthy subjects the reduction of maximal levels from 64dBA to 56 dBA was not enough to improve sleep quality.

10:40–11:00 Break

11:00

4aAAa5. Speech privacy at community pharmacies. Yumi Koyama (School of Pharmacy, Nihon University, 7-24-1 Narashinodai, Funabashi, 7-24-1 Narashinodai, Funabashi, Chiba, Japan, koyama.yumee@nihon-u.ac.jp), Toshiki Hanyu (Department of Construction, Junior College, Nihon University, 7-24-1 Narashinodai, Funabashi, Chiba, Japan), and Kazuma Hoshi (Department of Construction, Junior College, Nihon University, 7-24-1 Narashinodai, Funabashi, Chiba, Japan)

To identify practical ways to assess privacy protection at counseling area in community pharmacies, we conducted site-visit investigations at 84 community pharmacies including on 10 telephone interviews, and internet questionnaire survey on 160 patients. In the typical Japanese community pharmacies, space is small, the counseling area is open, and there are many patients who are waiting until their name is called. In that situation, busy and noisy and loose concentration, patients must try to accurately grasp medical information, and pharmacist are also working in the same situation. At the site-visits, we asked about aural or visual privacy issues and performed a psychological experiment to determine whether the patient-pharmacist conversation could be heard from the waiting seats for patients. At the internet questionnaire, we asked about counseling environment at community pharmacies. Responses to the site-visit investigation and internet questionnaire survey revealed that privacy-related problems were classified into 4 factors: physical environment (speech privacy), information sharing, pharmacist's social role, and a complex mechanism of the medical system. These factors appear to be interrelated, making it difficult to improve patient-centered care.

11:20

4aAAa6. Speech intelligibility in hospitals. Timothy Hsu, Mike Moeller, Jr., Arun Mahapatra, and Erica Ryherd (Georgia Institute of Technology, 771 Ferst Drive, Atlanta, GA 30332-0445, tissue@gatech.edu)

In hospitals, background noise has been shown to be problematic, not only for the patients but also for the staff. With respect to staff members, perceived stress and psychosocial factors can be affected negatively by noise. One particular factor that noise can inhibit is effective speech communication. Speech communication is essential for functions such as evaluation, admittance and treatment of patients. This paper will discuss results from measurements made in several different hospital units where traditional speech intelligibility metrics were analyzed. Additionally, newer analysis techniques such as Noise Occurrence Rates were investigated for their potential usefulness in speech intelligibility applications. Preliminary results show that in general, the hospital units show "poor" to "marginal" Speech Intelligibility (SII) qualitative scores. The results of this study help to better explain how speech is understood in various locations within the hospitals and can aid in hospital designs that support speech communication.

11:40

4aAAa7. Investigation into the acoustical performance of single stud steel wall assemblies. John LoVerde (Veneklasen Associates, 1711 16th St, Santa Monica, CA 90404, jloverde@veneklasen.com), Wayland Dong (Veneklasen Associates, 1711 16th St, Santa Monica, CA 90404), and Aaron Betit (Acentech, 33 Moulton St., Cambridge, MA 02138)

The draft "Interim sound and vibration design guidelines for hospital and healthcare facilities" by the Joint Subcommittee on Speech Privacy of the ASA includes recommended STC ratings for partitions between exam rooms. In a typical hospital, these partitions are about 15 feet high and constructed with 16 gauge studs at 16 inches on center. However, virtually all laboratory testing is performed on 8 foot high walls with 25 gauge studs at 24 in. on center. These tests are used for design and evaluation even though there is little published data on the effects of stud gauge and spacing and wall height on transmission loss. A previous study investigated the acoustical effects of stud gauge and spacing, and documented substantial changes in transmission loss and STC rating [A. Betit, "Performance Dtails of Metal Stud Partition," J. Sound and Vibration, 44(3), 14–16 (2010)]. A second testing program was established to extend the investigation to the effect of changes in wall height. Transmission loss (STC) tests were performed on drywall partitions of various heights and construction. The results of the testing program are presented.

Contributed Papers

12:00

4aAAa8. Aco(s)ustainability (Acoustics and Sustainability)—is a space truly functional for its intended use? Daniel Butko (University of Oklahoma, College of Architecture, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu), and Jack Randorff (Randorff and Associates Inc, 11 W Canyon View Dr, Ransom Canyon, TX 79366)

Numerous scientific studies defining acoustical values of common building materials and assemblies have been performed throughout history and then correlated to occupant health, productivity, and speech intelligibility within the defined space. Since occupant well-being should be a driving factor during design and construction phases, the effects of sound and noise should be considered an inherent component of sustainable design. The functionality of the inhabitable spaces for the intended purpose suddenly increases the scope of sustainability. Without knowledge of previous experiments or publications, various University of Oklahoma College of Architecture students were tasked with answering the following question: Is a space truly functional for the intended purpose? Various built environments were evaluated, basic SPLs and frequencies were documented, and the results were compared with published data for construction materials and methods including OSHA and ANSI regulations. Students were anonymously surveyed concerning acoustical conditions they experienced to define what they considered helpful and distracting conditions. This paper focuses on introducing architecture students to the intrinsic link between acoustics and sustainability, allowing an appreciation for both the art and science contributing to inhabitable space(s).

12:20

4aAAa9. Public address system reinstallation. Wilson Ho and Eddy Ng (Wilson Acoustics Limited, Unit 616, Technology Park, 18 On Lai Street, Shatin, Hong Kong, China, who@wal.hk)

In the last decade, new functions for the Public Address (PA) system have been continuously developed and integrated. On the other hand, the existing PA systems are being less repairable and maintainable due to lack of supply of old version components although the system and loudspeakers are still functioning well within the specifications after many years of use. It appears to be sensible way to reinstall the control and amplification system without replacing existing loudspeakers in order to accommodate the new function and zoning requirements. This approach is being tested in Lantau and Airport Railway (LAR) line in Hong Kong operated since 1998. This paper presents the pros and cons using this reinstallation approach in practice and investigates the feasibility and effectiveness through acoustic simulation and on-site verification test of sound coverage and speech intelligibility.

Session 4aAAb

Architectural Acoustics, Physical Acoustics, Noise, and Signal Processing in Acoustics: Development and Applications of Micro-Perforated Sound Absorbers (Lecture/Poster Session)

Christian Nocke, Cochair
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Jonathan Botts, Cochair
bottsj@rpi.edu

Chair's Introduction—9:15

Invited Papers

9:20

4aAAb1. Brief review on micro-perforated sound absorbers. Christian Nocke, Catja Hilge (Akustikbüro Oldenburg, Katharinenstr. 10, D-26121 Oldenburg, nocke@akustikbuero-oldenburg.de), and Jean-Marc Scherrer

The theory of microperforated panel sound-absorbing constructions has been introduced by D.-Y. Maa in 1975. Since then many variations of micro-perforated sound absorbing devices and materials have been introduced. Materials that have been used to be micro-perforated have been metal, wood, plastics and many others. Stretched sheets used as ceilings, wall coverings and other set-ups have been applied for more than 40 years. In 2001 a nearly invisible micro-perforation has been introduced to the stretched material making it highly sound absorptive. The classical set-up of a micro-perforated sound absorber consists of a micro-perforated panel in front of an air cavity. The sound absorption coefficient of these set-ups can easily calculated with a high accuracy according to the well-known approximation of D.-Y. Maa if all defining geometrical parameters (diameter of microperforation, distance between orifices, panel thickness and air cavity depth) are known. For other assemblies no closed calculation model exists so far. In this contribution measured sound absorption coefficients of various set-ups with micro-perforated materials as well as combinations with different porous materials will be presented.

9:40

4aAAb2. Coupled mode analysis of thin micro-perforated panel absorbers. Cedric Maury (Ecole Centrale Marseille, Laboratoire de Mécanique et d'Acoustique (LMA), CNRS UPR 7051, 31 chemin Joseph-Aiguier, 13402 Marseille cedex 20, France, cedric.maury@centrale-marseille.fr), Teresa Bravo (Centro de Acustica Aplicada y Evaluacion No Destructiva (CAEND), CSIC-UPM, Serrano 144, 28006 Madrid Spain), and Cedric Pinhede (Laboratoire de Mécanique et d'Acoustique (LMA), CNRS UPR 7051, 31 chemin Joseph-Aiguier, 13402 Marseille cedex 20, France)

The prediction of the isolating properties of lightweight Micro Perforated Panels (MPP) is a subject that has been intensively studied due to their important applications in a wide range of areas such as building acoustics and the aeronautic, astronautic and automotive industries. MPPs have been mostly considered as rigid structures, accounting only for inertia and neglecting any vibrating effects. However, simulation and experimental studies on thin MPPs have found that the absorbing performance can experience variations in the low frequency range from the results expected assuming a rigid structure. The work presented here is a theoretical and experimental study on the influence of panel vibrations on the sound absorption properties of thin MPP absorbers. Measurements show that the absorption performance generates extra absorption peaks or dips that cannot be understood assuming a rigid MPP. A theoretical model is established that exactly accounts for structural-acoustic interaction between the micro-perforated panel and the backing cavity without restriction on the absorber cross-sectional shape or on the panel boundary conditions. This model is verified experimentally against impedance tube measurements and laser vibrometric scans of the cavity-backed panel response. The effect of micro-perforations on panel-cavity or hole-cavity resonances is revealed through coupled mode analysis.

10:00

4aAAb3. (Micro-)Perforated wooden panels as sound absorbers. Adrian Eichhorn (Akustik Plus GmbH & Co. KG, a.eichhorn@eichhorn-holzwerkstaette.com), Michael Beckmann (EGGER Holzwerkstoffe Brilon GmbH & Co. KG), and Christian Nocke (Akustikbüro Oldenburg)

Different materials have been used as micro-perforated or perforated panels for applications as sound absorbers. The theory of microperforated panel sound absorbers introduced by D.-Y. Maa in 1975 is independent on the material of the panel. So also micro-perforations in wooden panels will give sound absorption. In combination with porous absorbants the efficiency of the absorbers set-ups can be improved. Modern manufacturing tools for wood and wooden veneers allow for perforations of submillimeter diameters of the single pores. Optically these perforations hardly change the impression of the wooden panel. Acoustically the combination of perforated wooden panels and a backing cavity give highly effective sound absorbers. Measured sound absorption coefficients of various set-ups with (micro-)perforated wooden panels will be presented. The new possibilities in design and applications in architectural acoustics will be discussed.

10:20

4aAAb4. Application of microperf and microslit absorbers. Peter D'Antonio (RPG Diffusor Systems, Inc., 651C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

The theory of microperf and microslit absorbers is well established. Theoretical predictions match measurements with a high degree of accuracy. A normal incidence impedance tube measurement of a microperforated metal panel will illustrate this agreement. In this presentation, I would like to discuss how these transparent and translucent foils and panels have been used as both acoustical and decorative elements in projects. The microperf design is available in polycarbonate and ETFE foils as well as panels. The foils can also be printed for light shading. Since polycarbonate transmits in the infrared, the translucent foils can be used with back lighting on radiant ceilings. Several projects will be presented. The microslit design is typically available in panels, which can be transparent, translucent and digitally printed in graphic designs and signage. A graphically printed microslit atrium project will be profiled illustrating the architectural acoustic potential of this approach. These relatively new absorbers add an important tool to the acoustical palette and this presentation will illustrate how they have been used in a wide range of projects.

10:40–11:00 Break

Contributed Papers

11:00

4aAAb5. Acoustical perforated facings: a synthesis study of theoretical and experimental developments. Luc Jaouen, Fabien Chevillotte, and François-Xavier Bécot (Matelys, 1 rue Baumer, 69120 Vaulx-en-Velin, France, luc.jaouen@matelys.com)

Perforated facings (including Micro Perforated Panels) or perforated ceiling tiles have been widely studied since the fundamental work by Uno Ingard. This work specially focus on flow modifications in the vicinity of the perforations leading to modifications of the reactance of the panel. This communication is a synthesis study of pioneer and recent works on this topic for linear flow regimes, clarifying and correcting some results. One aspect of this communication revisits the length corrections for circular, rectangular and slits perforated panels. A second aspect aims at accounting for perforated panel vibrations based on Biot's theory. Coupling this last approach to the one by Atalla & Sgard [J. Sound Vib. 303, 195–208 (2007)], the current work allows to model perforated facings as elastic-frame porous media. Simulations results are validated using a recently proposed method for the characterization of perforated facings. Based on these results, general trends for dimensioning the acoustical performances of perforated panels are drawn.

11:20

4aAAb6. Hybrid sound absorbers combining micro-perforated panels with conventional absorption mechanisms. Marc Buret (Vipac Engineers and Scientists, 279 Normanby Road, Port Melbourne, VIC 3207, Australia, marcb@vipac.com.au), and King Kwong Iu (NAP Acoustics Far East Ltd., Room 1811, 18/F., Hong Kong Plaza, 188 Connaught Road West, Hong Kong)

Combination of microperforated panels with optimised efficiency in the low frequency range and other sound absorption systems, that provide performance in the mid and high frequencies, is presented for two sound absorber designs. In the first instance, proprietary fabric cover and fibrous absorption have been used to extend the performance range of discrete microperforated absorber units by optimising sound absorption by the edge of the units. The second development consists of conventional Helmholtz resonator perforated panels that have been customised using a micro-perforated panel in view to tune and enhance the low frequency performance. Results of testing conducted in a reverberation chamber are presented.

11:40

4aAAb7. Noise attenuation by sonic crystal barriers made of microperforated units. Victor M. García-Chocano, Suitberto Cabrera, and José Sánchez-Dehesa (Wave Phenomena Group, Universitat Politècnica de València, Camino de vera s.n. (Edificio 7F), E-46022 Valencia, Spain, vicgarch@upvnet.upv.es)

This work studies the absorptive properties of periodic arrays of microperforated cylindrical shells. Structures made of cylinders 3 meters height have been constructed and their reflectance and transmittance spectra are measured in open air at normal incidence. A broadband strong attenuation is found in the low frequency region. Experimental data are supported by model simulations performed in the framework of multiple scattering theory. It is concluded that these structures in combination with high frequency absorbing units are suitable to produce general purpose broadband noise barriers. Work supported by ONR (USA) and MICNN (Spain).

12:00

4aAAb8. Modeling vibro-acoustics behaviour of micro-perforated structures using patch transfer function approach. J.-L. Guyader, L. Maxit (Laboratoire Vibrations Acoustique, Institut National des Sciences Applique'es (INSA) de Lyon, 69621 Villeurbanne, France, jean-louis.guyader@insa-lyon.fr), and L. Cheng (Department of Mechanical Engineering, The Hong Kong Polytechnic University, Kowloon, Hong Kong Special Administrative Region)

Micro-perforated structures with a backing cavity is a device for providing efficient noise absorption. In a practical and industrial setting, the efficiency of Micro-Perforated Structure (MPS) may be influenced by the vibro-acoustic behavior of the surrounding systems, the shape of the micro-perforated structure as well as different kinds of excitation. In this paper, the Patch Transfer Functions (PTF) approach is proposed to model the MPS behaviour in such Complex Vibro-Acoustic Environment The PTF method is a substructuring approach which allows assembling different vibro-acoustic subsystems coupled through surfaces. The proposed PTF formulation of the MPS is capable of taking the micro-perforations and the flexibility of the structures into account and allows easy prediction of the efficiency of a MPS in a practical vibro-acoustic environment. In order to validate the present approach, PTF results are compared with experimental measurements.

The following abstracts will be presented in poster format. The posters will be on display and the authors will be at their posters from 12:20 p.m. to 1:00 p.m.

4aAAb9. A study on the sound insulation performance of obliqe micro-perforated absorbers in shading louver. Chu Wen-Sung, Wang Yu-Che, and Lai Rong-Ping (Department of Architecture, National Cheng Kung University, No. 1, University Road, Tainan 701, Taiwan, n78981222@mail.ncku.edu.tw)

Generally shading louver are used for shading and natural ventilation, but in noisy urban area, sound insulation is necessary, in the trend of green building design, shading natural ventilation and insulation are taken the same weighting, but it is contradictory, meanwhile, the performance of water proof also required in louvre. In study, we try to use oblique micro-perforated for shading louvre, and reach it sound insulation. Acoustics louvres normally use in silencers for Air conditioning duct, it is composited by Perforated plate any glass fiber, for the water proof season, we use oblique micro-perforated instead of glass fiber, and design and some influence factors those are specified of form, blade length, blade width, blade angle and air gap and blade is

composed of an infill of sound absorption material enclosed by perforated sheet material, sound insulation of an acoustic louvres generally is not high, particularly at low frequencies. We designed several type of the louver and test it acoustical performance, and discussed it is low, median. high frequencies. Finally, we calculate it's R_w by ISO 140, by the way, we also asses it ventilation, lighting, shading, affect, it is very important in green building.

4aAAb10. Non-flammable woven acoustic flow resistive textile. Marek Kierzkowski (Marek Kierzkowski Acoustic Consultancy, P.O. Box 1217, Mountain Gate, 3156 Victoria, Australia, psowy@bigpond.net.au)

Flow resistive textiles are becoming more popular almost in all aspect of acoustic applications where sound absorption is a primary noise reduction countermeasure: industrial acoustics (reduced sound propagation in work places), architectural acoustics (shaping interior acoustic properties) and the automotive acoustics (bonnet liners, firewall insulators). While the automotive applications are not very demanding in terms of flammability, the industrial and architectural applications must comply with severe flammability restrictions. As it is today only non-flammable porous materials like mineral fibre, ceramic fibre or melamine foam would comply with stringent fire specifications. We will show that the wise choice of the flow resistive textile enable to widen the range of materials complying with actual standards. The traditionally good sound absorbers like polyester fibre or foam could then become available to architects again.

THURSDAY MORNING, 17 MAY 2012 S424, 9:20 A.M. TO 12:40 P.M.

Session 4aAB

Animal Bioacoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Biosonar II

Wang Ding, Cochair
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Cynthia Moss, Cochair
cynthia.moss@gmail.com

Invited Papers

9:20

4aAB1. Findings on bat sonar through Telemike system. Hiroshi Riquimaroux and Shizuko Hiryu (Doshisha University, 1-3 Miyakotani, Tatara, Kyotanabe, Kyoto 610-0321, Japan, hrikimar@mail.doshisha.ac.jp)

In order to understand how bats conduct echolocation recording what they listen to is essential. Then, we developed an onboard wireless telemetry microphone system (Telemike) for flying bats in our flight chamber. We also developed a sensitive microphone array system for field recordings and for the flight chamber. Some data through microphone array with Telemike will also be introduced. Both CF-FM (Hipposideros turpis, Hipposideros terasensis and Rhinolophus ferrumequinum nippon) and FM bats (Pipistrellus abramus, Miniopterus fuliginosus, and Eptesicus fuscus) were used in the flight chamber experiments, while only FM bats (Pipistrellus abramus and Eptesicus fuscus) were used for the field experiments. In the flight chamber experiment, through Telemike Doppler-shift compensation was confirmed from flying CF-FM bats. Echo amplitude compensation was found in both FM and CF-FM bats. The bats can fly without any collision against walls or their conspecifics in a narrow space. Telemike has recorded overlapped echoes originated from adjacent bats. How they extract weak echo signals in the situation where own echoes are masked by echoes of other bats may be revealed by Telemike experiments. Further, how bats get information from extremely weak echoes coming from a flying small insect will also be clarified in our experiments with Telemike. [Research supported by ONR grant]

9:40

4aAB2. A broad band dolphin mimetic sonar—inspiration and modification from the nature. Tomonari Akamatsu, Tomohito Imaizumi, Koki Abe (National Research Institute of Fisheries Engineering, Fisheries Research Agency, akamatsu@affrc.go.jp), Yasushi Nishimori, Young Wang (Furuno Electric Co., Ltd.), Ikuo Matsuo, and Masanori Ito (Department of Information Science, Tohoku Gakuin University)

Broadband techniques are getting popular for underwater sensing methods because of its high spatial resolution and target discrimination abilities. We have been developing a broadband split beam system to locate and identify each species in the ocean. Our system initially learned from dolphins and then modified architecture appropriately. For example, biological sonar sound was effective for the short range sensing to locate individual target, chirp sound provided clear target image for the long range up to 200 m. Multi angle scanning of a target was proved to be essential for the species discrimination in our system. It was just like a finless porpoises rolled their body possibly to enlarge sensing volume and change beam incident angles to a target. Unlike dolphins, the split beam system was not able to change transmitting beam directions. The sound incident angle to a fish was calculated using the body movement vector and the position of a target fish in the beam. Reconstruction of target strength spectrum according to the incident angle provided significant difference of species between jack mackerel and chub mackerel that has not been possible by conventional active sonar systems.

10:00

4aAB3. Immediate changes in whale hearing sensitivity. Paul E. Nachtigall (University of Hawaii, Marine Mammal Research Program, P.O. Box 1106, Kailua, Hawaii 96734, nachtiga@hawaii.edu), and Alexander Ya Supin (Russian Academy of Sciences, 33 Leninsky Prospekt, Moscow, Russia)

We have been examining the hearing of both the outgoing clicks and the returning echoes of actively echolocating odontocetes using evoked auditory potential techniques. In order to protect themselves from the loud outgoing sound while still maximizing the hearing of the acoustic echo return, odontocete echolocators appear to have developed both passive and active control of hearing. Passive control has been demonstrated by comparing hearing of their own outgoing signals to similar signals presented to them from the outside. Clicks produced by the animal itself are heard about 40 dB down. Active control has been demonstrated by a comparison of hearing outgoing clicks during target present and target absent trials. During target absent trials, when searching for targets, hearing is 20 dB more sensitive than during target present trials. The current critical question is: If the animal is warned that a loud sound is about to arrive, does it possess a mechanism of self-mitigation that will allow it to control its own hearing and reduce the level of the incoming sound? Initial results indicate that a false killer whale will reduce hearing sensitivity by at least 15 dB when warned that a 170 dB signal is about to arrive.

10:20

4aAB4. Localization of the moving object by echolocation. Ikuo Matsuo (Department of Information Science, Tohoku Gakuin University, 2-1-1 Tenjinzawa, Sendai 981-3193, Japan, and Neurosensing and Bio-Navigation Center, Doshisha University, 1-3 Miyakodani, Tatara, Kyotanabe, Kyoto 610-0321, Japan, matsuo@cs.tohoku-gakuin.ac.jp)

Using the echolocation, bats can capture moving objects in real 3D space. Bats emit the frequency modulation sound and can accurately localize these objects from echoes. The object's range could be estimated from delay times between the emitted sound and echoes from objects. These positions in 2D space could be estimated from the difference between delay times at two ears, and the accuracy of localization was dependent on the range accuracy, which was dependent on the frequency width of the emitted sound, the signal-noise-ratio (SNR), and the Doppler shift. It has been shown that the previous proposed model could accurately estimate each range of static objects by using the frequency modulation sound at the low SNR. However, it is unknown whether this model could estimate the moving object in 2D space. In this study, the echoes were measured from the rotating pole by emitting intermittently the LFM sounds. These echoes were analyzed by using the Gaussian Chirplet filters with a carrier frequency compatible with emission sweep rates. It was clarified that this proposed model could track the moving object by estimating object's position in 2D space at each time.

10:40–11:00 Break

11:00

4aAB5. Cochlear structural variants in echolocators. Darlene R. Ketten (Woods Hole Oceanographic Institution, Biology Dept., Woods Hole, MA 02543 and Harvard Medical School, Boston, MA 02114, dketten@whoi.edu), James Simmons (Brown University, Neurosciences, Providence, RI), Hiroshi Riquimaroux (Doshisha University, Graduate School of Life and Medical Sciences, Kyoto, Japan), Scott Cramer, and Julie Arruda (Woods Hole Oceanographic Institution, Woods Hole, MA)

Although microchiropteran bats and odontocete cetaceans operate in radically different media, both have sophisticated sonar capabilities and evident similarities in their ability to detect and analyze ultrasonic signals. This paper compares the similarities and differences of cochlear cytoarchitecture and its implications for ultrasonic encoding and acuity amongst these groups through the use of three-dimensional models obtained via micro-CT imaging of intact heads and temporal bones. Inner ear anatomy was fundamentally similar with notable parallels in fenestral placement and ratios, membrane dimensions, and neural density and distribution across bats and dolphins with common cochlear types. Specialist ears are present in both groups, suggesting that like some CF-CM bats, one or more odontocete species have cochleae with specialized basilar membrane "foveal" regions. Cochlear specializations in both groups are primarily linked to peak spectra of signal, expanded frequency representation, and may enhance tuning in adjacent ear segments by generating standing wave phenomena. [Supported by N45- US Navy Environmental Division and the Office of Naval Research]

Contributed Papers

11:20

4aAB6. An approach for moving target detection with airborne CTFM sonar. Yang Wang, Yong Xu, Benxi Cao, Jingyao Wang, and Jun Yang (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, wy8008@126.com)

Continuously Transmitted Frequency Modulated (CTFM) sonar transmits wideband acoustic signal, demodulates the echo signal, and calculates the target distance by acquiring the frequency of the demodulation output signal. Still objects can be accurately detected using CTFM sonar. However, moving target brings Doppler shift to the echo signal frequency, which makes the output signal frequency of CTFM sonar deviate from the accurate distance of the target. In this paper, a novel airborne sonar sensing approach is proposed. Using a modified transmitted signal, single moving target that appears in a static environment can be detected. Compared to traditional CTFM sonar, a single tone is added to the transmitted signal. The existence and radial velocity of a moving target can be calculated based on the single tone Doppler shift of the echo. Furthermore, the simultaneous kinematic information of the target can be extracted by the system algorithm. An experimental system is developed, and the result of experiments verified the feasibility of the approach.

11:40

4aAB7. Research on blind source separation of marine mammals signal processing under watercraft emitted noise. Zhang Liang and Guo Long-Xiang (Harbin Engineering University, 150001, qq-zhangliang@163.com)

As statistical independence exists between marine mammals sound and watercraft emitted noise, and among different organism signals, the paper introduces blind source separation (BSS) into marine mammal signals processing. BSS with single hydrophone is a special underdetermined blind separation problem, and BSS based on matrix calculating is no longer suitable.

The problem can be solved by expanding channels, and further study finds that second iteration of blind separation can improve the performance. Algorithm simulation and experimental data analysis show that not only marine mammals signal but also different organism signals can be separated by this method with single hydrophone. It is proved that the correlation coefficient of the separated signal is obviously improved, which lays the foundation for the feature extraction and recognition of marine mammals signal. Keywords-marine mammals signal processing; BSS; maximum signal noise ratio criterion; second iteration

12:00

4aAB8. Recovery cycles of inferior collicular neurons in the leaf-nosed bat, Hipposideros armiger. Jia Tang, Zi-Ying Fu (School of Life Sciences, Central China Normal University, Wuhan 430079, China, bobaytang@yahoo.com.cn), Philip Hung-Sun Jen (Division of Biological Sciences, University of Missouri-Columbia, MO 65211), and Qi-Cai Chen (School of Life Sciences, Central China Normal University, Wuhan 430079, China)

When stimulated with biologically relevant constant frequency–frequency modulation (CF–FM) sounds, the inferior collicular neurons of the CF–FM bat, Hipposideros armiger, either only discharged impulses to the onset (76%, single-on neurons) of the CF-FM sounds or to the onset of both CF and FM components of CF–FM sounds (24%, double-on neuron). Some neurons were single-on responders at low sound amplitude but become double-on responders at high sound amplitude. Single-on responders had longer latency and recovery cycle than double-on responders. While most neurons did not respond to the second sound when the paired CF-FM sounds overlapped, 3 single-on and 7 double-on neurons did such that they had “cyclic” recovery cycles with inter-pulse intervals. The different response latency and dynamic variation in the recovery cycle of these two types of neurons suggest they may serve as the neural basis underlying a bat’s ability to perform echo ranging throughout different phases of hunting.

12:20

4aAB9. Echolocation beam shape and focusing in the false killer whale (*Pseudorca crassidens*). Laura Kloepper, Paul Nachtigall, and Marlee Breese (Hawaii Institute of Marine Biology, P.O. Box 1106, Kailua, HI 96734, kloepper@hawaii.edu)

Odontocete echolocation signals are thought to be focused by the melon and air sacs, although active focusing has yet to be demonstrated empirically. Because odontocete echolocation signals are variable and the emitted click frequency greatly affects the echolocation beam shape, investigations of beam focusing must account for frequency-related beam changes. Using a fine scale hydrophone array, we measured the shape of the echolocation beam and tested whether the echolocation beam of a false killer whale changed depending on target difficulty and distance while also accounting for frequency-related changes in the echolocation beam. The false killer whale produced a single-lobed echolocation beam that changed in size depending on target distance and difficulty which may be a strategy of actively controlling the emitted beam to maximize energy of the target echo.

THURSDAY MORNING, 17 MAY 2012 S224 + S225, 9:20 A.M. TO 12:40 P.M.

Session 4aBA

Biomedical Acoustics: Bone Quantitative Ultrasound I

Pascal Laugier, Cochair
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Dean Ta, Cochair
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Invited Papers

9:20

4aBA1. Correlations between propagation characteristics of guided ultrasonic waves and long bone fatigue. Dean Ta, Zhenggang Zhang, and Weiqi Wang (Fudan University, 220 Handan Rd, Shanghai, China, tda@fudan.edu.cn)

Ultrasonic assessment of long bone has become a topic of interest in recent years. The objective of this paper is to analyze the propagation characteristics of guided ultrasonic waves in fatigue long bones, further to study the influence of varying elastic modulus (EM) of long bones on these characteristics. A hollow cylinder was used to mimic long bones and then to calculate the velocities and wave structures with various EMs. Besides, finite difference time-domain (FDTD) method was used to simulate the propagation of guided waves in long bones at different EMs. The results show that phase/group velocities and central frequencies of guided waves decrease with the decrease in EMs. However, the attenuation of wave modes decrease with increasing EMs. The simulated results of FDTD for all wave modes and parameters are in good agreement with theoretical values. Those results demonstrated that different modes have diverse sensitivity to the variation of EMs, and mode velocities, central frequencies and attenuations can reflect the change of EMs of long bone. Therefore, the propagation characteristics of guided ultrasonic waves may provide a feasible approach to evaluate the early stages of fatigue damage in long bones.

9:40

4aBA2. Progress and challenges in axial transmission measurements of guided modes in cortical bone. Jean-Gabriel Minonzio, Josquin Foiret, Ludovic Moreau, Maryline Talmant, and Pascal Laugier (CNRS - UPMC LIP, 15 rue ecole de medecine, 75006 Paris, France, jean-gabriel.minonzio@upmc.fr)

Cortical bone porosity has been evidenced as being a major if not the major ‘footprint’ of bone loss and fragility. Several studies report that cortical bone behaves like a waveguide. Measurements of guided mode wavenumbers together with appropriate waveguide modeling have therefore the potential for providing estimations of effective stiffness coefficients (which are largely determined by cortical porosity) and also cortical thickness. However, data interpretation is challenging due to the heterogeneous, dissipative and irregular nature of the wave guide. Moreover surrounding and internal tissues modify the guided modes. This paper presents current progress by our group in the measurement of the wavenumbers of guided wave modes, using a multi-transmitter multi-receiver axial transmission probe. The guided mode wavenumbers are obtained after projection of test vectors onto the basis of the singular vectors of the transfer response matrix, at each frequency. The method has been validated, first on isotropic elastic and visco-elastic plates, then on bone-mimicking plate and tube phantoms made of transverse isotropic absorbing material. The effect of soft-tissue mimicking layers on top of a bone mimicking phantom has also been studied. Finally, preliminary in vivo testing of the approach on human radius will be presented.

Contributed Papers

10:00

4aBA3. A simulation and experimental study of long cortical bones fracture evaluation using lamb waves. Kailiang Xu, Runxin He, Dean Ta (Department of Electronic Engineering, Fudan University, Shanghai 200433, China, xukl@fudan.edu.cn), Yixian Qin (Department of Biomedical Engineering, Stony Brook University, Stony Brook, NY), Peng Sun, and Weiqi Wang (Department of Electronic Engineering, Fudan University, Shanghai 200433, China)

Ultrasonic guided waves are promising to evaluate fractured long cortical bones. Guided modes conversion, occurred always on the sites of fractures, contains rich information of bone geometric and material properties. The study is to analyze the correlation between Lamb modes conversion and crack depth of fractured long bones. Axial experiments were designed to investigate the influences of different deep fractures in steel-made bone phantoms and sheep diaphyseal tibias over Lamb modes propagation, especially modes conversion. For comparison, the phantoms were modeled by two dimension finite-difference time-domain method. Group velocities and energies of S0, A0 and converted modes were extracted and analyzed under varying crack depth conditions. Being consistent with the simulation, experimental data showed obvious modes conversions occur between S0 and A0 modes. The fracture positions can be predicted from converted modes velocities. It illustrated the modes conversions become gradually observable with the fracture depth increasing, which can be indicated by the parameter, converted energy proportion (EP). The sensitivity and practicality of EP to assess bone fractures were validated in quantitative analysis of phantoms, simulations and sheep tibias experiments in vitro. In conclusion, ultrasonic guided wave is a reasonable method for long cortical bone fracture detection and evaluation.

10:20

4aBA4. Determination of bone thickness: effect of group velocity filtering in multi-component waveguides. Petro Moilanen (Department of Physics, P.O. Box 35, 40014 University of Jyvaskyla, Finland, petro.moilanen@jyu.fi), Maryline Talmant (Laboratoire d'Imagerie Parametrique, UPMC Univ Paris 06, CNRS, UMR 7623, 75005 Paris, France), and Jussi Timonen (Department of Physics, P.O. Box 35, 40014 University of Jyvaskyla, Finland)

Previously, the Jyväskylä group has used the F11 guided wave of an empty tube as the reference dispersion curve for identification of in vitro bone thickness. The role of the marrow has remained an open issue. The objective of this study was thus to test several reference waveguides, empty, fluid-filled and fluid-coated tubes, in modelling of guided modes. For instance, fluid filling in a tube strongly modifies the spectrum of guided waves found for an empty tube, because of coupling of guided waves in the tube wall and in the fluid-filled cavity. However, it is shown that, using group velocity filtering on the signal component of highest amplitude, the effect of fluid filling can essentially be eliminated. The dispersion curve of an apparent F11 mode is obtained with a slight shift only at the lowest frequencies. This small shift means that the dispersion curve of this apparent mode is, in that frequency range, between those of A0 (2D plate) and F11 (3D cylinder), and can to some extent be interpreted as either one of these.

10:40–11:00 Break

11:00

4aBA5. Time-frequency spectrum segmentation method for separating multimodal guided waves in long bones. Zhenggang Zhang, Dean Ta, and Weiqi Wang (Department of Electronic Engineering, Fudan University, Shanghai 200433, China, zhangzg0311@gmail.com)

Ultrasonic guided waves have great potential for evaluating long bones. However, the measured signal often contains multiple wave modes because of the complicated characteristics of guided waves, which cause difficulties to further analysis. In this study, a hollow cylinder filled with a viscous liquid was used to model the long bone, and the multimodal signals were simulated using finite difference time-domain (FDTD) method. The time-frequency spectrum segmentation method was proposed to separate multiple modes. First, the Gabor time-frequency spectrum of multimodal signals was calculated. Second, a multilevel image segmentation algorithm, including the watershed and region growing, was used to find the corresponding area of each mode in the spectrum. Finally, time domain signals representing individual modes were reconstructed from these areas. The validations of this method were analyzed by simulated multimodal signals under different elastic modulus (EM) of long bones, with or without noise. The results showed that the Gabor time-frequency representations of the individual modes were in good agreement with the theoretical dispersion curves. This study suggests that time-frequency spectrum segmentation method can correctly separate multimodal guided waves, which provides a foundation for feature extraction of individual guided modes.

11:20

4aBA6. Circumferential cortical wave propagation at the proximal femur predicts bone strength. Julien Grondin, Quentin Grimal (University Pierre et Marie Curie, Paris, F-75006, France, grdjulien@gmail.com), Sandra Guérard (Arts et Metiers ParisTech, F-75013 Paris, France), Reinhard Barkmann, Claus Glüer (Universitätsklinikum Schleswig-Holstein, Kiel, Germany), and Pascal Laugier (University Pierre et Marie Curie, Paris, F-75006, France)

The importance of predicting hip fracture risk and the key role of cortical bone to maintain femur neck mechanical integrity both motivate one important aspect of the research presented here which is to focus ultrasound measurements on cortical bone at the femoral neck. Hypothesizing that the circumferential propagation at the femur neck may be predictive of femur strength, this in vitro experiment investigates the relationship between ultrasound circumferential propagation and femur strength. For nine femurs of women we measured: (1) the time-of-flight (TOF) of the first arriving circumferential wave guided by the cortical shell at the femoral neck; (2) structural features and density (BMD) using quantitative X-ray computed tomography; (3) femur strength in one-legged stance configuration with state-of the-art mechanical tests. Significant relationships were observed between TOF and mechanical parameters: failure load: $R2=0.79$; elastic energy: $R2=0.63$; apparent stiffness: $R2=0.70$; TOF was also well correlated with BMD in the inferoanterior quadrant of the neck, consistently with a circumferential propagation path along the thicker inferior cortex. Our results evidencing that circumferential propagation TOF is related to strength and reflects local properties of the femoral neck cortex offer perspectives for enhanced in vivo assessment of bone strength directly at the hip.

11:40

4aBA7. Photo-acoustic excitation and detection of fundamental antisymmetric Lamb mode in coated bone phantoms. Petro Moilanen (Department of Physics, P.O. Box 35, 40014 University of Jyvaskyla, Finland, petro.moilanen@jyu.fi), Pasi Karppinen, Timo Karppinen (Department of Physics, P.O. Box 64, 00014 University of Helsinki, Finland), Zuomin Zhao, Risto Myllyla (Department of Electrical and Information Engineering, P.O. Box 4500, 90014 University of Oulu, Finland), Edward Haeggstrom (Department of Physics, P. O. Box 64, 00014 University of Helsinki, Finland), and Jussi Timonen (Department of Physics, P.O. Box 35, 40014 University of Jyvaskyla, Finland)

Photo-acoustic (PA) imaging was combined with skeletal quantitative ultrasound (QUS) for multi-mode ultrasonic assessment of human long bones. This approach permits tailoring the ultrasonic excitation and detection to efficiently receive the fundamental antisymmetric Lamb mode (A0) through a coating of soft tissue. The method was tested on five axisymmetric bone phantoms of individualized wall thickness (1-5 mm) made of a composite material and coated with a layer (2.5 mm) of soft material that mimics the soft tissue. Signals were excited with a pulsed Nd:Yag laser at 532 nm wavelength and detected on the same side of the coated phantom with (i) a laser Doppler vibrometer (LDV) and for comparison also with (ii) a piezoelectric contact ultrasound receiver, scanning a source-receiver distance of 20-50 mm along the phantom. At a centre frequency of 50 kHz, a phase velocity consistent with that of the theoretically predicted A0 mode was identified in the recorded signals. Our results thus suggest that photo-acoustic quantitative ultrasound enables assessment of the thickness-sensitive A0 mode in bone through a layer of soft tissue. Ultrasonic in vivo characterization of the cortical bone thickness may thus become possible.

Invited Paper

12:00

4aBA8. Multiscale elastic imaging & modeling of musculoskeletal tissues. Kay Raum, Susanne Schrof (Charité-Universitätsmedizin Berlin, Julius Wolff Institute, Augustenburger Platz 1, 13353 Berlin, Germany, kay.raum@charite.de), Sara Tiburtius (Technische Universität Darmstadt, Fachbereich Mathematik, Dolivostr. 15, 64293 Darmstadt, Germany), Quentin Grimal (Laboratoire d'Imagerie Paramétrique, UMR CNRS 7623 - Université Paris 6, 15 rue de l'école de médecine, 75006 Paris, France), and Alf Gerisch (Technische Universität Darmstadt, Fachbereich Mathematik, Dolivostr. 15, 64293 Darmstadt, Germany)

Sophisticated technical materials that are used in everyday life are often inspired by nature. Hard biological tissues, e.g. mineralized tendons, bone and teeth are natural examples of achieving unique combinations and also great variability of stiffness and strength. In order to achieve these goals, bone uses various design concepts, e.g. reinforcing a soft and flexible collagen matrix by stiff, but brittle mineral particles, sandwich compounding of anisotropic (directional) films, weight reduction by directional pores and spongy networks. Although many details of the genetics, biology, pathology and mechanics of bone have been uncovered, we still lack of a detailed understanding of bone structure at the nano- and microscales. Towards this goal, both experimental data of heterogeneous elastic and structural parameters from all length scales (from the centimeter to the nanometer scale) and theoretical models that can simulate the deformation behavior based on these data are required. In this presentation the concept of multi-modal coupled multi-scale assessment of tissue properties (using quantitative ultrasound, synchrotron radiation μCT and vibrational microscopy) and modeling (using various homogenization techniques) will be presented with an emphasis of applications in musculoskeletal research, e.g. bone and cartilage healing.

Contributed Paper

12:20

4aBA9. Dependence of local wave velocity in bovine cortical bone on the decalcification. Kenji Fukui, Ryo Tsubota, and Mami Matsukawa (Doshisha Univ., Kyoto, Japan, kicomry38@gmail.com)

Bone is a composite material, mainly composed of HAp crystallites and type I collagen. It is known that the amount and orientation of HAp crystallites contribute to the "bone quality", which affects the bone elasticity. In this study, using a micro-Brillouin scattering technique which is able to evaluate wave velocity in the minute area, the effect of HAp amount on the velocity was measured. 36-plate-specimens in the plane of bone axis and radial directions were obtained from the middle part of a bovine femur. Wave velocity and HAp amounts were evaluated by the micro-Brillouin and XRD techniques, respectively. The specimens were then decalcified using ethylenediaminetetraacetic acid and measured again. Before decalcification, the average velocity was 5.06×10^3 m/s, and showed a moderate correlation with the HAp amounts (R^2}=0.56). After decalcification, the average velocity dramatically decreased to the value of 3.28×10^3 m/s, showing a strong dependence on the HAp amounts. In addition, the wave velocities except for the lateral part shows the moderate correlation (R^2}=0.30) before and after decalcification, which implies the possible effects of collagen on the wave velocities.

Session 4aEA

Engineering Acoustics: Flow Noise and Mitigation Methods

Randolph Leung, Chair
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Invited Paper

9:20

4aEA1. The comparison between passive and active methods of online cavitation detection. Jin Liu (College of Science, China University of Petroleum-Beijing, No. 18, Fuxue Road, Changping, Beijing, China; Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, liuj1314@gmail.com), Zhaoli Yan, Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China), and Wenxiao Qiao (College of Science, China University of Petroleum-Beijing, No. 18, Fuxue Road, Changping, Beijing, China)

Cavitation is the Achilles' heel of kinetic pumps and propellers. It can lead to performance degradation, structure vibration and noise, and bring about material erosion. Therefore some methods should be taken to detect cavitation. In this work, passive and active acoustics methods of online cavitation detection are set up to recognize cavitation and non-cavitation state. The former uses a hydrophone to receive emitted hydroacoustics signal. The signals from 10 kHz to 60 kHz are analyzed to extract features for pattern classification. The latter applies ultrasound to acquire flow field message. The ultrasound received is demodulated and the modulating signal is also analyzed for pattern classification. Experiments based on the two methods are carried out. Classification accuracy, computational complexity and installation difficulty are compared. Their applicability is also summarized.

Contributed Papers

9:40

4aEA2. Simultaneous measurement of density and viscosity of fluid using vibration of structure constrained by fluid. Deokman Kim (Hanyang University, deokman@hanyang.ac.kr)

Measurement of rheological properties of fluid using vibration of structure constrained by fluid. The fluid density and viscosity is the quantity to be measured and monitored during various manufacturing process. In this study, a real-time experimental method to simultaneously measure the density and viscosity of the fluid is proposed. The effects of fluids on flexural vibration of the beam structure partially immersed in fluid are analyzed. The density and viscosity have effects on the fluid-structure interaction. To analyze the fluid-structure interaction effects, the fluids are modeled as a simple support at one end of the beam. Using the proposed method, the density and viscosity of viscosity standard fluids were measured and its result was verified. The proposed method is advantageous in that the setup is possible to be installed in any fluid undergoing manufacturing process for real-time monitoring.

10:00

4aEA3. Flow noise from the transition region of an axisymmetric body in water. Xuegang Li, Kunde Yang, and Yuanliang Ma (Institute of Acoustic Engineering, Northwestern Polytechnical University, xuegang608@126.com)

Flow noise from transition region of an axisymmetric body is important for predicting the self-noise of a sonar mounted on an underwater platform. Numerical calculation of the flow noise for an axisymmetric body is presented and the diffracted loss on the head surface of the body is calculated by the geometrical theory of diffraction. The main physical features of flow noise are obtained. The flow noise in horizontal symmetry profile of the axisymmetric body is non-uniform, but it is omni-directional and has little difference in the cross section of the body. Based on the simulation, the noise power level increases with velocity to approximately the fifth power at high frequencies, which is consistent with the experiment data reported in the literature. Meanwhile, the flow noise received by the acoustic array on the curved surface has a stronger correlation than that on the head plane at the designed center frequency, which is import for sonar system design. Furthermore, the flow noises of two models with different shapes are compared and a rather optimum fore-body geometric shape is given.

10:20

4aEA4. Modelling and computation of boundary layer flow around body of revolution. Xie Hua, Shen Hong-cui, and Tian Yu-kui (P.O. Box 116, WuXi City, Jiangsu Province, China, xie621@163.com)

The characteristic parameters of boundary layer are inputs for the flow noise calculation, their change influence the power spectra of wall pressure fluctuations, so as to affect the analysis of the flow radiation noise. In this paper Hess-Smith boundary element method was adopted to model thick boundary layer for body of revolution. The corresponding code is developed. The computation of characteristic parameters of boundary layer for a body of revolution is carried out. The computed results including boundary layer thickness, shape factor, momentum thickness and friction coefficient are analyzed. The variation of characteristic parameter is obtained. The result showed that the code developed in this paper can be applied to the analysis of the body of revolution's 3D boundary layer calculation, which can offer input parameter for flow noise calculation. Key words: flow noise, characteristic parameter, boundary layer, body of revolution, Hess-Smith boundary element

10:40–11:00 Break

11:00

4aEA5. On the characterization of acoustic two-port sources using multi-load method. Hao Zhang (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, zhanghao@mail.ioa.ac.cn), Tao Feng (Department of Mechanical Engineering, Beijing Technology and Business University, Beijing 100048, China), Chengguang Zhou, Bilong Liu, and Ke Liu (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China)

A multi-load method for determining the source data of acoustic two-port sources is presented. By eliminating the term of source strength, the scattering matrix can then be obtained by solving a set of nonlinear algebraic equations, and the source strength is determined by the scattering matrix and the directly measured spectrum matrix corresponding to one of the acoustic loads. Numerical simulations indicate that the proposed method is effective. The method has been tested on a loudspeaker and an axial flow fan in a duct. The source data obtained by this method show reasonable agreement with that measured by the direct method. The multi-load method avoids using external sources, and it can be used as an alternative method when the external source is not easy to realize in practice.

11:20

4aEA6. Flow-induced acoustic resonance prediction using the transfer matrix method. Fei Liu, Sam Yang, and Lou Cattafesta (University of Florida, Gainesville, FL 32611, U.S.A., lfeicq@ufl.edu)

Acoustic resonance in a flow piping system may trigger an occupational or safety issue and can lead to equipment damage. Computational fluid dynamics (CFD) simulations are often used to predict such resonance phenomenon. However, this approach is generally time consuming and requires specialized training and expensive software. In addition, CFD is not viable for routine design purposes due to its computational expense. In this study, an alternative plane-wave based model for a hydraulic piping system is therefore developed using the transfer matrix method (TMM). Such a model can offer a fast yet reasonably accurate prediction for self-sustained pressure oscillation in a piping system. The advantage of the TMM is the simplicity with which the transfer matrix of a system can be generated from a combination of the TMs of its subsystems via matrix operations. The hydraulic piping system under study consists of a duct with constant cross-sectional area, diffuser, nozzle, bends, valves and orifice plate. The TM of each component is developed and compared to either CFD predictions or available experimental data, the TM of the complete system is derived. Design recommendations are made to reduce and/or avoid resonance in the piping system.

11:40

4aEA7. A numerical study of the effects of a winglet on airfoils. Chao-Nan Wang, Chuan-Cheung Tse (National Taiwan University, No. 1, Sec. 4, Roosevelt Rd., Taipei, 106 Taiwan, wangcn@ntu.edu.tw), and Ya-Ju Chang

The purpose of this research is to investigate the effects of a winglet on aerodynamic noise of an airfoil. For simplicity, Reynolds Averaged Navier-Stokes equations combined with Realizable turbulence model are used to solve the turbulent flow. In order to verify the accuracy of flow field analysis, a uniform flow past a three-dimensional rectangular airfoil is analyzed and tracks the center line of the tip vortex. The agreement between the simulated and measured center line is good. For the sound field analysis, the flow induced noise around a rectangular airfoil is computed by the Broadband Noise Source (BNS) model. Proudman's formula was used to evaluate acoustic power per unit volume of aerodynamic noise. This study focuses on the sound power of aerodynamic noise generated by tip vortex when the flow passes through an airfoil and an airfoil with winglet. In order to understand the effects of the winglet on the aerodynamic noise, the different winglet characteristics are investigated and discussed. It is found that with a winglet the dynamic coefficient is improved and the generated sound power is also reduced by about 5.4 dB in this study.

12:00

4aEA8. Low-dimensional modelling of sound generation by a flow past a bluff body. K. H. Seid, Randolph C. K. Leung (Department of Mechanical Engineering, The Hong Kong Polytechnic University, kahimseid@googlemail.com), and Garret C. Y. Lam (Department of Building Services Engineering, The Hong Kong Polytechnic University)

Searching for a unified methodology for controlling aeroacoustics of common structural discontinuities (e.g. bluff body, open cavity, etc.) has been a major topic among aeroacoustics research community. However, constrained by the complexity and nonlinearity of the flow governing equations, the process of reduced order modeling is usually required in which a low-dimensional, reduced-order model is created for approximating the full high-dimensional dynamics of the flow unsteadiness for control implementation purpose. The present study aims to extend two model reduction approaches, namely Proper Orthogonal Decomposition (POD) and Dynamic Mode Decomposition (DMD) via Galerkin projection method, to develop the reduced-order models for full compressible flows. The versatility of these methods are evaluated and compared by applying them to the aeroacoustics of flow past a square cylinder. It is expected that the outcome of this study could facilitate the development of a unified and versatile closed-loop control methodology for effective aeroacoustics control.

Session 4aHT

Hot Topics: Aeroacoustics I

Xiaodong Li, Cochair
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Fang Q. Hu, Cochair
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Chair's Introduction—9:15

Invited Papers

9:20

4aHT1. Noise source identification in high speed jets based on virtual microphone arrays. Philip Morris (Penn State University, 233 Hammond Building, University Park, PA, 16802, pjm@psu.edu), and Yongle Du (Penn State University, 229 Hammond Building, University Park, PA 16802)

Phased arrays have become a popular experimental technique for noise source identification. These techniques are generally limited in resolution due to the number of available microphones. In addition, noise source models are relatively simple, involving either uncorrelated point sources or simple coherent line sources. The former limitation is not present in numerical simulations, where the number and location of virtual microphones is effectively unlimited. The present paper describes numerical simulations of high speed jet noise using a Detached Eddy Simulation turbulence model. The nozzle, which is included in the simulations, is typical of the geometries found in high performance military aircraft. An Immersed Boundary Method is used simulate the effect of chevrons at the nozzle exit. The far field noise is predicted using solutions of the Ffowcs Williams – Hawkings equation. Phased array results are presented for both a baseline and chevron nozzle and the differences are discussed. In addition, near field virtual arrays are sampled and analyzed to include both radiating and non-radiating components. The implications of the results from the far and near field arrays in terms of noise source characteristics are presented. The results are compared with available experimental observations.

9:40

4aHT2. Investigation of noise radiation from a jet engine inlet by direct numerical simulation. Sarah Parrish and Christopher Tam (Florida State University, Department of Mathematics, Tallahassee, FL 32306-4510, parrish@math.fsu.edu)

In a jet engine, strong tones are produced by the fan and are radiated out of the inlet. Such fan noise is an important contributor to the total aircraft noise during take-offs and landings. Experimentally, it has been found that the sound radiation patterns from in-flight tests are quite different from those measured in static conditions. What accounts for this difference? In the current work, the radiation problem is studied computationally using direct numerical simulation based on the most advanced computational aeroacoustics methods. Both static conditions and flight conditions are reproduced. A thorough study of the computed results involving static and flight conditions leads to a physical explanation of the observed difference in the sound radiation patterns. (Invited for presentation in the Aeroacoustics Session)

10:00

4aHT3. Numerical simulation of grazing incidence of sound waves on an acoustic liner. Christopher Tam and Nikolai Pastouchenko (Department of Mathematics, Florida State University, Tallahassee, FL 32306-4510, tam@math.fsu.edu)

Acoustic liner is extremely effective for suppressing fan noise of jet engines. A resonant acoustic liner consists of a face sheet with cavity backing. Numerous small holes are drilled on the face sheet. When a sound wave is incident on a liner, pressure on the liner surface alternates from high to low. At high pressure, fluid is forced into the liner cavities through the holes. At low pressure, the process is reversed. The oscillatory motion of the fluid masses at the hole-openings is crucial to the damping of the sound waves. However, the hole diameter is typically one millimeter or less. Because the holes are small, experimental measurements of the fluid motion around the hole-openings are difficult to perform. Hence, this task is best carried out by numerical simulation; as small holes are not detrimental to numerical computation. The objective of this investigation is to seek an understanding of the flow physics responsible for acoustic damping by a liner. In all previous investigations, only one resonator is simulated. In this study, a liner with eight resonators is simulated. This allows, for the first time, a study of the aggregated effect of multiple resonators on an acoustic field. (Invited for presentation in the Aeroacoustics Session)

10:20

4aHT4. Simulation of compressible flows using Hermite methods. Thomas Hagstrom (Southern Methodist University, P.O. Box 750156, Dallas, TX 75275-0156, thagstrom@smu.edu), Daniel Appelo (The University of New Mexico, Albuquerque, NM), Tim Colonius, Matthew Inkman (California Institute of Technology, Pasadena, CA), and Chang Youn Jang (Southern Methodist University, Dallas, TX)

Spectral element methods based on Hermite interpolation have a number of unique properties. First of all, the stabilization inherent in the interpolation process is sufficient to suppress nonlinear instabilities observed with other discretization schemes and leads to accurate linear transport of nonsmooth solutions. Second, and most important, they allow purely local time-stepping procedures limited only

by geometric domain-of-dependence requirements. Thus high-order Hermite methods maximize the computation-to-communication ratio and therefore they admit highly efficient implementations on multicore processors. In this talk we focus on the application of Hermite methods to simulate unsteady compressible flows. Examples will include the direct simulation of the aeroacoustics of a low Reynolds number subsonic jet, as well as studies of more basic sound radiating flows. The latter will illustrate the coupling of Hermite methods with more standard discontinuous Galerkin discretizations to handle physical boundaries.

10:40–11:00 Break

11:00

4aHT5. Effect of Riemann flux solver on the accuracy of spectral difference method for CAA problems. Junhui Gao, Xiaodong Li (Beihang University, Beijing, China, gaojhui@buaa.edu.cn), and Qiqi Wang (Massachusetts Institute of Technology, Cambridge, MA 02139, U.S.A.)

The spectral difference (SD) method is a new high-order method for unstructured grids proposed recently by Liu et al. (2006). In this paper, a two dimensional computation aeroacoustics (CAA) tool based on SD method is developed. Five Riemann solvers are implemented in the current code, including Roe scheme, advection upstream splitting method (AUSM), flux-vector-splitting scheme, Rusanov scheme, convective upwind and split pressure (CUSP) scheme. A comparison of these Riemann solvers is carried out with three CAA workshop benchmark problems. The relative error of each solver in simulating of entropy, vorticity and acoustic waves is presented. The accuracy of the SD method with each Riemann solver is obtained. It is found that the usually used Rusanov scheme is less accurate than other solvers. AUSM and CUSP schemes are more accurate than others in simulating acoustic waves. Meanwhile, the effect of mesh quality on the accuracy of SD method is investigated. Gaussian distributed random error is superimposed on a base mesh to change the mesh quality. The accuracy of each solver on the skewed mesh is presented and compared with the results on base mesh. It is shown that mesh quality has little effect on the accuracy of SD method if the mesh resolution is sufficient.

11:20

4aHT6. Assessment of nonlinear perfectly matched layer boundary conditions for CAA benchmark problems. Dakai Lin (Beijing Aeronautical Science & Technology Research Institute of COMAC, lindakai@comac.cc), Xiaodong Li (School of Jet Propulsion, Beihang University, China), and Fang Q. Hu (Department of Mathematics and Statistics, Old Dominion University)

Non-Reflecting Boundary Conditions (NRBCs) are very crucial for accurate numerical simulation of aeroacoustic problems. This paper aims to assess the performances of recently developed nonlinear Perfectly Matched Layer (PML) NRBCs by several Computational Aeroacoustics (CAA) benchmark problems through the comparison with the linearized PML, the characteristic and the asymptotic NRBCs. Numerical results show that the performances of the nonlinear PML NRBCs are tantamount to each other, and there is no substantial difference. But for strongly nonlinear cases, the error caused by using nonlinear PML NRBC is 1~2 orders of magnitude smaller than the one caused by using the linearized PML NRBC. Thus, using nonlinear PML is necessary in strong nonlinear aeroacoustic problems. Numerical tests also demonstrate that the nonlinear PML NRBCs outperform the characteristic NRBCs significantly, and have better performances than the asymptotic NRBCs.

11:40

4aHT7. Control of edge-scattering noise via permeable surfaces. Young J. Moon, Ikhyun Bai, and Seungtae Hwang (Korea University, School of Mechanical Engineering, Seoul 136-701, Korea, yjmoon@korea.ac.kr)

The edge-scattering noise generation mechanism is first studied, in line with the existing thoeries of Howe, Amiet, and others. Then the edge-scattering noise is controlled by attempting various permeable edges such as porous surfaces and slitted edges. The basic underlying mechanism of noise reduction is to be understood, examining the three-dimensional scattering of a line-vortex embedded in the laminar boundary layer over the flat plate with the porous and slitted trailing-edges. More realistic investigations will follow by the large-edge simulation (LES) of a turbulent boundary layer over the flat plate, solving the filtered, three-dimensional, compressible Navier-Stokes equations with the six-order compact finite-difference scheme and the four-stage Runge-Kutta method.

12:00

4aHT8. Control of weak perturbations. J. E. Ffowcs Williams (Emmanuel College, University of Cambridge, Cambridge CB2 3AP, U.K, jef1000@cam.ac.uk), and L. Huang (Dept of Mechanical Engineering, University of Hong Kong, China)

We define sound as being a weak perturbation in the properties of material consistent with the Navier-Stokes and continuity equations. Lighthill's pioneering paper on aerodynamic noise gives an exact theory that enables interesting connections to be made between flow and sound. Aerodynamic noise being caused by quadrupoles is a good point of view, but what caused the quadrupoles? Were they possibly initiated by sound? Conclusions deduced from such a theory are not necessarily helpful, but they are true and might be very helpful indeed. The linear perturbations we call sound obey linear rules and it can be suppressed by anti-sound, a subject now becoming both practical and useful. The same must apply to any weak perturbation of a dynamic system perturbed from rest. Some perturbations are unstable and grow exponentially in their early weak state. They might be eliminated altogether by suppressing their linear form. The Rijke tube experiment shows that to be practical and shows also the close similarity that exists between acoustics and control theory. The lecture will give more examples of that type and suggest others that have yet to be demonstrated.

Contributed Papers

12:20

4aHT9. A new algorithm for deghosting in passive acoustic air surveillance systems. Xuelei Zhang, Jie Feng, and Zhaoli Li (The Third Research Institute of China Electronics Technology Group Corporation, Beijing 100015, China, zhangxuelei2008@gmail.com)

This paper addresses the false association (called ghosts) problem of multi-target tracking in a distributed passive acoustic sensor network. To eliminate these ghosts, a new deghosting scheme based on the generalized triangulation has been proposed. First, the received angle information of different targets is reordered by using the gray theory to obtain the correct sequence. Successively, time span match triangulation based on pretreatment algorithm of angle association is used to get the most-likely position of different targets. Last, the residual ghosts are cancelled by using a reasonable hypothesis and third site notarization. The validity of the proposed scheme is evaluated using both simulation and experimental results.

12:40

4aHT10. A robust selection method of time-delay difference for DOA estimation. Zhiyu Li, Zhiguo Hou, and Zhaoli Li (The Third Research Institute of China Electronics Technology Group Corporation, lizhiyu79@126.com)

DOA estimation method base on time-delay differences is used widely in the field of passive source location because its solidity in interference circumstance. To meet the real-time demands of some location systems, a time-delay differences robust selection method is presented in this paper by analysis of DOA estimation errors. In this method, only part of time-delay differences is estimated for the measuring of DOA to reduce the computational complexity. Computer simulation results indicate the new method has less computational cost by contrast with traditional method, and based on the reasonable time-delay differences the DOA estimation results are more accurate.

THURSDAY MORNING, 17 MAY 2012 S425, 9:15 A.M. TO 12:20 P.M.

Session 4aID

Interdisciplinary: Workshop on Publishing Excellence in the Journal of the Acoustical Society of America

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Chair's Introduction—9:15

Invited Papers

9:20

4aID1. Gauging the likelihood for acceptance of a paper submitted to the Journal of the Acoustical Society of America. Allan Pierce (Acoustical Society of America, P.O. Box 339, East Sandwich, MA 02537, adp@bu.edu)

Authors contemplating submitting papers to the Journal of the Acoustical Society of America (JASA) should first determine whether JASA is an appropriate journal, as it is possible that submissions for which this is not the case will be immediately rejected. A principal criterion is whether the paper will find a wider readership with JASA publication than with an alternative journal. If the appropriateness for JASA may not be manifestly obvious to the editors, then the authors should submit a cover letter explaining why, and they should write their paper so that there is a clear tie-in with articles previously published in JASA, preferably recent articles. Authors are advised to select the Associate Editor who is most likely to be familiar with the subject matter of the paper. Given that the paper is appropriate and an Associate Editor can be identified who is willing to handle the paper, it will be subsequently judged for possible acceptance based on several criteria: the most important being whether the paper is (i) original, (ii) significant, (iii) clearly written, and (iv) suitably limited in scope. The significance criterion is discussed at some length. The talk is illustrated by several disguised examples of recent submissions which were rejected.

9:40

4aID2. Understanding the peer-review process in the Journal of the Acoustical Society of America. Ning Xiang (Graduate Program in Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, New York, 12180, xiangn@rpi.edu)

The Journal of Acoustical Society of America (JASA) is an archival, peer-reviewed journal that has served the acoustics community for over 80 years. A rigorous peer-review process often results in significantly improved manuscripts qualified for publication; it also allows relative academic freedom and fairness. To help prospective authors disseminate their research results and progress, the JASA editorial board regularly publishes detailed, updated information, and guidelines. In addition to following these guidelines, a better understanding of the peer-review process is also critically important for prospective authors. This talk will provide some insights into the different roles and the interrelationship between the author(s), and the reviewers. This talk will also discuss criteria for accepting manuscripts, and engagement of knowledgeable reviewers using some disguised examples.

10:00

4aID3. One person view on publishing in JASA and JASA Express Letters. Whitlow W. L. Au (Hawaii Institute of Marine Biology, 46-007 Lilipuna Road, wau@hawaii.edu)

One of the strengths of the Acoustical Society of America (ASA) is the multi-disciplinary aspects of the society. Engineers, physicists, mathematicians, biologists, psychologists, physiologists, and speech researchers plus others are part of our society. The ASA can be broadly divided into two groups, those in the physical sciences and those in the life or natural sciences with signal processing bridging both groups. JASA reflects this diversity. Manuscripts by those in the physical sciences depend heavily on mathematical and physical modeling and various types of equations, while in the natural sciences spectrograms, various types of statistics and hypotheses testing are often used. Nevertheless, certain factors with regard to quality must be met by all authors. The articles should contribute new knowledge, be written in English and be grammatically correct. The approach or methodology must be sound. The results should be clearly presented whether in tables, graphs or in verbiage. Finally, the discussion and conclusions should be to the point, clearly presented so the authors' arguments and points can be easily understandable. Associate editors should be assisting authors in getting their papers published. Responsibility for the content lies totally with the authors and not one bit with the associate editors.

10:20

4aID4. Publishing in the Journal of the Acoustical Society of America. Zhaoyan Zhang (UCLA School of Medicine, 31-24 Rehab Center, 1000 Veteran Ave., Los Angeles, CA 90095-1794, zyzhang@ucla.edu)

Peer-reviewed journals such as the Journal of the Acoustical Society of America provide researchers the best venue to distribute research accomplishments to a broad readership. While this publishing process may be time-consuming and sometimes frustrating, authors can take an active control of this process by preparing a good-quality manuscript and acquiring a better understanding of the peer-review process. The purpose of this talk is to share the present author's experiences as an JASA author and a current JASA associate editor in preparing a publishable manuscript for JASA. Suggestions are given on how to prepare a scientifically significant manuscript and how to benefit from the peer-review process to further improve the manuscript.

10:40–11:00 Break

11:00

4aID5. Authors' sharing on handling reviewers' comments. S.K. Tang (Department of Building Services Engineering, The Hong Kong Polytechnic University, besktang@polyu.edu.hk), and L. Cheng (Department of Mechanical Engineering, The Hong Kong Polytechnic University)

JASA is a highly reputable journal in acoustic community and publishing in the journal is very important to academia. In the universities in Hong Kong at least, a publication in JASA could affect staff appraisal outcome. The journal has very professional editors and a secretariat office, who screen all the submissions before sending them out to reviewers. To publish in JASA, quality of the work is certainly of prime importance. However, the responses to reviewers' comments can sometimes play an important role in the process, especially on controversial issues. We believe that authors and reviewers are equal in the whole process but comments from both the editors and reviewers must be handled professionally. In this presentation, we would like to offer our views from an author's perspective. In particular, we would like to share some experiences and discuss some issues which we think could affect the final decision on a submission.

11:20

4aID6. Guidelines for prospective authors to submit acceptable manuscripts to the Journal of the Acoustical Society of America. Sean F. Wu (Wayne State University, sean_wu@wayne.edu), and Ning Xiang (Rensselaer Polytechnic Institute, Graduate Program in Architectural Acoustics, 110 8th Street, Troy, New York, NY 12180)

The Editor-In-Chief of the Journal of the Acoustical Society of America (JASA) has recently compiled a list of problems in the manuscripts submitted to JASA that often lead to an outright rejection by the Associate Editors handling their review. These problems often occur during submission and writing of a manuscript. They include selecting a title, listing the authors, composing an abstract, defining the scope of work, presenting the background and significance of the research, reporting and discussing the major discovery, ideas and results, showing the work and results that have been published by others in other journals already, drawing concise conclusions, citing references, displaying equations, figures, tables, etc. Last but not the least is the English writing that should be grammatically correct and easy to understand by someone with a similar background. This talk gives a quick overview of these potential problems, which are frequently shown in the manuscripts submitted by authors overseas. The goal of this talk is to provide helpful suggestions and guidelines to the prospective authors whose native language is not English to submit manuscripts that can pass the initial screening and ultimately get published in JASA. Disguised examples of some problematic manuscripts are discussed and analyzed.

11:40

4aID7. Discussions on publishing excellence in the Journal of the Acoustical Society of America and JASA Express Letters. Ning Xiang (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, Troy, New York, xiangn@rpi.edu), and Li Cheng (Department of Mechanical Engineering, The Hong Kong Polytechnic University, Hung Hom, Hong Kong)

This workshop involves a number of invited speakers from the editorial board members of the Journal of the Acoustical Society of America (JASA, including JASA Express Letters) and from representative JASA paper authors to discuss the JASA peer-review process, the criteria for accepting manuscripts, and successful preparation of manuscripts for JASA publications. Following the presentations by the invited speakers, a panel discussion offers a platform for those in the audience, particularly potential authors, to ask relevant questions and for the panelists to give multiple replies. The panelists also include a liaison from the American Institute of Physics to the Acoustical Society of America for all publishing affairs. [The panelists: Whitlow Au, Li Cheng, Robert Harington, Allan Pierce, S.K. Tang, Sean Wu Ning Xiang, Zhaoyan Zhang]

Session 4aMU

Musical Acoustics and Psychological and Physiological Acoustics: Musical Timbre: Perception and Analysis/Synthesis I

James W. Beauchamp, Cochair
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Andrew B. Horner, Cochair
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Invited Papers

9:20

4aMU1. Real-time segmentation of the temporal evolution of musical sounds. John Glover, Victor Lazzarini, and Joseph Timoney (National University of Ireland, Maynooth, john.c.glover@nuim.ie)

Since the studies of Helmholtz, it has been known that the temporal evolution of musical sounds plays an important role in our perception of timbre. The accurate temporal segmentation of musical sounds into regions with distinct characteristics is therefore of interest to the study of timbre perception as well as to different forms of sound modelling and manipulation. Following on from recent work by Peeters and Caetano et al, this paper presents a new method for the automatic segmentation of the temporal evolution of isolated musical sounds in real-time. We define attack, sustain and release segments using cues from a combination of the amplitude envelope, the spectro-temporal evolution and a measurement of the stability of the sound that is derived from the onset detection function. We conclude with an evaluation and discussion of some potential applications of the method.

9:40

4aMU2. Impact of MP3-compression on timbre space of sustained musical instrument tones. Chung Lee, Andrew Horner (Hong Kong University of Science and Technology, im.lee.chung@gmail.com), and James Beauchamp (University of Illinois at Urbana-Champaign)

MP3 compression is widely used in music sharing and storage. A number of studies have investigated the discrimination of instrument tones after MP3 compression. Additionally, a number of previous studies have evaluated other data reduction methods including frequency modulation (FM) synthesis, wavetable synthesis, and principal component analysis (PCA). However, these studies have not considered the impact on timbre space after data reduction. In this study, listening test subjects were asked to rate the dissimilarity of all pairs of original instrument tones. The same process was done on MP3-compressed tones with various bit-rates. Correlation analysis was done on the dissimilarity data of the original and compressed tones to see if MP3 compression caused a significant impact on the perceptual distance between instrument pairs. The multidimensional scaling (MDS) solutions of the original and compressed tones were also compared to see if the timbre space was significantly altered after MP3 compression (e.g., would a clarinet sound more or less similar to an oboe after MP3 compression?) [This work was supported by RGC grants 613510 and 613111.]

10:00

4aMU3. Investigation of timbre saliency, the attention-capturing quality of timbre. Song Hui Chon and Stephen McAdams (CIRMMT, Schulich School of Music, McGill University, 555 Sherbrooke Street West, Montreal QC, Canada H3A 1E3, songhui.chon@mail.mcgill.ca)

Timbre saliency is defined as the attention-capturing quality of timbre. Saliency differences between timbres were measured using a tapping technique in which the stronger beat in ABAB isochronous sequences was reproduced by the listener, the idea being that the more salient timbre would capture listeners' attention and be chosen more often as the strong beat. A timbre saliency space was defined in which the distance between a pair of timbres corresponded to the difference in timbre saliency. Stimuli were generated with 15 orchestral instruments, equalized in pitch, loudness and duration. Data from 40 participants yielded a one-dimensional CLASCAL solution with two latent classes and specificities. Latent class structure shows no relation with gender, musicianship or age. Testing audio descriptors from the Timbre Toolbox [Peeters et al., 2011, J. Acoust. Soc. Am., 130, 2902-2916], the odd-even harmonic energy ratio explains 51% of the variance along this dimension. A combination of tristimulus (band 3) and odd-even ratio explains 73% of the variance in the mean saliencies of individual sounds across all other comparison sounds. Mean saliency thus seems to depend on the high-frequency harmonic energy and spectral envelope jaggedness, whereas saliency comparisons between timbres depend more on spectral envelope jaggedness.

10:20

4aMU4. Toward an effective use of timbre in data sonification. Hiroko Terasawa (University of Tsukuba/JST-PRESTO, terasawa@tara.tsukuba.ac.jp)

The spectro-temporal structure of a sound determines its timbre, and carries musically interesting information such as instrument type and performance expressions. Using timbre in data sonification can be viewed as an inverse transform of this process: Expressing data with timbre is equivalent to designing the spectro-temporal structure of a sound. Taking that into account, timbre is most effectively used in sonification by projecting time-series data onto the spectro-temporal structure of a sound. The temporal structure of the data

often differs from the archetypal spectro-temporal structure of traditional instrumental sounds. But this discrepancy contributes to novel musical expression, based on novel timbre design. From this perspective, some sonification works are presented, such as ones produced by sonification of dynamic motion of genetically-modified worms and dynamic transitions of brain-wave data. Based on these examples, methods for effective and expressive use of timbre in data sonification will be presented.

10:40–11:00 Break

11:00

4aMU5. Wind instrument sound design with centroid-controlled spectral template synthesis. Simon Wun, Andrew Horner (HKUST, simonwun@ust.hk), and James Beauchamp (UIUC)

Most previous sound synthesis research has been oriented to instrument imitation and data reduction, whereas perceptual control has received little attention. In other words, the parameters in traditional sound synthesis are not perceptually meaningful. Using common synthesis techniques such as multiple wavetable synthesis and additive synthesis, we have no obvious way to imitate acoustic tones while allowing perceptual control. An important perceptual parameter is spectral centroid, which strongly correlates with a sound's brightness. Spectral centroid and attack time are two universally-recognized timbral features that strongly influence discrimination and identification of musical instruments. This paper investigates the generality and effectiveness of spectral template synthesis, a perceptually-based technique for synthesizing wind instrument tones. Synthesis from spectral envelope templates is driven by spectral centroid or other control functions. Control by spectral centroid has the advantage of direct manipulation of a perceptually-salient feature. Unlike other synthesis techniques, spectral template synthesis is designed to track changes in spectral centroid and mimic acoustic tones at the same time. This work has application in the synthesis of natural realistic sounds that go beyond the normal timbral boundaries of acoustic instruments. [This work was supported by RGC grants 613510 and 613111.]

11:20

4aMU6. Relating timbre discrimination to perceptual distances between interpolated percussive timbres. William L. Martens and Mark McKinnon-Bassett (Faculty of Architecture, Design and Planning, University of Sydney, NSW 2006, william.martens@sydney.edu.au)

A set of percussive timbres was generated using a hybrid resynthesis that was based upon the analysis of recorded conga and bongo drums. Comprising the set were drum timbres resulting from parametric variation in both damping of the low-frequency resonance associated with pitch of the drum, and variation in a higher-frequency resonance associated with percussive attack transients. Listeners were presented with all pairwise comparisons of the synthetic drum sounds, and were asked first to perform timbral discriminations for each pair, and subsequently to produce pairwise dissimilarity judgments. Underlying perceptual scales values were derived for each timbre from discrimination performance along the two manipulated stimulus dimensions, and these values predicted well the perceptual distances that were fit to the stimulus space coordinates derived from the dissimilarity judgments. Taken together, the results provide a basis for developing a reliable control structure for the synthesis of such percussive timbres.

Contributed Papers

11:40

4aMU7. Individual differences in the relative salience of percussive timbre dimensions. Mark McKinnon-Bassett and William L. Martens (Faculty of Architecture, Design and Planning, The University of Sydney, NSW 2006, mbas4365@uni.sydney.edu.au)

A set of nine percussive timbres was generated by varying parameters of two resonant filters incorporated in a hybrid resynthesis of recorded drum sounds. Three values of damping for a lower-frequency resonance were factorially combined with three center-frequency values for a higher-frequency resonance associated with percussive attack transients. Two groups of listeners were asked to produce dissimilarity judgments for all pairwise comparisons of the nine sounds on a ten-point scale. The dissimilarity judgment data from two groups of subjects were combined to form a single dataset for submission to Individual Differences Scaling (INDSCAL) analysis. A common timbre space of just two dimensions was derived along with a subject space that revealed the different weights placed by each subject on each of those dimensions of the derived timbre space. Individual differences in the relative salience of these percussive timbre dimensions were related to the musical training of the listeners.

12:00

4aMU8. A new sinusoidal model for synthesis of musical instruments. Sudhendu Raj Sharma (Purdue University, School of Electrical and Computer Engineering, Electrical Engineering Building, 465 Northwestern Ave., Mailbox 429, West Lafayette, IN 47907, sharmasr@purdue.edu), Zhenhao Ge, and Mark J. T. Smith (Purdue University, School of Electrical and Computer Engineering, Electrical Engineering Building, 465 Northwestern Ave., West Lafayette, IN 47907)

Many algorithms have been developed over the years to synthesize acoustic sounds and are now used commercially in acoustic synthesizers and digital keyboard products. The issue with these algorithms is the trade-off among sound fidelity, algorithm complexity, and hardware/storage requirements-the last two of which are directly related to the cost of the system. In this reported work, we introduce a new sinusoidal model for digitally synthesizing musical instruments. The new algorithm has unusually high fidelity, minimal memory requirements, and high computational efficiency. The algorithm is based on the "analysis-by-synthesis overlap add" (ABS/OLA) sinusoidal model, which models musical sounds as a short-time weighted sum of constant frequencies, phases, and amplitudes. The new model we introduce incorporates a novel dynamic pitch and frequency control feature in synthesis that allows very high quality instrument sounds to be generated over a wide range of pitches from a very short sampled recording of the musical instrument. Sound modifications can be performed parametrically within the framework all using fast Fourier transforms (FFTs) for high efficiency. Examples of synthetically generated non-western musical instruments will be presented during the conference and contrasted with competing technologies to illustrate the advantages of the new method.

Session 4aNSa

Noise, Engineering Acoustics, Physical Acoustics, and Signal Processing in Acoustics: Active Noise Control II

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Invited Papers

9:20

4aNSa1. Challenges in the implementation of active noise control technologies. Xiaojun Qiu and Ningrong Li (Institute of Acoustics, Nanjing University, xjqiu@nju.edu.cn)

A number of projects have been carried out in Nanjing University to implement active noise control technologies, which include active control of transformer noise, active sound barrier, active noise control in communication chassis, active noise control in natural ventilation windows, active control of large impulsive noise in headset and active noise control in a train compartment. Unlike our previous research, these projects are all funded by industries and the aim is not for doing academic research, but to make commercial prototypes. Various challenges in the implementation of active noise control technologies in these projects to make commercial products are reported and discussed, and main issues to make successful commercial active noise control products are pointed out.

9:40

4aNSa2. On family of fractional lower order moment (FLOM)-based algorithms for active noise control of impulsive noise sources. Muhammad Akhtar (The Center for Frontier Science and Engineering (CFSE), The University for Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan, akhtar@ice.uec.ac.jp), and Wataru Mitsuhashi (Department of Communication Engineering and Informatics, The University for Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan)

Active noise control (ANC) is based on the principle of destructive interference of propagating acoustic waves; essentially a cancelling signal is generated and combined with the primary noise to achieve acoustic cancellation around location of the error microphone. In this paper, we consider a very challenging application of ANC for impulsive noise. The impulsive noise can be modeled using non-Gaussian stable process for which second order moment does not exist. The most famous filtered-x-LMS (FxLMS) algorithm for ANC systems, is based on minimization of the variance of the error signal, and therefore, becomes unstable for the impulsive noise. It has been shown that filtered-x least mean p-power (FxLMP) algorithm; based on minimizing the fractional lower order moment (FLOM) that does exist for stable distributions; gives robust performance for impulsive ANC. However, the convergence speed of the FxLMP algorithm is very slow. Recently; we have proposed various variants of FxLMP algorithm, so that an improved convergence and noise reduction performance is achieved. In this paper, we propose modifying and employing generalized normalized LMP algorithm (GNLMP) algorithm for ANC of impulsive noise. The computational complexity of proposed algorithm is comparable to the existing FLOM-based ANC algorithms. Extensive simulations are carried out, which demonstrate the effectiveness of proposed algorithm. We observe that, in comparison with the existing FLOM-based ANC algorithms, the proposed algorithm gives best performance for ANC of impulsive noise sources.

Contributed Papers

10:00

4aNSa3. Performance of active noise barrier with a moving sound source. Jiancheng Tao, Yiqing Deng, and Xiaojun Qiu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, jctao@nju.edu.cn)

Active noise control is an effective technology to improve noise reduction performance of traditional passive noise barriers at low-frequencies. Previous researches on active noise barriers are mainly on the assumption that the primary sound source is stationary. Effects of source motion on the performance of an active noise barrier are investigated in this paper. First, an analytical model of a passive barrier with a moving sound source is introduced, which can be used to calculate the primary sound field distribution in time domain. Then, the performance of applying active noise control on such a barrier is investigated numerically. Finally, the experimental results with a practical prototype of active noise barrier will be reported and be compared with the numerical results.

10:20

4aNSa4. Active control of exhaust noise using an air horn. Dongki Min, Deokman Kim, and Junhong Park (Hanyang University, 133-791, dkmin@hanyang.ac.kr)

The noise generated by internal combustion engine is reduced by passive muffling systems. For the passive muffling system, its performance significantly degrades especially when multiple low frequency tonal components exist in the flow. The tonal components occur from explosion process of the engine, and radiates as a monopole from the outlet. In this study, active noise cancelation using FxLMS algorithm is proposed to reduce the exhaust engine noise. Air-horn which is capable of being operated at high temperatures is proposed for cancelation of the radiated noise. The vibration input the diaphragm of the air-horn allowed the active control of the frequency and phase of the radiated sound. The FxLMS algorithm was used to actively control the sound radiation from the air-horn to achieve cancelation of the noise from the muffler. The sound radiation from the air-horn induces dipole-like noise radiation from the exhaust system, and significantly reduced the radiated sound power.

10:40–11:00 Break

11:00

4aNSa5. Research on decentralized adaptive active control for a single-layer vibration isolation system. Fengyan An, Hongling Sun, Xiaodong Li, and Jing Tian (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, anfy@mail.ioa.ac.cn)

In this paper, an investigation on decentralized adaptive active control for a single-layer vibration isolation system is introduced. The vibration isolation system consists of a honeycomb table as a vibration source, four rubber isolators in parallel with electromagnetic actuators, and another honeycomb table connected with a rigid base by four rubber isolators as a flexible base. Both simulations and experiments show that the decentralized controller exhibits good performances at frequencies where the tables could be dealt with as rigid bodies. At higher frequencies, however, the system could not work stably because flexible vibrations of the tables become dominant. An experimentally validated optimization method for internal parameters of the decentralized control algorithm is proposed to improve the stability and convergence of the control system.

11:20

4aNSa6. The design of a multi-channel active noise controller with ultra low latency. Kai Chen, Jing Lu, and Xiaojun Qiu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, chenkai@nju.edu.cn)

For wide band noise control, the performance of active noise control systems depends largely on the latency of the controller. The latency of the controller is usually caused by the AD and DA converter, and the normally used Delta-Sigma audio codec is not suitable for a real time active noise control system because several milliseconds time delay of the codec will lead to non-causality of the whole control system. In this paper, an ultra low latency multi-channel audio input and output system is described. The data of the system can be interacted with a float-point DSP, where a high efficient multi-channel feed-forward control algorithm is embedded. The proposed controller also includes an ARM processor, which is in charge of the friendly user interface.

11:40

4aNSa7. Novel application of PVDF film in active noise control through windows. Jeremy Lane (NZi3, University of Canterbury, 69 Creyke Rd, Ilam, Christchurch 8041, jeremy.lane@pg.canterbury.ac.nz), John Pearse, and Stefanie Gutschmidt

PVDF film has been used widely used in active control solutions for noise and vibration. In this work, due to the transparent property of PVDF film and the proven possibility of transparent electrodes, the feasibility of PVDF film's use in the construction of a second source for active noise control (ANC) through windows is considered. Sound pressure level measurements are described to establish the feasibility of PVDF in this application. Different configurations, using glass and acrylic glass backing materials of varying thicknesses, of PVDF film speaker are reported and compared. Finally comments are made on relative performance and the overall likelihood of use in an ANC application for windows.

12:00

4aNSa8. A broadband active control algorithm without cancellation path modeling. Min Gao and Xiaojun Qiu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, gaomin1221328@163.com)

Recently, an active control algorithm without cancellation path modeling has been investigated, which does not require identification of the cancellation path. The algorithm adopts the standard LMS to update the adaptive filter coefficients, but unlike the FXLMS algorithm, the reference signal does not need to pass through the cancellation path model and the proper update direction of the adaptive filter coefficients is chosen automatically by monitoring the excess noise power. Simulation and experimental results show that the algorithm works well to sinusoidal noise, multi-tune noise and narrowband noise within 40 Hz bandwidth, with similar noise reduction performance to that of the FXLMS algorithm. However, it is found that the algorithm does not work well for broadband noise and wider narrowband noise. Aiming at this, this paper investigates the mechanism of the algorithm for broadband noise and explores potential solution to the problem.

12:20

4aNSa9. An investigation on passive-active absorption system in a water-filled impedance tube. Xiaolin Wang, Bilong Liu, and Jing Tian (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, wangxiaolin@mail.ioa.ac.cn)

A hybrid passive-active sound absorption structure is experimentally investigated in a water-filled impedance tube. The surface impedance of the passive material is manipulated by active parts to match the impedance of medium, and consequently to improve sound absorption performance at low-frequency range. Parameters such as boundary, material thickness and position of secondary source are optimized by numerical analysis to achieve high absorption coefficient. Experimental results show that the developed hybrid structure has potential to improve sound absorption at low frequency range.

Session 4aNSb

Noise, Architectural Acoustics, Animal Bioacoustics, and ASA Committee on Standards: Soundscape and Its Application I

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Invited Papers

9:20

4aNSb1. Why soundscape? The new approach to "measure" quality of life. Brigitte Schulte-Fortkamp and Kay Voigt (Technische Universität Berlin, Germany, bschulte_f@web.de)

It is now about 15 years that Soundscape came into the field of community noise and sound quality. The Soundscape approach has provided essential knowledge for the demanding tasks which are required for the design and planning of sustainable environments to support to wellbeing, health, and quality of life, respectively. The multidimensional Soundscape approach puts emphasis on the way the acoustic environment is perceived, experienced and understood by the individual and by society (ISO/TC 43/SC 1/WG 54). Moreover, it accounts for people's concerns and integrates the exposed people as experts. The process of tuning of noise pollution or sound design with respect to the expertise of people's mind is related to the strategy of triangulation of interdisciplinary data. Moreover, the Soundscape approach provides the frame work to integrate contextual and subjective variables to improve the respective Soundscape with regard to people's expertise. This paper will highlight the process of Soundscape and its application with respect to ISO/TC43/SC 1/WG 54 and the COST network TD0804 on Soundscape and Landscape with regard to its implementation and dissemination in the diverse fields of acoustic environments and its definitive meaning concerning quality of life.

9:40

4aNSb2. Perceived soundscapes in relation to transport related annoyance, context and personal characteristics; psychometric analyses. Irene van Kamp (MGO, RIVM, P.O. Box 1, 3720 BA, irene.van.kamp@rivm.nl), Elise van Kempen, and Danny Houthuijs

Most studies into perceived soundscapes have addressed subjective soundqualities at a (very) low scale level, such as parks, recreational area's and squares. Studies into the effects of transport related noise seldom incorporate perceived soundscapes and are typically focussed on negative effects, such as annoyance, sleepdisturbance and environmental worry. It would be valuable to know how people describe their sound environment in areas with varying levels of road-, air or rail noise. Available data on perceived soundscapes from the two Schiphol surveys in 2002 and 2005 allowed us to perform such analyses. The intercorrelations were studied between dimensions of perceived soundscapes, annoyance, arousal as well as several measures at the contextual, social and psychological level, sometimes referred to as non-acouticsl factors. Results shed light on the construct validity of the perceived soundscape scale and may contribute to further refinement of this instrument.

10:00

4aNSb3. The link between soundscape perception and attention processes. Fiebig André (HEAD Acoustics GmbH, Ebertstr. 30a, 52134 Herzogenrath, Germany, andre.fiebig@head-acoustics.de)

In order to understand the perception and evaluation of soundscapes, it appears mandatory to concentrate not only on constellations of sources and their contributions to the acoustic environment, but also to consider attention processes towards sound sources. It is widely known that a listener can easily focus on a certain source and can suppress the noise of other sources, which is called cocktail-party effect. It is assumed that this effect greatly influences the general appreciation of the whole soundscape. However, the process, why people focus on certain sound sources and how this influences the overall evaluation, has to be explored. A detailed knowledge about the (often subconscious) focussing on sources in multi-source soundscapes would be very helpful for design purposes, to attract deliberately attention to certain sound sources leading to positive feelings for the majority of soundscape visitors. Laboratory results dealing with the effect of source attention and its impact on soundscape evaluation were already published. In these surveys it was found that the processes, in which way the global impression changes due to the attention attraction to certain sources, seem to be complex. The paper will focus now on in-situ assessments and will show new results gained in field experiments.

10:20

4aNSb4. From noise control to sound design: the class room as a soundscape project. Juergen Bauer (Waterford Institute of Technology, Ireland, jbauer@wit.ie)

As part of an overall campus building project, the Department of Architecture in Waterford Institute of Technology in Ireland moved to provisional premises in autumn 2011, in a city centre former warehouse, dating from 1875. While this building is a fine example of historic industrial architecture which was previously used successfully as a museum, as a school venue it is "acoustically seen" inappropriate. The studios are more halls rather than rooms and have an approx. height of 5 meters; two classes share one unit and are subdivided by screens, with lectures and tutorials needing to be scheduled at different times in order to avoid (acoustic) clashes. Most surfaces are hard, and in some cases, the class units are even exposed to open galleries and circulation areas. How can the noise problem be transformed into a soundscape project? How can the current situation be used to develop sound as a design tool that informs the awareness about sound phenomena, strengthen the understanding of sound mitigation and instill the confidence to design it? This paper investigates different approaches as to how to introduce sound as a design tool in early architectural education and summarizes the learning outcomes from using the class room as a sound design lab.

10:40–11:00 Break

11:00

4aNSb5. A case study of soundscape design based on acoustical investigation. Hui Ma and Sen Zhang (School of Architecture, Tianjin University, No. 92, Weijin Road, Nankai District, Tianjin 300072, China, mahui@tju.edu.cn)

Jinwan square is an important part of Haihe river, mother river of Tianjin, a Chinese city. In order to create a lively and comfortable environment for Jinwan square, soundscape design is necessary because the acoustical situation in this square is evaluated to be noisy and boring. Based on the sound investigation held in four seasons including sound level, sound type, the relationship between different sounds and sound expectation, a soundscape design focusing on adding natural sounds and controlling both road traffic noise and construction noise was done. Through this study, the process and the method of how to do a soundscape design in certain area were tried to be concluded.

11:20

4aNSb6. Human-machine interaction as influencing factor of indoor soundscape evaluation. Jochen Steffens (Duesseldorf University of Applied Sciences, ISAVE, Josef-Gockeln-Strasse 9, 40474 Duesseldorf, Germany, jochen.steffens@fh-duesseldorf.de), Brigitte Schulte-Fortkamp (Technische Universität Berlin, ISTA, Einsteinufer 25, 10587 Berlin, Germany), and Joerg Becker-Schweitzer (Duesseldorf University of Applied Sciences, ISAVE, Josef-Gockeln-Strasse 9, 40474 Duesseldorf, Germany)

In many ways people interact with machines like vehicles or household appliances. Thereby they consciously and unconsciously perceive information about the machines' performance and operating state. Apart from visual or haptic feedback important information is transferred via the acoustical input. In order to design well-assessed indoor soundscapes like car interiors or kitchen it is essential to involve people in early product development processes. However, in order to evoke everyday perception processes in listening studies and to ensure ecological valid sound evaluations realistic human-machine interaction has to be reconstructed. Only under quasi-real conditions the user needs acoustical feedback from the used devices. This determines the user's attitude to the sound and thus to its evaluation to a large extent. Within this contribution case studies will be presented which expound the influence of interaction on cognitive, emotional, and motivational aspects within the sound evaluation in soundscape research.

11:40

4aNSb7. Soundscape case study: acoustics, ecology and its anthropological sense. Zhiyong Deng (College of Music, Capital Normal University, No. 105, Xisanhuan North Road, Beijing 100048, China, dzy@cnu.edu.cn), Guowen Zhou, Wei Hua (College of Music, Guangxi Arts Institute, Nanning 530022, China), Yun Wang (College of Music, Capital Normal University, No. 105, Xisanhuan North Road, Beijing 100048, China), and Jian Zhang (The 3rd Research Institute of CETC, Beijing 100015, China)

Base on six soundscape investigation case studies in some small urban or historical areas, included the South Putuo Temple of Xiamen city in 2006, the downtown of Kashkar city in 2006, the local worship music performing area of South Gaoluo village in 2007, the Nanxiangkou local music performing area in Hebei University of Technology Gymnasium of ShijiaZhuang city in 2007, the downtown riverside of Liuzhou city in 2010, the local worship and music performing areas of Qianjuntai village and Zhuanghu village of Beijing city in 2011, a certain relationship analysis between the acoustical parameters and its audience's behavior or subjective assessment is put forward in this paper. Due to the two-dimensional (Leq and the subjective assessments) fuzzy clustering or curve fitting, it shows that the sound ecology would have a critical connection to the anthropological sense, which means some keynotes sound and soundmarks must be keep constantly during the soundscape design or the urban development. Furthermore, a rough concept of sound history and the constant of keynotes and soundmarks in the view of soundscape are also discussed in this paper.

12:00

4aNSb8. Soundscape variation in a historical city centre due to new traffic regulation. Luigi Maffei, Maria Di Gabriele, and Francesco Aletta (Seconda Università di Napoli- Center RiAS- Via S.Lorenzo, 81031 Aversa (Italy), luigi.maffei@unina2.it)

In recent years the number of candidate historic city centers to be included in the World Heritage List is increasing. This inclusion must be supported by a Management Plan programming all intervention to be implemented for the preservation of the "outstanding universal value". So far the Management Plans do not consider the preservation and valorization of the soundscape. Consequently, urban renewal processes are based on conservation and restoration of tangible cultural heritage, in order to increase touristic attraction and to improve the quality of life. All these efforts privilege visual perception and do not take in account the auditory perception. Soundscape of a site can be considered an intangible cultural heritage to be preserved and valorized as it constitutes a peculiar characteristic of the place. It makes the place recognizable and attractive. Recently the historic centre of Naples (Italy), as World Heritage Site, has been under renewal and for sustainable mobility the largest restricted traffic area (ZTL) in Europe has been introduced. The results of soundwalks carried out in the historic center of Naples before and after the implementation of ZTL are presented. The variations of acoustical and other environmental parameters influencing the subjective perception of environmental quality are analyzed.

12:20

4aNSb9. Study on how to create a comfortable soundscape for commercial open space. Jianwei Song and Hui Ma (School of Architecture, Tianjin University, No. 92, Weijin Rd., Nankai District, Tianjin, China, boobu530@163.com)

Commercial open space plays a vital role in urban life and the soundscape design of those areas becomes a new problem worthy of researching. In this study, the sound situation of two famous commercial open areas in Tianjin, China, including sound type, sound expectation, and environmental evaluation was analyzed through physical measurement and social surveys. Finally, ten sound samples were obtained from those commercial open areas. Except noise level, both temporal and spatial factors of the sound samples were analyzed. Combined the laboratory experiments and sound signal analysis, the principle of how to create a comfortable soundscape in commercial open space was explored from block design, architecture style and material selection.

THURSDAY MORNING, 17 MAY 2012 S223, 9:20 A.M. TO 12:20 P.M.

Session 4aPA

Physical Acoustics and Engineering Acoustics: Emerging Technologies and Concepts in Ultrasonics

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Contributed Papers

9:20

4aPA1. Ultrasonic set up for the assessment of the stability of a cylinder inserted in a solid. Vincent Mathieu (CNRS, Université Paris-Est, Laboratoire Modélisation et Simulation Multi Echelle, UMR 8208 CNRS, 61 avenue du Général de Gaulle, 94010 Créteil, France, vincent.mathieu@u-pec.fr), Fani Anagnostou, Emmanuel Soffer (CNRS, Université Paris-Diderot, Laboratoire Bioingénierie et Biomécanique Ostéo Articulaire, UMR 7052 CNRS, 10 avenue de Verdun, 75010 Paris, France), and Guillaume Haiat (CNRS, Université Paris-Est, Laboratoire Modélisation et Simulation Multi Echelle, UMR 8208 CNRS, 61 avenue du Général de Gaulle, 94010 Créteil, France)

The study aims at proposing a new experimental ultrasonic methodology for the estimation of the stability of a cylinder inserted in a solid. Such a technology may have various fields of application: aeronautics, car industry, mechanics or also surgery. The present prototype is dedicated to the study of the stability of dental implants. Cylindrical titanium implants were inserted in four groups of rabbit femurs, each group corresponding to a controlled level of stability of the cylinders. The 10 MHz ultrasonic response of the cylinder is processed to derive quantitative indicators based on the temporal variation of the signal amplitude. Analysis of variance (ANOVA) ($p < 10^{-5}$) tests revealed statistical distributions of indicators significantly correlated with the stability of the cylinders. A numerical finite-difference time-domain model was considered in order to understand the origin of the different echoes and the importance of lateral wave propagation was evidenced. The numerical model also enabled to estimate the sensitivity of the indicators to variations in the material properties of the materials in contact with the cylinders.

9:40

4aPA2. The study of dissipative nonlinearity in oil sand. Jiehui Liu, Jinlin Zhu, Xiaozhou Liu, Xiufen Gong, and Dao Zhou (Key Laboratory of Modern Acoustics, Ministry of Education, Institute of Acoustics, Nanjing University, Nanjing 210093, China, wljh@nju.edu.cn)

The acoustic waves propagating in sand have the nonlinear dissipative phenomenon that the dissipative coefficient of acoustic waves with larger amplitude is smaller than that with smaller amplitude. The results of experimental investigations on the propagation of acoustic waves in oil sand with different oil content are presented in the paper. The nonlinear dissipative phenomenon in oil sand is studied and the analytical description is given to explain the phenomenon. It is found that the relative growth coefficient and the dissipative index are dependent on the oil content in oil sand. According to the dependence relationships between the sensitive coefficients and the oil content, a new prospective approach to measure the oil content in oil sand is provided in the paper for oil exploration.

10:00

4aPA3. Cell structure in waves diffracted by a wedge. Mitsuhiro Ueda (Predio Meguro Science Laboratory, 4-20-13 Meguro, Meguro-ku, Tokyo 153-0063, Japan, ueda-mt@nifty.com)

Waves diffracted by a wedge made of perfectly reflecting material exhibit characteristic spatial pattern depending on an aperture angle of the wedge. For examples, in the wedge of aperture angle π, that is, a perfectly reflecting plane and that of aperture angle $\pi/2$, that is, a corner cube, diffracted waves are identically zero. And in the wedge of aperture angle 2π, that is, a semi-infinite plane, diffracted waves are symmetric with respect to a central axis of the wedge. The relation between the pattern and the aperture angle, however, has not been studied in detail so far since there is no appreciate model for diffracted waves and the rigorous solution for diffracted waves is so complex that only the simplest case can be analyzed. We have proposed the new mathematical model for diffracted waves where they can be expressed as a sum of two more fundamental quantities called elementary diffracted waves. The new model reveals that cell structure exists in waves diffracted by the wedge of aperture angle π multiplied by a rational number less than 2. The cases mentioned above can be explained in terms of the cell structure.

10:20

4aPA4. Measurement of the sound pressure in the focal spot area of line-focus ultrasound field by Schlieren technique. Xue-Ping Jiang, Qian Cheng, and Meng-Lu Qian (Institute of Acoustics, Tongji University, 1239 Siping Road, Shanghai 200092, China, 0720106002@tongji.edu.cn)

Schlieren method is an effective method for studying the sound field in transparent medium which called phase objects. The method is used to research the acoustic field by analyzing the refractive index changes induced

by the acoustic wave. Calculations of the sound pressure distribution radiated by the line-focus ultrasonic transducer are implemented, and the acoustic field in the focal spot area is obtained. Then the diffraction light intensity on the Fourier transform plane in a Schlieren system is calculated with two-dimensional Fourier transformation when the phase object is the ultrasonic wave. Because the light intensity of the different diffraction spots on the back focal plane of the transform lens depends on the sound pressure, the sound pressure in the focal spot area can be determined using Schlieren technique. A Schlieren system is set up. The images of the line-focus ultrasonic field are obtained and the sound pressure in its focal point area is measured non-invasively by measuring and comparing the light intensities of the different diffraction spots. This work is supported by the National Natural Science Foundation of China (No. 10804085)

10:40–11:00 Break

11:00

4aPA5. A pipe-like one-way structure of acoustic wave. Bo Yuan (Key Laboratory of Modern Acoustics, MOE, and Institute of Acoustics, Department of Physics, Nanjing University, Nanjing 210093, China, xillo.yuan@gmail.com)

We proposed a pipe-like structure with linear materials to obtain the acoustic unidirectional transmission. The system consists of a bending pipe and a phononic crystal and designed by extending the idea of frequency selection in nonlinear acoustic diode into the mode selection of a linear acoustic system. The system has a significant transmission efficiency and good rectifying ratio which is designed to work in the air. We also experimentally realized the unidirectional transmission behavior for acoustic waves and the experimental results agree well with the theoretical simulation. This device is expected to have potential applications in ultrasonic devices such as acoustic diodes.

11:20

4aPA6. The surface acoustic waves in the phononic crystal surface based on the locally resonance mechanism. Yong Li (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, kyolee2010@gmail.com)

We proposed a two dimensional structure consisted of a resonant unit stubbed periodically on a phononic crystal plate with finite thickness to investigate the surface acoustic waves (SAWs). Numerical results shown that two types of SAWs, one is the classical Rayleigh SAWs, whereas the other is the scattering SAWs overcoming the limitation that SAWs can only exit below the bulk cone of the substrate, can be found in the structure. Furthermore, the band gaps of the SAWs are obtained, as well as slow modes of the SAWs are also observed in the band gaps of the phononic crystal substrate. It could open the probability to control effectively the propagation of these SAWs. The results should have the impact on the SAWs communications.

11:40

4aPA7. Utilizing negative diffraction effect in phononic crystal to realize resting X-shaped wave localization. Weiwei Kan and Jianchun Cheng (Department of Physics, Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, P.R. China, rdchkww@gmail.com)

Motivated by the promising impacts in acoustic devices and applications that can efficiently trap, guide, and manipulate sonic waves, researches of Photonic Crystals (PCs) have received great attention in the past decade. Many anomalous refractive and diffractive effects, such as superprism effects and negative refraction, have been found in such PCs. In this report, by studying the wave packet superposed by Bloch modes in the isofrequency surface at unique points of the PC band structure, the negative diffraction effect in certain directions of the PCs is exploited to realize X-shaped wave localization settled in one point, which cannot exist at rest in homogeneous media.

12:00

4aPA8. Ultrasonic cleaning of the root canal. Bram Verhaagen, Christos Boutsioukis (Physics of Fluids, University of Twente, P.O. Box 217, 7500AE Enschede, The Netherlands, b.verhaagen@tnw.utwente.nl), Lei-Meng Jiang, Ricardo Macedo (Academic Center for Dentistry, Gustav Mahlerlaan 3004, 1081LA Amsterdam, The Netherlands), Damien Walmsley (School of Dentistry, University of Birmingham, St Chad's Queensway, Birmingham B4 6NN, United Kingdom), Luc Van der Sluis (Paul Sabatier University, 118 route de Narbonne, 31062 Toulouse cedex 9, France), and Michel Versluis (Physics of Fluids, University of Twente, P.O. Box 217, 7500AE Enschede, The Netherlands)

A crucial step during a root canal treatment is the irrigation, where an antimicrobial fluid is injected into the root canal to eradicate all bacteria from the root canal system. Agitation of the fluid using a miniature file oscillating at 30 kHz has shown a significant improvement in the cleaning efficacy over conventional syringe irrigation. However, the exact cleaning mechanisms, being acoustic streaming, cavitation or an enhanced chemical effect, are not fully understood. Here we investigate ultrasonically activated irrigation through experiments and numerical simulations in order to understand the relative importance of each of the three cleaning mechanisms. We combine high-speed imaging and micro-Particle Imaging Velocimetry to visualize the flow pattern and cavitation in a root canal model (sub-millimeter dimensions), at timescales relevant to the cleaning processes (microseconds). Measurements of the acoustic streaming are coupled to the oscillation characteristics of the file as simulated numerically and measured with a laser vibrometer. Comparison between the streaming pattern inside the root canal and in the free field shows the importance of the confinement of the root canal on the acoustic streaming. The results give new insight into the role of acoustic streaming for the cleaning of root canals.

Session 4aPP

Psychological and Physiological Acoustics: Current Issues in Auditory Cortex Physiology

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Invited Papers

9:20

4aPP1. Cortical representations of pitch: theories and experiments. Xiaoqin Wang (Department of Biomedical Engineering, Johns Hopkins University, 720 Rutland Avenue, Traylor 410, Baltimore, MD 21205, xiaoqin.wang@jhu.edu)

Pitch perception is one of the most important auditory perceptual phenomena. Its underlying neural mechanisms have not been well understood. Recent human imaging studies and neurophysiology experiments in non-human primate have begun to reveal possible neural coding mechanisms in the cerebral cortex. These studies have pointed to a specialized area in the rostral region of primate auditory cortex where harmonic pitch is extracted. How pitch-selective neurons in this cortical area extract harmonic pitch at the cellular level, however, is yet known. Moreover, it remained to be explored whether other auditory cortical areas process aspects of pitch that are not processed by this rostral pitch-region. An important issue in the study of cortical representations of pitch is whether pitch embedded in harmonic complex sounds is extracted and uniquely represented by a specific cortical area or a subset of neurons in that area. Simply showing that pitch information exists in neural firing in a cortical area is not an adequate demonstration of pitch processing mechanisms. (Research supported by NIH grant R01-DC003180)

9:40

4aPP2. Acoustic motion processing in auditory cortex. Stephen Lomber (The Brain and Mind Institute, The University of Western Ontario, London, Ontario, Canada, steve.lomber@uwo.ca)

Within extrastriate visual cortex of humans, monkeys and cats, individual cortical areas are specialized for spatial or motion processing. In cat auditory cortex, four regions have been identified to be critical for accurately determining the spatial location of an acoustic stimulus. The purpose of the present investigation was to determine if there is an area in auditory cortex specialized for acoustic motion processing or if areas involved in spatial localization are also critical for acoustic motion processing. Or, is there an acoustic MT? To accomplish this, cats were trained to perform two tasks: a spatial localization task using a static stimulus and a task that required the animals to discriminate leftward from rightward apparent acoustic motion. Focal reversible cooling was used to bilaterally deactivate each of the thirteen areas of cat auditory cortex. Overall, the results show that areas involved in acoustic motion processing are also involved in static spatial localization. An area that is uniquely involved in acoustic motion processing was not identified. These results suggest that spatial localization functions may be a prerequisite for acoustic motion processing in auditory cortex. Supported by the Canadian Institutes of Health Research and the Natural Sciences and Engineering Research Council of Canada.

10:00

4aPP3. Rapid plasticity in auditory and prefrontal cortex during active listening. Shihab Shamma (A.V. Williams Bldg, University of Maryland, College Park, MD 20742, sas@umd.edu)

Humans and other animals often attend to sounds in their environment so as to approach mates and competitors, or to avoid predators. Numerous neural processes orchestrate the performance of these behavioral tasks, including sensory adaptive responses in the auditory cortex and executive control functions in the prefrontal cortex. The multitude of mechanisms observed to be involved in neurophysiological recordings from several auditory and prefrontal cortical fields in behaving ferrets will be reviewed. The findings reveal that rapid changes in auditory receptive fields take place only during task performance, and that these serve to enhance discrimination and detection of target stimuli from their backgrounds. Interestingly, the changes also depend on the meaning of the sounds (aversive or appetitive) and the level of behavioral performance.

10:20

4aPP4. Micro-organization and plasticity of the primary auditory cortex. Patrick Kanold (University of Maryland, Dept. of Biology, 1116 Biosciences Bldg, College Park MD 20742, pkanold@umd.edu)

The auditory cortex is a laminated structure that adaptively processes sensory information from the external environment. The precise nature of the transformation of sensory information at the level of cortical networks is unknown. We use in vivo two-photon calcium imaging techniques to measure response properties and functional organization of primary auditory cortex (A1) neurons in mouse. We find that frequency selectivity in supragranular layers is heterogeneous on small spatial scales and that this heterogeneity is likely created from sampling of diverse inputs to supragranular neurons. The large frequency range of inputs available to each neuron might provide a

substrate for a large degree of plasticity in individual neurons. We tested the capacity of A1 neurons to rapidly change tuning properties by using micro-stimulation of top-down projections to A1 and pairing such stimulation with a particular sound. We find that the frequency tuning of individual neurons can rapidly be changed leading to an increase in the representation of the paired sound. Collectively, these results provide insight into how sensory information is represented and adaptively transformed in auditory cortex.

10:40–11:00 Break

11:00

4aPP5. Auditory cortical plasticity—from synapse to perception. Christoph Schreiner (UCSF, 513 Parnassus Ave., San Francisco, CA 94044, chris@phy.ucsf.edu), and Robert Froemke (NYU, 540 First Ave., New York, NY 10016)

Synapses and neuronal receptive fields of the cerebral cortex are plastic. Enhancements and decreases to auditory cortical excitatory synapses can be induced by pairing acoustic stimuli with activation of the nucleus basalis neuromodulatory system. Similarly, the feature selectivity of individual neurons and cell assemblies can be modified in a manner that depends on the patterns of network activity, the engagement of neuromodulatory systems, as well as by sensory experience. Perceptual performance has been shown to change as a consequence of experience and learning. The relationship between synaptic and receptive field changes and perceptual plasticity, however, is poorly understood. We used in vivo whole–cell recordings and behavioral testing to explore that relationship in more detail. We will discuss how synaptic modifications and receptive field changes are reflected in the encoding of frequency and intensity information in rat auditory cortex and demonstrate that these changes are expressed in the perceptual behavior of animals in sensory detection and classification tasks. It is concluded that direct modification of specific cortical inputs leads to wide-scale synaptic changes, which collectively support improved sensory perception and enhanced behavioral performance. Supported by NIH Grants DC02260 (to CES) and DC009635 (to RCF)

11:20

4aPP6. State-dependent changes in background discharge of auditory core neurons in freely moving guinea pigs. Hisayuki Ojima and Masato Taira (Tokyo Medical and Dental University, 113-8549, Japan, yojima.cnb@tmd.ac.jp)

In natural situations, sounds, such as predator noises and nursing and mating calls, are used as signals in determining an animal's behavior. Shifts in behavior are adaptively controlled by activation of higher-order brain regions. Utilizing forced change in behavior as a trigger, we show that brain state shifts affect the background discharge pattern of neurons in primary auditory cortex (A1). Single unit recordings were made from freely moving guinea pigs, which were impelled to shift from a passive/stationary state to an active/exploratory state by a sudden change in ambient illumination from light to dark. Upon the illumination shift, background discharge was reduced significantly for several minutes. This behavioral shift was induced even when the animals were actively engaged in eating food. Acoustic stimulation during the time of reduced background discharge resulted in an improved S/N ratio of the neuron's response. These neurons were localized almost exclusively in upper layer 5, from which cortical feedback to subcortical stations originate. In naturalistic situations, a subset of A1 neurons may transfer information about the cortical state to subcortical auditory stations. Supported by KAKENHI to H.O. (No. 22500368).

Contributed Papers

11:40

4aPP7. Dynamic binaural-correlation processing in rats' inferior colliculus, medial geniculate body, and primary auditory cortex. Qian Wang (Department of Psychology, Department of Machine Intelligence, Speech and Hearing Research Center, Key Laboratory on Machine Perception (Ministry of Education), Peking University, Beijing, China, aleinwangba@126.com), Shuyang Cao, Jingyu Li, Xihong Wu (Speech and Hearing Research Center, Key Laboratory on Machine Perception (Ministry of Education), Peking University, Beijing, China), and Liang Li (Department of Psychology, Department of Machine Intelligence, Speech and Hearing Research Center, Key Laboratory on Machine Perception (Ministry of Education), Peking University, Beijing, China)

Interaural correlation processing is critical for grouping and segregating auditory streams under reverberant environments with multiple sources. Although detection of dynamic changes in interaural correlation has been extensively studied in the field of psychoacoustics, the underlying neural mechanism remains largely unknown. In this study, frequency-following responses (FFRs) to narrow-band noises were measured at various levels of the auditory system in rats, including the inferior colliculus (IC), ventral division of the medial geniculate body (MGB), and primary auditory cortex (A1). The results of Experiment 1 show that FFRs recorded in the IC were affected by both interaural correlation and the interaural time difference (ITD). Moreover, results of Experiment 2 show that a break in interaural correlation (BIC) could elicit marked FFRs in each of the three central auditory structures, and the BIC-induced FFRs were significantly affected by the ITD. The results suggest that the rat's central auditory system is able to resolve and compare fast changes in fine-structure details of arbitrary noises presented at the two ears.

12:00

4aPP8. Associative visuoauditory memory in the auditory cortex. Jufang He, Xi Chen, Yiping Guo, Zhengli Liao, Xinjian Li, Haitao Wang, and Xiao Li (Laboratory of Applied Neuroscience, Department of Rehabilitation Sciences, The Hong Kong Polytechnic University, Hung Hom, Kowloon, Hong Kong, rsjufang@polyu.edu.hk)

This paper presents direct evidences of the establishment of associative memory traces in the rat auditory cortex and the participation of the entorhinal cortex in the establishment and retrieval of these memory traces. We produced an association between cortical electrical activation and a visual stimulus with classical conditioning. The memory traces were physiologically visualized from auditory neuronal responses to the visual stimulus after conditioning and behaviorally confirmed with a memory recall experiment. Formation of new associative memory in the auditory cortex with classical conditioning was bilaterally blocked when the entorhinal cortex was unilaterally temporarily inactivated, but returned if the entorhinal cortex was not inactivated. Retrieval of the established associative memory in the ipsilateral neocortex was affected by the inactivation of the unilateral entorhinal cortex, while that in the contralateral neocortex was not affected, thus suggesting a less dependence of the hippocampal system in the retrieval than in the formation of associative memory. Supported by Hong Kong Research Grant Council (PolyU 9/CRF/09)

Session 4aSP

Signal Processing in Acoustics, Acoustical Oceanography, and Underwater Acoustics: Model-Based Processing and Analysis III (Poster Session)

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Contributed Papers

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

4aSP1. An analysis of influencing factors for structural damage imaging using Lamb waves. Haiyan Zhang (149 Yanchang Road, School of Communication and Information Engineering, Shanghai University, Shanghai 200072, hyzh@shu.edu.cn)

The use of Lamb waves for the inspection of large plate-like structures and their structural health monitoring (SHM) has been a topic of considerable interest in the development of advanced quantitative nondestructive evaluation techniques. Current interrogation algorithm using Lamb waves can roughly identify the location and severity of potential damages. It is well documented that Lamb waves may have multiple modes at a certain frequency, and accompanied by possible dispersion for each specific mode at different frequencies. In order to improve the accuracy of damage identification, some factors such as frequency, mode and cycle number of excitation signal must be taken into account. This paper presents a probability-based imaging approach for evaluating through-thickness hole damage in an aluminium plate. This method predicts the damage in terms of the probability of its occurrence at a certain spatial position of the structure. The influences of various factors on imaging results are analyzed. The results demonstrate that the imaging algorithm can be used to identify the location and severity of damage, but the identifying accuracy is highly related to frequency, mode and cycle number of exciting Lamb wave signal.

4aSP2. Reduction of computation time using Graphics Processing Unit for the detection of a crack in a large scale concrete structure. Yuhei Katsurakawa, Toyota Fujioka, Yoshifumi Nagata, and Masato Abe (Iwate University 020-8551, h19j12@cis.iwate-u.ac.jp)

This paper describe a method for estimating a crack position in a concrete structure using several accelerometers. An array of accelerometers is installed to the concrete structure and a low frequency vibration is made with a small impulse hammer. A reflection wave is generated from the crack position if a crack exists. Because the concrete structure is elastic, it has three wave-propagation modes. It is difficult to estimate the position precisely because the power of necessary primary-wave mode is weaker than that of surface-wave mode. To estimate the crack position precisely, we have proposed a method for eliminating the unwanted surface-wave and side-wall reflections, in which five parameters are used to estimate an unwanted surface-wave or a side-wall reflection by least mean square technique. Since it takes, however, a very long time to estimate a single unwanted wave, the method did not work if two waves overlap with each other. Therefore, we propose the parallelization of genetic algorithm on GPU using CUDA. As a result, the processing time was shortened dramatically compared to conventional one, and we could distinguish two waves reflected from two close boundaries of a caisson, a huge concrete structure which is used as a breakwater.

4aSP3. Nearly Perfectly Matched Layer (NPML) absorbing boundary condition for elastic waves propagation in solid. YiFeng Li (No. 30, Puzhu Road(S), Nanjing 211816, China, 79 Box number, lyffz4637@163.com), Olivier Bou Matar, and YaPing Bao

In this work, a method named Nearly Perfectly Matched Layer (NPML) using a Complex Frequency Shift (CFS) stretched-coordinate metrics is presented to extend the Perfectly Matched Layer (PML) to simulate elastic wave propagation in solid media. This non-physical layer is used at the computational edge of a Discontinuous Galerkin Finite Element Method (DG-FEM) algorithm and a Pseudo-Spectral (PS) algorithm in time domain, as an Absorbing Boundary Condition (ABC) to truncate unbounded media. The main advantages of NPML is linked to the facts that (a) the obtained system of equations has the same form exactly as the original system of equations and so strongly hyperbolic, and (b) the introduced NPML variables are updated by Ordinary Differential Equations (ODE) in place of Partial Differential Equations (PDE) in classical PML implementation. Numerical results show that the NPML has the same ability of energy absorption as the Convolutional Perfectly Matched Layer (CPML) for attenuating the outgoing waves, moreover, it facilitates implementation in the DG-FEM scheme than CPML and preserves the highly parallelisable capabilities of this numerical scheme.

4aSP4. The study of time-frequency analysis the nocturnal snoring signal based on the wavelet transform. Zhang Yinhong (College of Physics and Information Technology, Shaanxi Normal University, Xi'an, Shaanxi 710062, China, zhangyh@snnu.edu.cn), Li Quanlu (Applied Acoustics Institute, Shaanxi Normal University, Xi'an, Shaanxi 710062, China), and Wu Jing (College of Physics and Information Technology, Shaanxi Normal University, Xi'an, Shaanxi 710062, China)

This paper presents a time-frequency analysis method for non-stationary snoring signal that based on the discrete wavelet transform. The snoring is an important characters of upper airway obstruction and a typical inspiratory sound appearing during sleep. The severe snoring sound leads to the Obstructive Sleep Apnea Syndrome during persistent ventilatory movements and it can result in cessation of breathing. The resulting experimental shows that the characters of different time-frequency domain of the snoring signal and offers value for analyzing the temporal feature of snoring sound in health medical treatment. It is an interesting application of the wavelet transform theory in the medical field.

4aSP5. The whisper sensitive scale in the application of speaker identification. Wei Lin (Nanjing University of Aeronautics and Astronautics, 211100, wlin@nuaa.edu.cn)

In this paper, the frequency characteristics of numerical whispered speech were investigated by a filter bank analysis. It was shown that the first and the third formants were more important than the other formants in the speaker identification of Chinese whispered speech. The experiment showed that the 800-1200 Hz and 2800-3200 Hz ranges were the most significant frequency ranges in discriminating the speaker. Based on this result£, a new feature scale named whisper sensitive scale (WSS) was proposed to replace the common scale, the Me1 scale£, and to extract the cepstral coefficient from whispered speech signal. Furthermore, a speaker identification system in whispered speech was presented based on WSCC (the whisper sensitive cepstral coefficient. And the new system performed better in solving the problem of speaker identification of numerical whispered speech than the traditional method.

4aSP6. A modified weighted overlap and add-based spectral subtraction method. Yang Sun, Meng Yuan, and Haihong Feng (Shanghai Acoustics Laboratory, Institute of Acoustics, Chinese Academy of Sciences, No. 456, Xiaomuqiao Road, Xuhui District, Shanghai, China, sunyang0109@mails.gucas.ac.cn)

The Weighted Overlap and Add (WOLA) method is an optimization of frequency modulated DFT filter banks. However, the decrement in frequency resolution of speech signal caused by WOLA leads to noise estimation error. In this study, a speech power compression and gain compensation method was proposed to solve the low-resolution problem from the WOLA process. Finally, a multi-band spectral subtraction method was developed by combining the WOLA and the spectral subtraction algorithm in frequency domain. The noise reduction degrees were determined by the individual Signal-to-Noise Ratios (SNRs) in each frequency bands. Objective evaluation of the proposed noise reduction method was performed based on Itakura-Saito Distortion Measure under different types of noise. Statistical results showed that this proposed method could improve noisy speech by 5 dB SNR with little distortion. This noise reduction method has an advantage on low computation load, real-time processing and good performance on noise reduction. So this method can be implemented in embedded devices, e.g. hearing aid and cochlear implants. # This work is supported by National Natural Science Foundation of China (11104316). and Shanghai Natural Science Foundation (11ZR1446000).

4aSP7. Improvement on coherent signal-subspace method using several reference frequencies. Dahang Feng (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, Beisihuan West Road, Haidian District, Beijing 100190, fengdh@mail.ioa.ac.cn), Ming Bao, Luyang Guan, Jianfei Tong, and Xiaodong Li

An improved coherent signal-subspace method is proposed using several reference frequencies for wideband direction-of-arrival (DOA) estimation. In this proposed method, the whole bandwidth of the received signal is divided into several parts, then a reference frequency is selected from each part for the coherent signal-subspace method to estimate the DOAs of wideband sources separately, and the results from all frequency parts are averaged to obtain a final estimate. Compared with the conventional coherent signal-subspace method, the proposed method achieves higher resolution and smaller root mean square error, especially when the bandwidth of the source is large. The performance of the proposed method is demonstrated and analyzed through the computer simulations.

4aSP8. The target tracking based on cubature Kalman filter. Yuanyuan Fang (Northwestern Polytechnical University School of Marine Engineering, Room 728, emma6663@hotmail.com)

A new extension of Kalman Filter to nonlinear system– Cubature Kalman Filter is introduced.This algorithm has its theory basis consisted of Gaussian Bayesian theory and Spherical-radial rules.In the light of the unique properties of Cubature Kalman Filter:a derivative-free on-line sequential-state estimator, computational complexity grows as n^3 and it eases the curse-of-dimensionality problem.The paper focuses on studying the CKF algorithm in depth, and applys it to bearing-only maneuvering target tracking.Two typical tracking model about tracking maneuvering ship from both non-maneuvering and maneuvering platform is selected.The Monte-Carlo simulations' results illustrate that the CKF algorithm outperforms EKF in accuracy and calculation efficiency of filter, and which make Cubature Kalman Filter easy and feasible to broad application prospect.

4aSP9. Combining Capon and Bartlett spectral estimators for detection of multiple sinusoids in colored noise environments. Chengshi Zheng 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, cszheng@mail.ioa.ac.cn)

Detection of multiple sinusoids in colored noise environments has many potential applications, such as sonar, radar, and communication. Most of conventional algorithms often use the local signal-to-noise-ratio (LSNR) as a test statistic to detect the sinusoids, where the LSNR is estimated in the frequency domain by using the Bartlett spectral estimator (BSE). Unfortunately, the BSE has a relatively low frequency resolution, which may degrade the detection performance significantly. To solve the frequency resolution problem of the BSE, this paper proposes a two-stage hybrid algorithm to estimate the LSNR. In the first stage, the BSE is used to estimate the noise power spectral density over frequency. After obtaining the noise power spectral density, the second stage employs the Capon spectral estimator (CSE) to estimate the LSNR. The proposed hybrid algorithm significantly improves the detection performance, especially when the sinusoids are closely spaced. Simulation results show that the proposed algorithm performs much better than the conventional algorithms in most cases.

4aSP10. Restoring clipped speech signal based on spectral transformation of each frequency band. Makoto Hayakawa, Masanori Morise, and Takanobu Nishiura (Ritsumeikan University, 1-1-1 Nojihigasi, Kusatsu, Shiga, Japan, is033080@ed.ritsumei.ac.jp)

In recent years, high-quality speech recording is requested for comfortably using a communication system. However, a clipping distortion caused by input exceeding the maximum range of amplifier is one of the problems with a sound quality degradation. Although linear prediction method has been conventionally proposed for restoring a clipped speech signal, it has a problem that the frequently clipped speech signal degrades the restoration performance by increasing the prediction errors. In this paper, we propose a method for restoring a clipped speech signal based on a spectral transformation of each frequency band. In this method, the spectral envelope of target speech signal in each frame is approximated to the spectral envelope of original speech signal to remove the influence of a clipping distortion. In particular, the spectral envelope in higher frequency domain including a static characteristic of the speaker is replaced with the spectral envelope of the unclipped speech signal prepared in advance. Then, the spectral envelope in lower frequency domain including a characteristic of phoneme is approximated with Gaussian Mixture Models. We carried out an evaluation experiment for sound quality of speech signal processed by the proposed method. As a result, we confirmed the effectiveness of the proposed method.

4aSP11. A study on the technique using fractal dimension in the selection of the kind of sound. Kenji Muto, Hideo Shibayama, Yoshiaki Makabe (Shibaura Institute of Technology, Dept. of Communications Engineering, 3-7-5 Toyosu, Koto-ku, Tokyo, Japan, k-muto@sic.shibaura-it.ac.jp), Kikuo Asai, and Kimio Kondo (Center of ICT and Distance Education)

There is a paper which described that the selection of the sound which has a different fractal dimension is possible by aural. The fractal dimension of each short time window of conversation sound was analyzed, referring to the paper. In this paper, we showed about the technique using fractal dimension in the selection of the kind of sound. The analyzed sound was a conversation sound with which the engine sound and the chirp sound mix. The value of fractal dimension in the case of the chirp sound of bird or the engine sound indicated a value different from the fractal dimension in the case of the conversation sound. We thought that our technique has shown the characteristic of sound source by one parameter. It is possible to use the fractal dimension to judge a mixing surrounding sound in the teleconference at the direction of the speaker where the voice was used. To transmit a clear voice in the teleconference, the voice is utilized to the estimation in the direction of the speaker. In the real system, the case that the voice mixes with surrounding sound, it is distinguishable by fractal dimension.

4aSP12. Reproduction of human-phonatory radiation characteristics with a polyhedron loudspeaker. Naoki Yoshimoto, Masanori Morise, and Takanobu Nishiura (Ritsumeikan University, 1-1-1 Nojihigashi, Kusatsu, Shiga, Japan, is046081@ed.ritsumei.ac.jp)

Spoken dialogue systems have been studied for car navigation systems and voice search systems. For evaluation, a loudspeaker is used instead of a human because these systems require various kinds of speech samples. However, the sounds radiated by loudspeaker can not reproduce human-phonatory radiation characteristics. Therefore, the mouth simulator is utilized to reproduce human-phonatory radiation characteristics. Although it is based on the average mouth shape, shapes of mouth are different among phonemes. Therefore, due to the hardware structure, it can not accurately reproduce various human-phonatory radiation characteristics affected by shapes of mouth. In this study, we developed a polyhedron loudspeaker to solve this problem. It consists of eleven loudspeakers which are independently controlled. Controlling eleven loudspeakers makes it possible to reproduce desired radiation characteristics. Besides, we try to reproduce human-phonatory radiation characteristics of each Japanese five vowel with an adaptive algorithm based on the MINT (Multi-input/output INverse Theorem). We carried out an experiment to verify the effectiveness of the proposed method. As a result, it was confirmed that phonatory radiation characteristics of Japanese five vowels could be accurately approximated compared with the mouth simulator.

4aSP13. A spatial domain processing method for the direct signal separation in the reverberant field. Benyu Wu (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, wubenyu@mail.ioa.ac.cn)

A spatial domain processing method for the direct signal separation from the same frequency reflections at different frequencies in the reverberant field is proposed. The reflections with the same frequency contained in the reverberant acoustic signals are distributed uniformly and irregularly in space. With the same direct signal being kept, the method makes the reverberant acoustic signals in the different spatial positions synchronized and superposed. Simultaneously the same frequency reflections are eliminated, and then the direct signal is separated. It overcomes the difficulty that the direct signal can't be obtained because of the effect of reflections. It is important and significant for the acoustic measurement or the other researches based on the direct signal. The results show that the method is verified feasible and effective, by the experiments at different frequencies.

4aSP14. Aircraft flight parameter estimation via multipath delays using ground-based microphone array. Wei Xie, Luyang Guan, Ming Bao, 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, xiewei@mail.ioa.ac.cn)

The acoustic signal from a low flying aircraft received by ground-based microphone array is characterized by the interference fringe pattern which is caused by reflection of the ground. In this paper, a model is developed to estimate the parameters of an aircraft's motion with the assumption that the aircraft flies in a straight line at a constant height. The model estimate the speed and height of the aircraft based on the different time of arrival due to the multipath delay at different locations. And the short-time cepstrum method is adopted to estimate the multipath delay accurately. To evaluate the performance of the model, the experimental results and error analysis are presented.

4aSP15. Digital communication system using beamsteering for difference frequency in a parametric array. Chong Hyun Lee, Jaeil Lee, Jinho Bae, Dong-Guk Paeng (Jeju National University, 690-756, chonglee@jejunu.ac.kr), Seung Wook Lee, Jungchae Shin, and Jin Woo Jung (Hanwha Corporation, 730-904)

Digital acoustic signal processing can be applied to sonar and acoustic communications. Especially, transmitting acoustic signal to the desired direction has many applications in military and industry fields. In this paper, we present a steerable digital communication system using parametric array transducers. To evaluate the proposed system, we build digital communication system by using transducer array, power amplifier and Labview software. The Labview software is composed of two parts. The first part is designed to generate beam to the desired direction by changing parameters such as number of sensors, complex weight of each sensor, type of transmit data and etc. The second part is to generate modulated signal of ASK, FSK and PSK, and to demodulate the received signals. With laboratory experiments, we verify the performance of the proposed communication system. Experimental results show that the system can be used to mitigate multipath effect in shallow water and can achieve high data-rate transmission

4aSP16. Interoperability of heterogeneous cores on acoustic signal processing and communication system. Shengchen Cao, Zhaoli Yan, Tianhao Cui, Xiaobin Cheng, 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, caoshengchen@mail.ioa.ac.cn)

An acoustic signal processing and communication system is typically constituted with FPGA, DSP and application processors. These heterogeneous cores have different speed, throughput and peripheral interfaces, which results in barriers of data sharing among them. Texas Instruments' open-source component DSPLINK is an applicable solution for ARM and DSP communication. In this paper, DSPLINK architecture is optimized, FPGA is designed to act as part of the memory pool and SPI based synchronization protocol between the three cores is implemented. As an example of application, a system for cavitations monitoring based on OMAP and FPGA platform is introduced subsequently. Linux OS and DSP/BIOS are ported to ARM end and DSP end respectively, based on which the interoperable architecture is implemented. The developers can use interoperable APIs for data transfer so that more applications can be easily derived.

4aSP17. The realization of precision time protocol for distributed acoustic and vibration measurements. Longhua Ma (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, malonghua@mail.ioa.ac.cn)

Distributed acoustic and vibration measurement based on Local Area Network (LAN) becomes a hot topic, recently. The performance of this measurement system is affected dramatically by the synchronization precision of different measurement nodes. Traditional synchronous methods, such as Network Time Protocol (NTP), Simple Network Time Protocol (SNTP) doesn't meet the precision needed for distributed acoustic and vibration measurements, because it can only achieves accuracy of microseconds. In addition, sync cable is not suitable for long distance distributed acoustic and vibration measurement due to the inconsistent delay. To sovle this problem, a method based on Precision Time Protocol (PTP, IEEE1588) is proposed in this paper for synchronization of distributed acoustic and vibration measurement device. A Field Programmable Gate Array (FPGA) is employed between Medium Independent Interface (MII) and PHY, which monitors all ingress and egress data packets. Only PTP packets are unpacked and proceeded. According to these packets, the time offsets are calculated. Then time offsets between master clock and slave clock are filtered, and the output of the filter is used to compensate the drift of crystal oscillator. Synchronization experiment show the proposed method can achieve synchronization accuracy of few hundred nanoseconds.

4aSP18. A distributed array processing using multi-channel signals over a network with an embedded time code by the network time protocol. Yoshifumi Chisaki, Tomohisa Mashima, and Tsuyoshi Usagawa (Kumamoto University, chisaki@cs.kumamoto-u.ac.jp)

A conventional microphone array system uses a conductor to wire from a microphone to an input via an amplifier. While, a wireless transmission for an array system makes the configuration flexible, and it is expected to provide novel applications widely, such as measuring of impulse response in wide area. One of the issues is a synchronization of time between channels. This paper proposes multiple signals transmission system over a network with a time code embedding to synchronize those signals. The system

consists of clients and a server. Each client at a receiving point runs a network time protocol client daemon, and a packetized audio signal with a time code is sent to a server. The signal from a client is reconstructed at a server based on the time code. Since the time differences between clients affects to performance of the multichannel signal processing, small error in time at a client is preferred. This paper discusses how the error in time between channels affects to performance of the distributed microphone array system.

4aSP19. Loudspeaker compensation using mixed phase technology. Chao Ye, Ming Wu, Shuaibing Wu, and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, cye@mail.ioa.ac.cn)

In a sound reproduction system loudspeakers always introduce distortions. The compensation of loudspeaker responses using digital signal technology is becoming an important part of improving sound reproduction quality. Several FIR and IIR filter design methods have been proposed to equalize the response of loudspeaker systems. However, high order filters are needed to obtain excellent resolution at low frequencies. From a psychoacoustic point of view, warped filters have been employed to improve the resolution at low frequencies, but at the expense of poor resolution at high frequencies. On the other hand, the use of warped filters increases the complexity of the filter structure. In this paper, a mixed phase technology is proposed to equalize the response of loudspeaker systems. Loudspeaker response can be decomposed into minimum and excess phase components, which are inverted respectively to construct the equalization filter. An all pass filter based on position-independent excess phase zeros is cascaded with a minimum phase filter to improve the phase response in the region. The experimental results demonstrate that the time-domain response of the loudspeaker systems is improved by using the presented mixed phase technology.

4aSP20. Estimation of demodulation ratio for the parametric loudspeaker based on spectral envelope. Daisuke Ikefuji, Masanori Morise, and Takanobu Nishiura (Ritsumeikan University, Nojihigashi 1-1-1, Kusatsu, Shiga, Japan, cm000074@ed.ritsumei.ac.jp)

A parametric loudspeaker with the higher directivity transmits the sound wave to only particular area. It emits an AM (Amplitude-Modulated) wave demodulating into an audible sound wave by nonlinear interaction in the air. Therefore, longer distance is required for fully demodulating the AM wave into the audible sound wave. On the other hand, a power of the audible sound wave decays depending on the distance. Thus, the parametric loudspeaker must be utilized on the suitable distance for reproducing the fully demodulated wave with an enough power. However, no criterion has been conventionally proposed to measure the demodulation ratio in the past. Accordingly, the criterion should be formulated to measure demodulation ratio for appropriately utilizing the parametric loudspeaker. Thus in this paper, we aim at formulating it for estimation of demodulation ratio. We therefore propose the criterion based on a spectral envelope of the reproduced TSP (Time-Stretched-Pulse) with parametric loudspeaker. The proposed criterion is defined based on the correlations between the spectral envelope in every distance and that in the target with the reproduced TSP. As a result of the objective experiment, we confirmed the availability of the proposed criterion.

4aSP21. Harmonic distortion measurement for a parametric loudspeaker with logarithmic time stretched pulse. Shohei Masunaga, Daisuke Ikefuji, Masanori Morise, and Takanobu Nishiura (Ritsumeikan University, 1-1-1 Nojihigashi, Kusatsu, Shiga 525-8577, Japan, is037089@ed.ritsumei.ac.jp)

A parametric loudspeaker which utilizes the ultrasound can transmit the audible sound to a particular area. However, the sound reproduced by a parametric loudspeaker contains the harmonic distortions because the sound is demodulated by the nonlinearity in the air. Thus, measuring the harmonic distortions is required to evaluate the sound quality of a parametric loudspeaker. Sinusoidal wave method has been used as the harmonic distortion measurement. In it, the harmonic distortion is measured by analyzing the integral multiplication frequency of a reproduced sine wave. Many measurements by using sine waves with each different frequency are required to measure the wideband harmonic distortions. Therefore, measuring the wideband harmonic distortions with sinusoidal wave method requires much more time. Recently, using Log-TSP (logarithmic time stretched pulse) signal was proposed to measure the wideband harmonic distortions in a short time for non-parametric loudspeakers. Thus in this paper, we attempt to measure harmonic distortion of a parametric loudspeaker by using Log-TSP signal. We carried out an objective evaluation experiment in a soundproof room. The result by using Log-TSP signal was compared with that by using sinusoidal wave method. As a result, we confirmed the result with Log-TSP is equivalent to that with sinusoidal wave method.

4aSP22. The realistic- reverberation sensation in 3-D acoustic sound field reproduction with parametric loudspeakers and indirect non-parametric loudspeakers. Hideya Tsujii, Masanori Morise, and Takanobu Nishiura (Ritsumeikan University, 1-1-1, Nojihigashi, Kusatsu, Shiga, Japan, cm007077@ed.ritsumei.ac.jp)

In the field of virtual reality, mixed reality (MR) has recently been drawing attention as a technology to experience the virtual space by superimposing the computer graphics (CG) objects in real space. The technologies to reproduce 3-Dimensional (3-D) acoustic sound fields in conjunction with MR can especially experience a higher realistic sensation than the conventional MR. A parametric loudspeaker based on an ultrasound with the higher directivity can reproduce 3-D acoustic sound fields by designing a sound image. However, it is difficult to present the realistic sensation depending on the reverberation, because an acoustic wave emitted by it shouldn't diffuse in the room. In this paper, we newly proposed to reproduce 3-D acoustic sound fields utilizing multiple parametric loudspeakers and indirect non-parametric surround loudspeakers for realizing the realistic- reverberation sensation. It could realize a higher realistic- reverberation sensation without affecting the sound image localization. The effectiveness of the proposed method was assessed by the subjective evaluation experiments for the realistic- reverberation sensation and the sound image localization. As a result, we confirmed the effectiveness of the proposed method.

4aSP23. Distant-talking speech enhancement based on spectrum restoring with phoneme labels. Naoto Kakino, Takahiro Hukumori, Masanori Morise, and Takanobu Nishiura (Ritsumeikan University 1-1-1, Nojihigashi, Kusathu, Shiga, Japan, is012085@ed.ritsumei.ac.jp)

With the development of communication and speech recognition technology, remote conference and robot communication systems have been developed in recent years. Although these are valid in close-talking speech, distant-talking speech degrades the performance of the systems. The reason is that it is affected by energy decay, noise and reverberations depending on the distances. To solve this problem, many noise reduction and speech enhancement approaches have been proposed. In the conventional approaches, the example-based speech enhancement is one of the effective noise reduction methods in distant-talking conditions. The most similar example of noisy speech to input signal is detected from model examples of clean and noisy speech signals. Thereby it can estimate and suppress the noise based on the clean speech. Although it can effectively reduce noise, it is not clear the performance in the distant-talking speech condition with the decay of speech energy. Thus in this paper, we proposed a distant-talking speech enhancement based on spectrum restoring with phoneme labels. It restores the clean spectrum from decayed spectrum based on spectrum envelopes for each phoneme label. We demonstrate the experiment to evaluate the effectiveness of proposed method. As a result, we confirmed that it can effectively enhance the target speech.

4aSP24. The determination of dynamic subtraction for spectral subtraction towards musical tone reduction. Keisuke Horii, Takahiro Fukumori, Masanori Morise, Takanobu Nishiura, and Yoichi Yamashita (Ritsumeikan University, 1-1-1, Nojihigashi, Kusatsu, Shiga, Japan, cm010077@ed.ritsumei.ac.jp)

We should reduce the unwanted noise from noisy speech. Spectral Subtraction (SS) which is one of the noise reduction methods has been proposed by S. F. Boll in 1979, and it can effectively reduce the unwanted noise by utilizing only the observed signal. SS however has a problem that dissonant noise called musical tone is generated after noise reduction. SS estimates the noise with non-speech part and subtracts the estimated noise from observed signal. Flooring process is also performed depend on estimated noise power for supporting the excessive subtraction. Since the musical tone is generated by it, the improvement of SS is required to reduce it. In the past, we proposed SS with the weighted subtraction coefficients in each frequency band for controlling flooring process. In this method, the equal-weighted subtraction coefficients were utilized in every frame, however speech and noise power are different in each frame. To overcome this problem, we newly propose an advanced SS with the color-weighted subtraction coefficients in each frame for effectively reducing the musical tone. Both objective and subjective experiments were carried out for verifying the effectiveness of the proposed SS. As a result, the proposed SS could subjectively reduce the musical tone.

4aSP25. Realtime face recognition system with ultrasonic sensing. Benxi Cao (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, xixiwelcome@gmail.com), Jingyao Wang, Yang Wang, Yong Xu, and Jun Yang

Unlike vision-based approaches, ultrasonic sensing systems have the ability to obtain the object distance and echo energy information. With the development of airborne ultrasonic detection, ultrasonic face recognition has been discussed. However, the existing ultrasonic face recognition systems store the echo waveform data and make processing and analysis afterwards. Seperated acquisition and analysis procedures make these systems non-realtime. In this paper, a realtime ultrasonic face recognition system is proposed. The system has the following functions: signal generation, transmitting, receiving, amplification, demodulation, spectral analysis, and feature extraction. A continuous wideband ultrasonic signal is transmitted, then the geometrical information can be extracted from the echo signal, and suitable pattern recognition methods are used to recognize human face. The system can implement realtime face recognition with ultrasonic sensing, and a considerable recognition rate is achieved.

4aSP26. Robustness of the ultrasonic face recognition method to age variation: analysis and experimentation. Jingyao Wang (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, xuancao2009@hotmail.com), Benxi Cao, Yang Wang, Yong Xu, and Jun Yang

Ultrasonic human face recognition systems transmit ultrasonic Continuously Transmitted Frequency Modulated (CTFM) signal, extract facial geometry characteristic information from the received echo signals, and process the information for face recognition. Compared to vision-based approaches, ultrasonic human face recognition systems reduce the influence of illumination and avoid privacy leak. The existing approaches showed the feasibility of ultrasonic face recognition and achieved acceptable recognition accuracy. However, the aging problem, which equals to performance degradation caused by facial geometry change with time going on, is not considered in the previous researches. In this paper, robustness of the ultrasonic face recognition method to age variation is analysed. A database is built, in which each subject's face information was acquired at intervals of months during the last two years. Based on the database, various recognition experiments are conducted using the different pattern recognition algorithms. Aiming at the experimental results, the analysis to age factor that influences face recognition performance is implemented. The feature extraction methods and the pattern recognition algorithms are developed to increase the recognition rate.

4aSP27. Multi-stage identification for abnormal/warning sounds with onomatopoeia models. Junpei Ogawa, Kohei Hayashida, Masanori Morise, Takanobu Nishiura, and Yoichi Yamashita (Ritsumeikan University, 1-1-1 Nojihigashi, Kusatsu, Shiga, Japan, cm002074@ed.ritsumei.ac.jp)

In recent years, the methods utilizing environmental sounds have been increasingly employed for monitoring the safety of the elder who lives in distant place. Environmental sounds should consist of various sounds in daily life, and identified ones enables to detect abnormity. To detect abnormity, it is therefore required that abnormal/warning sounds are accurately identified among environmental sounds. In the past, environmental sound identification method has utilized acoustic models constructed by each sound source for all environmental sounds. However, as it only stores a few training abnormal/warning sounds, it is difficult to accurately design each abnormal/warning acoustic model. To overcome this problem, we propose a multi-stage identification method for abnormal/warning sounds. In the first stage, it utilizes an abnormal/alarm acoustic sound model and each normal acoustic sound model for detecting the abnormal/alarm sounds with a few training abnormal/warning sounds. In the second stage, it utilizes some acoustic models specializing in onomatopoeia to accurately identify abnormal and alarm sounds. We conducted an evaluation experiment to confirm the effectiveness of the proposed method on identification accuracy of abnormal/warning sounds. As a result, we confirmed that the proposed method was superior to the conventional method in identification accuracy.

4aSP28. A general audience rating surveying system and channel retrieval algorithm based on TV set audio features. Jingru Huang, Xin Ma, Lan Tian, and Shibin Du (School of Information Science and Engineering, Shandong University, Shanda South Road #27, Jinan 250100, hjingru2007@126.com)

A general surveying system for audience rating based on audio information are introduced. In this system, the audio signal is sampled from the audio outlet of TV set and high compressed into a special digital package which includes watching timing, audio signal features, and the marks of the TV channel switching. The packaged audio features are high robustness for different types of television sets and different audio volume and have no disturbance of surroundings. In the channel retrieval algorithm, multi-thresholds are used for calculating the correlation coefficients of audio features and matching the audio pattern, and the second searching is adopted for combining match when the retrieved TV channel of the head package is different from the one of the end package. The simulation test results show that the audio recognition rate is above 93%, and for other audios without the TV standard channels can be detected steadily. Keywords: audience rating; audio retrieval; robustness; recognition rate

Session 4aUWa

Underwater Acoustics and Signal Processing in Acoustics: Time Series Analysis and Data Processing in Underwater Acoustics I

Nicholas Chotiros, Cochair
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Invited Papers

9:20

4aUWa1. The pattern of echoes in feature space. Xiukun Li (Acoustic Science and Technology Laboratory, Harbin Engineering University, Harbin, China, xiukun_li@yahoo.com.cn), and Zhi Xia (Acoustic Science and Technology Laboratory, Harbin Engineering University, Harbin, China)

In order to increase the performance of recognition in underwater bottom object detection, the distribution pattern of different kinds of echoes is researched in this paper. The past literatures more focus on the characteristics of object echoes in feature space, but the research of joint distribution pattern of object echoes, reverberation and other interference is few. In this paper, reverberation and fake object echoes are taken as the mainly interference for recognizing object echoes, and they are assumed have steady characteristic in time-frequency feature space, respectively. There are two problems are discussed: the separability between different kinds of echoes in feature space, and the stability of them. To deal with these two problems, feature compression and cluster analysis are adopted. And the feature space is generated by FDWT. The data acquired from a lake experiment is processed in this paper, and the processing results prove that object echoes, fake object echoes and reverberation within limited time scale have steady distribution pattern in FDWT feature space.

9:40

4aUWa2. Non-Rayleigh reverberation statistics. Nicholas Chotiros (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, Texas 78713-8029, chotiros@arlut.utexas.edu)

In sonar reverberation, the superposition of numerous scattered sound waves tends to produce a Rayleigh distribution, consistent with the Central Limit Theorem. Causes for deviation from the Rayleigh model are identified as sonar configuration and environmental variability. In the former, the sonar configuration determines the number of scatterers in each resolution cell, and when the number is too small, the Central Limit Theorem is violated. In the latter, the total environment may be considered as a patchwork of local environments that are resolvable by the sonar system, but not reliably distinguishable due to positional inaccuracies and overlap in the range of reverberation amplitude values. In that case, the resulting ensemble may have a probability distribution function that is a mixture of the probability distribution functions of the local components. The patchiness of the environment determines the number of components and their proportions in the mixture. The issue of stationarity in the context of a patchy environment is an important concern. Although the reverberation from a patchy environment is, strictly speaking, non-stationary, the perception of stationarity may be achieved. [Work supported by the Office of Naval Research, Ocean Acoustics].

10:00

4aUWa3. Acoustic forward scattering by a moving object and its range estimation in littoral experiment. Bo Lei, Kunde Yang, and Yuanliang Ma (Institute of Acoustic Engineering, Northwestern Polytechnical University, lei.bo@nwpu.edu.cn)

When an object crosses the source-receiver line, the sound field aberration can be caused by the forward scattering. However, the aberration is difficult to be seen because of the direct blast overwhelming. An experiment was conducted in littoral environment with several hydrophones deployed at different depths. A repeated wideband LFM pulse is transmitted and data is processed with correlation. The experimental results show that the sound field aberration takes minimum values if the object is located mid-point along the source-receiver line, whereas it attains its maximum if the object is close to the source or receiver. The total field is either enhanced or suppressed if the object crosses different Fresnel zones. In addition, the duration of shadow-induced aberration is dependent on the width of the first Fresnel zone, which is longest at the mid-point of the source-receiver line. Furthermore, a range estimation scheme is proposed. The scheme uses two-point field aberrations of the stable arrival caused by forward scattering of moving object. The ranges of intruder are estimated with a prior knowledge of the moving speed, which agree well with the measurements.

10:20

4aUWa4. Diversity combining for long-range acoustic communication in deep water using a towed array. Hee-Chun Song (UCSD 9500 Gilman Drive, La Jolla, CA 92093-0238, hcsong@ucsd.edu), and William Hodgkiss (UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238)

A recent experiment showed that coherent long-range acoustic communication is feasible in deep water over a ~550-km range between a source towed slowly at 75-m depth and a horizontal line array towed at 3.5 knots at 200-m depth. This paper further demonstrates that diversity combining mitigates channel fading and increases the output SNR. Using sparse channel-estimate-based

equalization, three transmissions are combined successfully to decode a 40-Hz bandwidth (230-270 Hz) 8-PSK (phase-shift-keying) communication signal, achieving an effective data rate of 17 bits/s at ~550 km range.

10:40–11:00 Break

11:00

4aUWa5. Fluctuations of arrival time and amplitude for short-range experimental data. Rui Duan, Kunde Yang, and Yuanliang Ma (Institute of Acoustic Engineering, Northwestern Polytechnical University, ykdzym@nwpu.edu.cn)

The arrival time and amplitude fluctuations of about 230-m propagation were analyzed for various source-receiver configurations using experimental data. The minutes-scale and seconds-scale fluctuations of the arrival time were observed on both the direct, surface-reflection and bottom-reflection arrivals while the minutes-scale fluctuation of the arrival amplitude was only observed on the direct arrivals. Cross-correlation coefficients and frequency spectrum of the fluctuation were calculated to explain the causes of the fluctuations. It shows that the movement of the source is an important cause for the fluctuations of the arrival time and the seconds-scale fluctuations of the arrival amplitude. The variability of the ocean structures contributes to both the arrival time and the amplitude fluctuations of the direct arrivals while the sea-surface scattering is the dominant cause for the surface-reflection arrivals. The fluctuation amplitudes of the bottom-reflection and the surface-reflection arrival amplitudes are around 2dB and 7dB, respectively. The fluctuation amplitudes of the direct arrival amplitude range from 1dB to 10dB for different source-receiver configurations.

Contributed Paper

11:20

4aUWa6. Temporal coherence of acoustic signals in range dependent background. Yin Quan Zhang and Ning Wang (Ocean University of China, Qingdao, zhyq_ouc@126.com)

Temporal coherence of acoustic signals is important for many practical applications, which has been studied both theoretically and experimentally. In previous theories, the fluctuation of sound speed field is assumed to be caused by linear random internal waves and the background is assumed to be range independent. This assumption has two obvious shortcomings: (1) actual fluctuations in shallow water are usually modulated by internal tides; (2) seabed has significant effects on acoustic signal propagation in shallow water, the properties of which is, in general, range dependent (such as topography). In the present presentation, internal tides and seabed properties are involved as the range dependent background, in which the background acoustic field is expressed as coupled mode matrix. A semi-analytic formalism for the temporal coherence of acoustic signal is presented. The dependence of temporal coherence on range and frequency, and the impact due to range-dependence is analyzed. The result is compared with the case range-independent.

Invited Paper

11:40

4aUWa7. Robust adaptive beamforming based on convex optimization. Lianghao Guo, Xin Guo, and Feng-Xiang Ge (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-Huan-Xi Road, Beijing, China, glh2002@mail.ioa.ac.cn)

Adaptive beamforming is an efficient way of spatial filtering in the presence of interference and noise. However, the conventional adaptive beamforming, e.g., MVDR, may degrade significantly due to the poorly estimated covariance matrix or steering vector errors. Convex optimization has now emerged as a major signal processing tool and made a significant impact on numerous problems because of its foundational nature and potential ability in signal processing. Thus in this paper, several robust adaptive beamforming algorithms based on the convex optimization are presented and evaluated, where a novel mathematical tool called "Yalmip" is used to solve the second-order cone problem in these algorithms. Numerical simulations and comparison show that these robust adaptive beamforming algorithms still approach robust to the above-mentioned problems.

Contributed Papers

12:00

4aUWa8. The study about through signature of underwater target based on focused beam-forming. Xue-li Sheng (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001, shengxueli@yahoo.com.cn), Yan-qiong Liu (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001 and Dalian Scientific Test and Control Technology Institute, Dalian 116013), Chun-ping Zhai (Dalian Scientific Test and Control Technology Institute, Dalian 116013), Shi-cai Zhu (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001 and Dalian Scientific Test and Control Technology Institute, Dalian 116013), Yu-dong Liu, Zuo-Xi Tian (Dalian Scientific Test and Control Technology Institute, Dalian 116013), and Jun-ying Hui (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001)

In shallow water, the peak and energy of through signature change because of the stack of reflected sound from the sea bottom and sea surface, it will cause much error in signal process. In this paper, the focused beaming-forming method is introduced in through signature processing. Factors of influencing through signature based on focused beaming-forming method are analyzed and the parameters describing through signature are presented. Processing results of actual measurement signal show that if the parameters describing through signature are not affected significantly, it can restrain noise effectively to smooth the through signature curve based on focused beaming-forming method. It also enhances the gain of through signature. Key words: multi-path effects; underwater acoustic image; beaming-forming; through signature; the gain;

12:20

4aUWa9. High-resolution direction finding without coherent signals number based on multibeam system. Jie Zhuo (Institute of Acoustic Engineering, Northwestern Polytechnical University, jzhuo@nwpu.edu.cn), and Bing Li (State Key Laboratory for Manufacturing System, Xi'an Jiaotong University)

Multibeam acoustic imaging systems are widely utilized for both large- and small-scale underwater investigations, especially for detecting underwater target. Processing backscattered echoes from the targets, the system can plot an acoustic image of target, and then estimate the parameters, such as direction and range. The backscattered echoes of target are coherent signals, and can be modeled as multiple highlights. Usually, the distance between the highlights is close, and the number of the highlights is unknown. Then, the directions cannot be directly estimated due to these highlights are not distinguished in the acoustic image. In this paper, an high-resolution method is proposed for estimating the directions of arrival (DOAs) of coherent highlight signals without signals number. The multibeam underwater acoustic imaging technology and the beamspace MUSIC algorithm are combined together. In this method, eigen-decomposition is skipped, so that coherent signals' DOA can be estimated by high-resolution MUSIC method under the situations with unknown signals number. Due to estimating the DOAs of coherent signals in beamspace, the precision of DOA estimation is enhanced, and the computation complexity is less than the element-space MUSIC method. The method's feasibility and robustness are analyzed and verified by the simulation data.

THURSDAY MORNING, 17 MAY 2012 — S426 + S427, 9:20 A.M. TO 12:40 P.M.

Session 4aUWb

Underwater Acoustics and Signal Processing in Acoustics: Waveguide Invariance Characterization and Processing

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Invited Papers

9:20

4aUWb1. Group speed versus phase speed analysis of sound speed fluctuations in a shallow water ocean. W. A. Kuperman, Bruce D. Cornuelle, W. S. Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238, wkuperman@ucsd.edu), and Philippe Roux (ISTerre, Universite Grenoble 1, CNRS UMR 5275, Grenoble, France)

Data collected during eight consecutive hours between two source-receiver arrays in a shallow water environment (Roux et al, J. Acoust. Soc. Am. 124, 3430-3439, 2008) are analyzed using the physics associated with the waveguide invariant. In particular, the use of vertical arrays on both the source and receiver sides provide launch and receive angles in addition to the travel times associated with each eigenray path in the waveguide. From travel times and source-receiver angles, each eigenray amplitude is projected into group velocity-phase velocity (vg-vp) space for each acquisition. The time evolution of the vg-vp representation during the 8-hour long

experiment is discussed. Group speed fluctuations observed for a set of eigenrays with upper turning points at or near the thermocline are compared to independent sound speed measurements obtained by a CTD chain at two locations in the area.

9:40

4aUWb2. Autocorrelation-function-based estimation approach and variability of waveguide invariant in fluctuated shallow water. Zhou Shihong, He Li, Ren Yun, and Zhang Renhe (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, shih_zhou@yahoo.com.cn)

Waveguide invariant, which describes the broadband acoustic field interference striations in underwater acoustic spectrograms due to multipath phenomena of sound propagation, is beneficial to be applied to temporal and spatial array signal processing specially for large-aperture horizontal line array, etc. However, the dependence of waveguide invariant on fluctuated water-column environment due to internal waves or internal tides is often complicated and shows different interference patterns, which influents the performance of temporal and spatial array signal processing. This presentation focuses on variability of waveguide invariant in fluctuated shallow water environment. One novel estimation approach of waveguide invariant based on phase-shift-compensation of autocorrelation functions at two hydrophones with horizontal range separation is presented. The experimental airgun-emitted pulse data acquired in iso-speed shallow water environment is used to verify the estimation approach. Based on it, the variability of waveguide invariant in fluctuated shallow water with internal wave effects are analyzed using long-term observed LFM signals obtained by one sea-floor located horizontal-line-array and one fixed sound source. The contribution of low- and higher- order filtered modes to interference patterns is given for explaining the mechanism of mode interference. Some interesting experimental results are shown.

10:00

4aUWb3. Variability of the waveguide invariant in a range independent shallow-water environment. Kevin Cockrell (Massachusetts Institute of Technology, Dept. Mech. Engineering, Cambridge, MA 02139, kevincockrell@gmail.com)

While it is generally true that the waveguide invariant β is approximately equal to one in many shallow-water environments, the value of β observed in acoustic intensity striation patterns often deviates from the assumed value of one by 30% or more. The precise value of β, like the acoustic field itself, depends on source frequency, source and receiver depths, sound speed profile, range, etc. For example, previous work has found that the observed value of β in shallow water can deviate from the typical value of one if the source and receiver are located below the thermocline, because in that case the acoustic intensity is strongly affected by lower order modes which do not have $\beta \approx 1$ [D. Rouseff, Waves in Random Complex Media, 2001]. This talk will further explore the dependence of β on the source and receiver depths, and on the sound speed profile by using the WKB approximation. Because the group of modes which dominates the acoustic intensity depends on range and frequency, the dependence of β on range and frequency will also be explored.

10:20

4aUWb4. The waveguide invariant β and bottom reflection phase-shift parameter P. Erchang Shang, Jinrong Wu, and Zhendong Zhao (Key Laboratory of Underwater Acoustic Environment, Institute of Acoustics, Chinese Academy of Science, 100190, ecshang32@aol.com)

It is known that the waveguide invariant β is affected by the shallow-water environment. In this paper the effect due to bottom sediment is investigated. It is found that the effect of sediment bottom can be concentrated on one parameter P – the bottom reflection phase shift parameter. For a Pekeris waveguide, under WKB approximation, a very simple analytic relation is given: $\beta \approx 1+P/(k0Heff)$, here Heff is the 'effective depth', and Heff =H+P/2k0. It is shown that the value of β related to different high-speed sediments (including layered sediment) is ranged in 1.0 and 1.5. Some numerical examples including the layered sediment case are conducted to verify this result. Good agreement between the results calculated by KRAKEN and by WKB with parameter P has been found. The advantage of using parameter P is that it can cover any type of high-speed sediment even including shear wave effect, moreover, it also works for layered sediment provided using P(ømega) instead of constant P. Hence, by using parameter P allows us to have a model-free platform to investigate the sound field in shallow-water including the bottom effect on the waveguide invariant β. [This work was supported by NSFC under Grant No.10874201 and No.11074271]

10:40–11:00 Break

Contributed Papers

11:00

4aUWb5. Waveguide invariant and dedispersion transform. Gao Dazhi (Ocean University of China, dzgao@ouc.edu.cn)

The waveguide invariant is described as a single scalar parameter for a given waveguide environment. The notion of waveguide invariant has been applied widely in under water acoustic source ranging, beamforming etc. A Fourier-like transform called to dedispersion transform, which can remove the dispersion of multi-modes at the same time, was proposed in our previous paper ("Ning Wang, 9th. Western Pacific Acoustic Conference Beijing 2009"). In this presentation, we refine the relationship between the waveguide invariant and the dedispersion transform, and the dedispersion transform is extended to low frequency region and typical summer waveguides. Numerical simulations and several real-data processing will be also reported.

11:20

4aUWb6. Acoustical monitoring of the second mode internal solitary wave on oceanic shelf. Andrey Lunkov (A.M.Prokhorov General Physics Institute, Russian Academy of Sciences, 38 Vavilov st., 119991 Moscow, Russia, landr2004@mail.ru), Hwung-Hweng Hwung (National Cheng Kung University, 5th F., 500, Sec.3, Anming Rd., Tainan 70955, Taiwan), Valeriy Petnikov (A.M.Prokhorov General Physics Institute, Russian Academy of Sciences, 38 Vavilov st., 119991 Moscow, Russia), Yu-Huai Wang (National Sun Yat-Sen University, No. 70, Lien-hai Rd, Gushan, Kaohsiung, Taiwan, 804, R.O.C), and Ray-Yeng Yang (National Cheng Kung University, 5th F., No. 500, Sec. 3, Anming Rd., Tainan, Taiwan 709)

The possibility of monitoring second gravitational mode internal solitons using interference pattern frequency shifts is discussed. The investigation is carried out for the shelf of the northern South China Sea near the

4a THU. AM

Dongsha Atoll by means of numerical modeling. We used the data of the in-situ internal solitons measurements in this area. The "vertical modes and horizontal rays" approach is implemented to calculate the low frequency sound fields in the 3d environment. Stationary acoustic path is oriented at the right angle to the preferred internal wave propagation direction. A sound source and receivers are deployed at the sea bottom. Sound receivers are located at the different ranges where horizontal refraction is pronounced, and where it is insignificant. Numerical experiments demonstrate that some of acoustic waveguide modes are focused, and others are defocused in the horizontal plane when the second mode soliton propagates across the acoustic path. It is shown that the second mode internal solitary wave parameters can be successfully reconstructed from the frequency shifts in the spectrum of received signals only if horizontal refraction effects are weak. [Work supported by Russian Foundation for Basic Research and National Science Council of Taiwan # 10-02-92005]

11:40

4aUWb7. Generalized array invariant and its application on broadband source ranging. Qi Chun Shang, Shuang Zhang, and Ning Wang (Ocean University of China, 238 Songling Road, Qingdao, China, maymayed2007@126.com)

A passive source ranging method based on the array invariant in shallow water is discussed in this paper. The arriving time and elevation angle of sound are used to describe the multi-modal propagation and single-mode dispersion. Based on the two parameters, we rederive the array invariant in a different way, which allows simple physical interpretation. The array invariant in its original form is not exact invariant, but depends (weakly) on mode number and frequency. The accuracy of the method based on this notion is limited when the sound speed of seabed is different significantly from that in the water column. A modified technique (generalized array invariant) is proposed in this talk to improve the problem provided when the sound speed of seabed is known. The proposed method is testified by simulation and experimental data.

12:00

4aUWb8. Impulse signal reconstruction using bi-receiver data. Haozhong Wang, Ning Wang, and Dazhi Gao (College of Information Science and Engineering, Ocean University of China, 238 Songling Road, Qingdao 266100, China, coolicejiao@hotmail.com)

Signal reconstruction is used widely in target identification and communication in underwater. A novel method for impulse signal reconstruction using the observed data of two receivers that are arranged in the same depth with a certain horizontal interval, is proposed in this talk. This method needs no a prior environmental information but the ranges between the source and the receivers. Although the Green's function depends on the range, frequency and on the source/receiver depth, the spectrum of signal is only dependent on the frequency. The waveguide invariant notion of shallow water provides a compensation mechanism between the frequency and range shift. According to this mechanism, the amplitudes and phases of Green function spectral ingredients can be extracted respectively. The impulse signal is then obtained by employing the deconvolution. The method is applied to the signal reconstruction of a high S/N ratio real data, the correlation coefficient between the reconstructed and original signals is over 0.95.

12:20

4aUWb9. Time-reversal focusing stability in the presence of background internal waves in shallow water. Valeriy Petnikov and Andrey Lunkov (A. M. Prokhorov General Physics Institute, Russian Academy of Sciences, 38 Vavilov St., 119991 Moscow Russia, petniko@kapella.gpi.ru)

Effect of background internal waves on spatial and temporal low frequency sound focusing stability is investigated for an open oceanic shelf by means of numerical simulations. Focusing is achieved with the time-reversal procedure at a single source-receiving element at 10km range. Calculations are performed in terms of normal mode coupling theory. Internal wave field modeling is carried out using an averaged experimental power spectrum of vertical thermocline displacements measured in the Shallow Water'06 experiment. The results of numerical experiments show that the focal spot is stable for about 1 hour in the presence of typical internal waves on an open shelf. Two adaptive time-reversal algorithms are proposed to increase this period up to 12 hour. [Work supported by RFBR 11-02-00779.]

THURSDAY AFTERNOON, 17 MAY 2012 HALL A, 1:55 P.M. TO 6:00 P.M.

Session 4pAA

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

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

Invited Papers

2:00

4pAA1. Syllables intelligibility in relation to the autocorrelation and cepstrum model: the case of Chinese in Taiwan. Chiung Yao Chen (Chaoyang Univ. of Technology, chychen@cyut.edu.tw)

The articulation of some special pronunciations of vowel inconsistently raises with the rapid speech transmission index (RASTI) tested using monosyllables for Chinese phonics. In the researching of speech intelligibility in room, the factors being considered have to include not only the qualities of sound field but the pronunciation characteristics of syllables as well. Therefore, with regard to the defect of RASTI measurements, we utilized the autocorrelation and the cepstrum of monosyllables recorded in rooms to compare the articulation with the physical phenomena of intelligibilities. Thus, we found that the minimum effective duration of autocorrelation function (τ)

of all testing syllables signals recorded in rooms, associated well with the articulation for each individual room. Specially, for individual syllable signals, they are further shown significant correlation between the cepstral energy of monosyllables and the articulation collected in all rooms. The cepstrum acts as a spatially intelligible detector for syllables, and the autocorrelation is a good response of the pronunciation characteristics of syllables.

2:20

4pAA2. The role of early reflections for definition and source separation. Antti Kuusinen, Jukka Pätynen, Philip Robinson, and Tapio Lokki (Aalto University School of Science, P.O. Box 15400 00076, Aalto, Finland, Antti.Kuusinen@aalto.fi)

The density and spatial distribution of early reflections change the perception of individual instruments in an orchestral concert. Here, we present results of a listening test that aims to find the role of early reflections to perceptual phenomena related to source separation. The stimuli are anechoic symphony orchestra recordings convolved with different impulse responses that comprise of simulated direct sound and early reflection patterns combined with the late reverberation measured from a real hall. The differences between samples are investigated with various listening test methods, enabling the simultaneous comparison of samples. Quantitative data are also collected with the applied test methods. The results are expected to confirm our hypothesis that listeners can distinguish individual instruments better if the early reflection patterns in conjunction with individual sources differ more from each other.

2:40

4pAA3. Does listener weighting of binaural cues take advantage of the binaural statistics of reverberant environments? G Christopher Stecker (University of Washington, 1417 NE 42nd St, Seattle, WA 98105, cstecker@uw.edu), and Andrew D Brown (University of Washington, 1417 NE 42nd St, Seattle, WA 98105)

A series of experiments quantified listeners' weighting of auditory spatial information conveyed by interaural differences of time (ITD) and level (ILD) across cue type (ITD-ILD "trading") and over the durations of brief sounds ("temporal weighting"). Results demonstrated the dominance of cues, especially ITD, carried by the onsets of rapidly modulated or continuous tones. Whereas post-onset ITD information received little weight for such sounds, post-onset ILD was more influential, especially near sound offset [Stecker and Brown, *JASA* 127:3092-103. 2010]. As a consequence, the relative weighting of ITD and ILD changes systematically with modulation rate, with greater weighting of ILD in cases where post-onset ITD is unavailable [Stecker, *Hear. Res.* 268:202-12. 2010]. Greater weighting of post-onset ILD than ITD is consistent with observations of dramatic ITD distortion by echoes [Rakerd and Hartmann, *JASA* 78:524-33. 1985] and of a greater role for ILD in dynamic aspects of the precedence effect [Krumbholz and Nobbe, *JASA* 112:654-63. 2002] resulting from changes in the acoustic environment. Evidence regarding temporal weighting of ITD and ILD will be reviewed and compared to the statistics of binaural cue values across a variety of reverberant recordings. [Supported by NIH R03-DC009482, R01-DC011548, F31-DC010543, T32-DC000033]

3:00

4pAA4. Binaural room acoustics II: Distributions and consequences of interaural differences. William M. Hartmann (Michigan State University, 4208 BPS Bldg., East Lansing, MI 48824, hartmann@pa.msu.edu), Brad Rakerd, and Eric J. Macaulay (same)

Binaural room acoustics attempts to generalize the acoustical properties of rooms as they appear at the two ears of a listener. Because of the importance of sound localization, interest has focused on interaural differences in intensity level and phase. Distributions of these interaural differences, as measured with an artificial head in different rooms, are in reasonable agreement with spherical-head computations having either the direct-to-reverberant ratio or the reverberation time as the main parameter. The psychological relevance of these distributions was tested in experiments using rooms with different reverberation times where listeners were required to report the source azimuth for steady-state pure tones having frequencies between 200 and 1200 Hz. Simultaneously probe microphones in the ear canals recorded interaural differences. Interest centered on the choices made by listeners between plausible and implausible interaural differences resulting from the sound fields in the room. Particular attention was given to measurements made near the binaural critical distance [Hartmann and Rakerd, J. Acoust. Soc. Am. **130** 2352 (2011). [Work supported by the AFOSR, grant 11NO002]

3:20

4pAA5. The contribution of interaural time and level differences to the precedence effect at high frequencies. Bernhard U. Seeber (MRC Institute of Hearing Research, University Park, Nottingham, NG7 2RD, UK, seeber@ihr.mrc.ac.uk)

The precedence effect (PE) allows us to locate sound sources correctly in rooms despite the presence of interfering reflections. It has been shown to function at high frequencies with highly modulated stimuli. These studies were done in the free-field where interaural time (ITDs) and level (ILDs) differences are in their natural combination. The present study investigated the relative contribution of ILDs and ITDs to the PE with high-frequency zero-phase harmonic complex tones. A localization dominance task was used in which participants indicated the location of the lead-lag stimuli and judged if sounds were perceived as fused. A preliminary analysis indicates that the PE emerged when either ITDs or ILDs were applied to lead and lag stimuli while the other binaural cue was held at zero. Patterns for localization dominance and fusion were nearly identical for ITD and ILD conditions, suggesting that ITDs and ILDs were equally effective for these highly modulated stimuli. Fusion of lead and lag extended to somewhat longer delays with smaller cue magnitudes, i.e. the more binaural cues differed between lead and lag the more likely they were to be segregated. The results support the idea that PE mechanisms are similar for ITDs and ILDs.

4p THU. PM

3:40

4pAA6. Echo thresholds for reflections from acoustically diffusive architectural surfaces. Philip W. Robinson (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, Troy, New York, philrob22@gmail.com), Andreas Walther, Christof Faller (Audiovisual Communications Laboratory, école Polytechnique Fédérale de Lausanne, Switzerland), and Jonas Braasch (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, Troy, New York)

Diffusive architectural surfaces play an important role in performance venue design for architectural expression and proper sound distribution. However, previous psychoacoustic research on perception of reflections and the precedence effect has focused on specular reflections. This study compares the echo threshold of specular reflections, measured using an adaptive up-down method with music and

speech stimuli, against those for reflections from realistic architectural surfaces, and against synthesized reflections that isolate individual qualities of reflections from diffusive surfaces, namely temporal dispersion and spectral coloration. It is found that temporal dispersion up to 16ms in the reflection response, and peak amplitude reduced as much as 18.5 dB results in an echo threshold shorter than that for a specular reflection of comparable amplitude. Rather, the threshold is comparable to that of a specular reflection of similar energy. This indicates that the auditory system integrates the temporally dispersed energy into a single stream.

4:00–4:20 Break

4:20

4pAA7. A binaural model that uses head-movements to evaluate acoustical spaces. Jonas Braasch (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180, braasj@rpi.edu), Samuel Clapp, Anthony Parks, and Ning Xiang (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180)

Binaural models have a long tradition in the instrumental analysis of acoustical spaces. Room acoustical parameters such as the Binaural Quality Index (BQI) are derived directly from the measured Binaural Room Impulse Response (BRIR) of a concert space. The BRIRs are measured using an artifical head with a fixed head position and consequently cannot show the effect of head movements, which are essential for human listening performance. Based on a novel model architecture that utilizes head movements [Braasch et al., 2011, J. Acoust. Soc. Am. 129, 2486 (A)] and psychoacoustic experiments, the effect of head movements on the perceptual judgment of room acoustical parameters will be discussed. In addition, BRIRs for different azimuth angles are calculated from higher-order ambisonic microphone data that were obtained in several concert halls in the Northeastern United States. [Clapp et al., 2011, J. Acoust. Soc. Am. 130, 2418 (A)]. It will be demonstrated how the standard acoustical parameters change with head orientation and how dynamic head-movement cues can be utilized to better predict the perceived quality of concert spaces.

4:40

4pAA8. Three dimensional representation method for a public indoor soundscape with multiple sound sources. Yasushi Shimizu (Tokyo Institute of Technology, 226-8502, shimizu.y.ai@m.titech.ac.jp), and Hiroshi Furuya (Shibaura Institute of Technology, 135-8548)

The author has been investigating an evaluation method of a reproduced sound such as paging sound, background music and so on in a public space. The current acoustical descriptors for an evaluation of such sounds, which are played to deliver information and attention to the people, have a difficulty to apply to multiple sounds environment. This report describes a new evaluation method which utilizes drawing an indoor soundscape with multiple sounds from listening experience, based on major aural impressions such as Loudness, Timbre, Apparent Sound Source Width, horizontal and vertical Sound Localization, Distance Perception, and Perspicuity, “KIWADACHI”. This will be applied to describe the Indoor Sound Environmental Character with multiple sound sources. The evaluation tests with this tool are carried out in the indoor sound environments of a retail shop, regarding aural impression of Perspicuity for the sound reproduction. And the results of both the subjective representation in the indoor sound environment and the acoustical descriptor of speech intelligibility are presented for the reproduced sound.

5:00

4pAA9. Using time-varying loudness to model the reverberance of rooms. Densil Cabrera, Doheon Lee, and William L. Martens (The University of Sydney, NSW 2006, Australia, densil.cabrera@sydney.edu.au)

The reverberant decay of sound over time is one of the most perceptually salient features of reverberation in rooms. This paper examines the concept that the perception of reverberation decay can be modelled using dynamic loudness. As well as being a plausible application of time-varying loudness modelling, this approach helps to explain why higher sound pressure level stimuli are more reverberant than otherwise identical reverberant stimuli – because the slope of the loudness decay function depends on the stimulus level. Loudness decay parameters are derived in analogy to conventional reverberation parameters (reverberation time and early decay time), which provide a better match to subjective experimental data concerned with: impulsive, running, and overall reverberance; and using artificial and measured room impulse responses.

5:20

4pAA10. Loudness asymmetry in real-room reverberation: cross-band effects. Andrew Raimond and Anthony Watkins (Reading University, Reading RG6 6AL, UK, andrew_raimond@hotmail.co.uk)

Although room reverberation adds sizeable “tails” at the ends of sounds, they are not prominent for listeners. Evidence for this comes from loudness judgements of stimuli with envelopes having fast onsets and slow offsets, thereby resembling sounds with reverberant tails. Such sounds are less loud than their reversed counterparts, and this difference is more substantial when the test-sound is preceded by a “standard” sound that has a similarly “tail-like” offset. A perceptual constancy may be responsible; one where sounds with decaying tails are “parsed” to separate source characteristics from effects of reverberation, and to discount energy in tails from listeners’ judgements. Indeed, this “loudness context effect” is even more substantial when real-room reverberation is used. Here we ask whether the effect is restricted to the frequency region occupied by the context; conditions where standard and tests occupy the same narrowband frequency region are compared with “cross-band” conditions, where the standard and test have widely separated frequency regions. Results show that the effect is markedly reduced in cross-band conditions, indicating that the perceptual constancy responsible is a “within band” phenomenon. Similar within-band characteristics are also evinced by a form of constancy in speech perception, where the salience of “tails” from reverberation is also reduced.

5:40

4pAA11. The desire for decorrelation—applications from the recording studio. Alexander Case (University of Massachusetts Lowell, 35 Wilder St, Lowell, MA 01854, alex_case@uml.edu)

Multitrack production solves challenges of masking, localization, and intelligibility while pursuing aesthetics associated with reverberance, distance, envelopment, and source width through any means available in the signal processing-rich environment of the recording studio. Contemporary sound recording techniques are presented that might influence the design for achieving similar results through architectural signal processing.

Session 4pAB

Animal Bioacoustics: Tropical and Sirenian Bioacoustics

Jun Xian Shen, Cochair
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Peter Narins, Cochair
pnarins@ucla.edu

Invited Papers

2:00

4pAB1. Ultrasonic hearing in frogs: inner ear morphological correlates. Peter Narins (UCLA, 621 Charles E. Young Drive S., Los Angeles, CA 90095-1606, pnarins@ucla.edu)

Three species of anuran amphibians (*Odorrana tormota, O. livida* and *Huia cavitympanum*) have recently been found to detect ultrasounds. We compared morphological data collected from the ultrasound detecting species with data from *Rana pipiens*, a frog with a typical anuran upper cut-off frequency of ca. 3 kHz. In addition, we examined the ears of two species of Lao torrent frogs, *O. chloronota* and *Amolops daorum* that live in acoustic environments resembling those of the ultrasonically sensitive frogs. Our results suggest that the three ultrasound-detecting species have converged on small-scale functional modifications of the basilar papilla (BP), the high-frequency hearing organ in the frog inner ear. These modifications are also seen in the ears of *O. chloronota*, suggesting that this species is a candidate for high-frequency hearing sensitivity. These data form the foundation for future functional work probing the physiological bases of ultrasound detection by a non-mammalian ear. Supported by NIDCD DC-00222, Paul S. Veneklasen Research Foundation, and the UCLA Academic Senate (3501).

2:20

4pAB2. High-frequency sound communication in the concave-eared frog. Jun-Xian Shen (Institute of Biophysics, CAS, 15 Datun Road, Chaoyang District, Beijing 100101, China, shenjx@ibp.ac.cn)

The concave-eared torrent frog, Odorrana tormota, is an arboreal, nocturnal frog living near noisy fast-flowing streams in Huangshan China. Recordings in the field show that males produce diverse melodic calls containing spectral energy extended to the ultrasonic range. Playbacks of the audible as well as the US components of a male call can evoke males' vocal responses. Auditory evoked potentials from the auditory midbrain confirm that males possess the US hearing capacity. Before ovulation, gravid females produce high-frequency short calls, which elicit vocalization and precise positive phonotaxis from males. Acoustic playbacks of male's calls also evoke vocal responses and phonotaxis from females, but the females show no ultrasonic sensitivity. This suggests that the high-frequency sound communication system has evolved in the frog species. [Work is supported by the National Natural Science Foundation of China (NSFC grants Nos. 30570463 and 30730029 to J.-X.S.]

Contributed Papers

2:40

4pAB3. Anatomical changes in the inner ear of the bullfrog across metamorphic development. Erika E. Alexander (Brown University, Campus Box 1821, Providence, 02912, Erika_Alexander@brown.edu), Andrew M. Tarr, and Andrea M. Simmons (Brown University, Campus Box 1821, Providence, 02912)

Metamorphic development in the bullfrog, Rana catesbeiana, is characterized by widespread changes in peripheral transduction pathways and in the auditory brainstem, in preparation for the transition from a fully aquatic to a semi-terrestrial existence. The time course of development of the inner ear organs has not been as extensively examined. A combination of immunohistochemical, cresyl violet and trichrome staining to were used to delineate the development of the saccule, an otolith organ sensitive to particle motion and to seismic stimuli, across metamorphosis. From early embryonic to metamorphic climax stages and extending to the froglet period, the saccule increases linearly in area, correlated with the growth in body size. Myosin VI label indicates that hair cell density in the central region of the saccule remains relatively stable in tadpoles, but then decreases between froglet and subadult stages. From these results, it is hypothesized that hair cell proliferation occurs more extensively in tadpoles than in froglets.

3:00

4pAB4. Photolyses of carboxy-hemoglobin of bar-headed goose studied by photoacoustic calorimetry. Jin-yu Zhao (Lab of Modern Acoustics, College of Physics, Nanjing University, Nanjing 210093, China, jyzhao04118@gmail.com), Jia-huang Li (Lab of Pharmaceutical Biotechnology, College of Life Sciences, Nanjing University, Nanjing 210093, China), Zheng Zhang (Nanjing First Hospital Attached to Nanjing Medical University, Nanjing 210006, China), Min Qu, Shu-yi Zhang (Lab of Modern Acoustics, College of Physics, Nanjing University, Nanjing 210093, China), Zi-qian Hua (National Laboratory of Protein Engineering and Plant Genetic Engineering, College of Life Sciences, Peking University, Beijing 100871, China), and Zi-chun Hua (Lab of Pharmaceutical Biotechnology, College of Life Sciences, Nanjing University, Nanjing 210093, China)

As a specialized species native to high altitude, bar-headed goose can fly annually over an altitude of 9000m, which means that its hemoglobin has a higher oxygen affinity than its lowland relatives, such as goose and chicken. To study the mechanism of the phenomena, laser ultrasonic calorimetry is used to study dynamic processes associated with photolyses of carboxy hemoglobin (HbCO), including the enthalpy and conformational volume changes, of bar-headed goose and its lowland relatives. Considering the

time scales of the reaction lifetimes in the photolyses processes of HbCO, two kinds of piezoelectric transducers, a PVDF film and a PZT ceramic, are used as acoustic signal detectors. For evaluating the relative enthalpy change and the relative conformational volume change in the process, the quantum yield of the photolysis must be taken into account, which has been measured by pump-probe technique. The results show that the enthalpy and conformational volume changes of bar-headed goose are obviously smaller than that of its lowland relatives and human. Some analyses and discussions on the differences of the amino acid sequences of Hb, the tetramer structures, as well as the salt bridges between subunits of Hb and HbCO among them are presented.

3:20

4pAB5. Design of a home-made, low-cost system for studies of vibratory courtship signals on *Pardosa Sierra* (Areneae: Lycosidae) spiders. Eduardo Romero-Vivas, Emiliano Méndez Salinas, María Luisa Jiménez Jiménez, and Francisco Javier García De León (Centro de Investigaciones Biológicas del Noroeste, S.C., Mar Bermejo 195 Col. Playa Palo de Santa Rita, La Paz, BCS, 23090, México, evivas@cibnor.mx)

Spiders possess peculiarities that make them attractive for the study of evolutionary phenomena such as adaptation and specialization. Among these processes, reproductive behavior (particularly courtship) is a main factor, allowing or preventing recognition between potential partners. Spiders sense their environment and communicate using chemical, visual and acoustical/vibrational signals. The study of the nature, variation and content of these signals, provides useful information to understand the role of communication in the formation of species. Vibrational signals excel in importance in the majority of spider families and have been previously studied, especially in leaf-living spiders, using non-contact laser Doppler vibrometers or accelerometers (adding extra mass to the system) coupled to charge amplifiers. Unfortunately, cost and availability of this equipment have limited the widespread of studies in this area. This paper describes how to build an alternative low-cost system for the study of vibrational signals on spiders, and presents the analysis of the acquired vibratory courtship signals of Pardosa sierra, a rocky substrate-dwelling lycosid spider.

3:40

4pAB6. Intraspecific variation in vocal repertoire among dugong populations. Kotaro Ichikawa (Research Institute for Humanity and Nature, 603-8047, Kyoto, Japan, ichikawa.kotaro.dugong@gmail.com), Tomonari Akamatsu (National Research Institute of Fisheries Engineering, 314-0408 Ibaraki, Japan), Kanjana Adulyanukosol (Phuket Marine Biological Center, 83000, Phuket, Thailand), Giovanni Damiani, Janet Lanyon (University of Queensland, St Lucia, Queensland, 4072, Australia), and Hiroshi Nawata (Research Institute for Humanity and Nature, 603-8047, Kyoto, Japan)

Previous studies have demonstrated that vocal signals facilitate acoustic communication of dugongs. We recorded wild dugong calls from around Talibong Island, Thailand (n = 586) and in Moreton Bay, Australia (n = 331). We also recorded vocalizations of a newborn calf (n = 315) kept at Phuket Marine Biological Center, Thailand, a 19 year old female (n = 73) at Toba Aquarium, Japan, and a 7 year old female (n = 203) at Underwater World, Singapore. Dominant frequency, duration and coefficient of frequency modulation were compared across populations and age. Statistical differences were found for almost all pairwise comparisons ($p < 0.05$) except between the captive dugongs kept in Japan and also between wild dugongs in Thailand and in Australia. A negative correlation was found between variance of the dominant frequency and dugong age, and a positive correlation was found between variance of the duration and age. The average dominant frequency of wild dugong calls collected in Thailand and in Australia were 5205.4 and 5760.2 Hz, respectively. These acoustic characteristics ranged between those of the 7 and 19 year old female. Our results suggest that dugongs change their vocal repertoire as they grow.

4:00–4:20 Break

4:20

4pAB7. Analysis of passive acoustic recordings made during a three month survey of cetaceans off the Northern Mariana Islands in the western North Pacific. Thomas Norris (Bio-Waves Inc., thomas.f.norris@bio-waves.net)

Passive acoustic monitoring using was used to complement a line-transect survey of marine mammals for a large (~580,000 km2) study site centered on the Northern Mariana Islands in the western North Pacific. A two-element towed hydrophone array was used to monitor and record during daylight hours. Sonobuoys were deployed opportunistically on sightings and areas of interest. Extremely poor sighting conditions hindered visual efforts but not the passive acoustics effort. Over 70 days of survey effort was completed from mid-January to April, 2007. Approximately 220 'unique acoustic detections' were made, of which 155 (70%) were preliminarily identified to 14 different species. The most frequent whale detected was the sperm whale (65), followed by minke whales (30) and humpback whales (12), respectively. The first recordings of calls from Sei whales in this region are characterized. Post-processing of minke and sperm whales recordings resulted in approximately 30 and over 70 localizations, respectively. We present the first acoustic-based estimates for minke whales abundance in this region. Numerous unidentified odontocete whistles were analyzed using ROCCA, a semi-automated whistle classification program, with promising results. We provide recommendations for additional analyses and improvements to methods of collecting and post-processing passive acoustic data on marine mammals.

Invited Paper

4:40

4pAB8. The Lombard effect in humpback whales. Michael Noad, Rebecca Dunlop (University of Queensland, Gatton, Qld 4343, Australia, mnoad@uq.edu.au), and Douglas Cato (Defence Science & Technology Organisation, Eveleigh, NSW 1430, Australia, and Univeristy of Sydney, NSW 2006, Australia)

The Lombard reflex is an increase in the subject's vocal levels in response to increased noise levels. While it has been demonstrated in humans and a small number of mammals and birds including some whales, it has not been demonstrated in humpback whales. During their southward migration off eastern Australia humpback whales were tracked visually from an elevated land station. An array of calibrated hydrophone buoys was used to simultaneously track vocalizing whales acoustically and to measure ambient noise. Two hundred and ninety two social vocalizations were recorded and analysed from 15 passing groups of whales when there was no detectable boat noise or singing whales in the area. Vocalization source levels increased significantly by a mean of 0.75dB per 1dB increase in background noise (broadband 40Hz – 2kHz). Unlike most previous Lombard studies, however, the vocal level increased even though the background noise was much lower than the vocal level. Thus the whales maintained a signal excess of approximately 75dB which suggests that these social vocalizations may function as signals over distances of several kilometres.

5:00

4pAB9. Variation in the songs of humpback whales (*Megaptera novaeangliae*) wintering in the Northwestern and Main Hawaiian Islands. Jessica Chen, Marc Lammers, and Whitlow Au (Hawaii Institute of Marine Biology, University of Hawaii, 46-007 Lilipuna Rd., Kaneohe, Hawaii 96744, jchen2@hawaii.edu)

A study of the humpback whale song in the Northwestern Hawaiian Islands (NWHI) and the Main Hawaiian Islands (MHI) during the 2009 season suggests that humpback whale song may be more variable than previously suggested. Data from five autonomous acoustic recorders deployed at locations in the NWHI and MHI were analyzed to compare the frequency of occurrence of song units by whales in the island chain. There appears to be a gradient of differences in song units throughout the Hawaiian Island chain, rather than the previously assumed, more discrete differences between breeding populations. Recordings from each site were randomly selected. Song units were classified as one of 23 units and counted to compare between sites. Changes in the frequency of occurrence in a few of the most abundant units suggest a gradual change throughout the island chain. However, this may be confounded by changes that occur throughout the season throughout the ocean basin. Further work examining the amount of variation both between and within humpback whale breeding populations should be conducted.

5:20

4pAB10. Comparison of automated and aural/visual techniques to classify humpback whale (*Megaptera novaeangliae*), song units. Adrienne M. Copeland, Whitlow W. L. Au (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, P.O. Box 1106, Kailua, HI 96734, acopelan@hawaii.edu), Marc O. Lammers (Hawaii Inst. of Marine Biology, Kaneohe, HI 96744), Adam A. Pack (Univ. of Hawaii at Hilo, Hilo, HI 96720), Julie N. Oswald (Bio-Waves, Inc., 517 Cornish Dr, Encinitas, CA 92024), and Jessica Chen (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, P.O. Box 1106, Kailua, HI 96734)

Humpback whale song research has focused on analyzing the full song structure rarely describing individual song units. Even less progress has been made in automatically distinguishing and classifying these individual units. Two different techniques were employed to study their call units, visual/aural and automated/statistical. Humpback whale songs were recorded in the Hawaiian Islands both remotely with an autonomous acoustic recorder and by a snorkeler with a portable digital tape recorder. Humpback whale song units collected by the autonomous acoustic recorders were aurally separated into 23 distinct units in a companion study. Song units collected by a snorkeler using the portable recorder off Maui were analyzed using a specialized Matlab script that defined 48 frequency and temporal parameters for each unit. From the 48 parameters, the units were separated into distinct categories using a multivariate categorical analysis. The distinct units were compared between the different techniques to gage if automated methods could be used in future humpback whale studies. After this comparison was made, a principal component analysis (PCA) determined which of the aforementioned 48 parameters were important in statistically distinguishing between the distinct units furthering our understanding of frequency and temporal importance in categorizing song structure.

5:40

4pAB11. Acoustic issues in studies of behavioral response of humpback whales to seismic ramp-up and hard start. Douglas Cato (Defence Science & Technology Organisation and University of Sydney, P.O. Box 44, Pyrmont, NSW 2009, Australia, doug.cato@sydney.edu.au), Robert McCauley (Curtin University of Technology, Perth, WA 6845, Australia), Michael Noad, Rebecca Dunlop (University of Queensland, Gatton, QLD 4343, Australia), Hendrick Kniest (University of Newcastle, Newcastle, NSW 2308, Australia), Nicholas Gales (Australian Antarctic Division, Kingston, TAS 7050, Australia), Chandra Salgado Kent (Curtin University of Technology, Perth, WA 6845, Australia), David Patton (Blue Planet Marine, Canberra, ACT 2614, Australia), John Noad (University of Queensland, Gatton, QLD 4343, Australia), Curt Jenner (Centre for Whale Research, Fremantle, WA 6959, Australia), Alec Duncan, and Amos Maggi (Curtin University of Technology, Perth, WA 6845, Australia)

Two large behavioral response studies (BRS) have been conducted with humpback whales migrating along the east Australian coastline (in project BRAHSS: Behavioural Response of Australian Humpback Whales to Seismic Surveys). Whales were exposed to four stages of ramp-up with nominally 6 dB increase in level at each step, and a hard start nominally 12 dB above the first stage. Observations of behavior were made by theodolite teams ashore and small boats following specific whale groups, DTAGs, and binoculars from the source vessel. The sound field throughout the area was recorded using five buoys that radioed data back to the shore station, four autonomous receivers and two drifting systems with a vertical array of four hydrophones. Measurements show that the propagation loss at the site is variable and includes patches of anomalously high loss. This complicates estimation of the sound levels received by whales, but may not be unusual in near shore environments. This paper presents preliminary results of the project to illustrate acoustic issues involved in designing and executing comprehensive BRS, including characterization of sources and the acoustic environment experienced by the whales, and monitoring cumulative exposure at individuals for mitigation.

Session 4pBA

Biomedical Acoustics: Bone Quantitative Ultrasound II

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Contributed Papers

2:00

4pBA1. Nonlinear resonant ultrasound spectroscopy is sensitive to the level of cortical bone damage. Sylvain Haupert (University Pierre et Marie Curie, F-75006 Paris, France, sylvain.haupert@upmc.fr), Sandra Guérard (Arts et Metiers ParisTech, F-75013 Paris, France), David Mitton (IFSTTAR, F-69675 Bron, France), Françoise Peyrin (INSA Lyon I, Lyon, France), and Pascal Laugier (University Pierre et Marie Curie, F-75006 Paris, France)

The objective of the study was to evaluate the sensitivity of nonlinear resonant ultrasound spectroscopy (NRUS) measurements to the accumulation of damage in cortical bone by fatigue or by controlled crack propagation. Two groups of human cortical bone specimens were prepared from the femoral mid-diaphysis. The specimens from the first group were taken through a progressive fatigue protocol consisting of four steps of cyclic four-point bending. The specimens from the second group were taken through a toughness protocol consisting of initiation and controlled propagation of a stable crack induced by 4-point bending mechanical loading. Our results evidenced a progressive increase of the normalized nonlinear elastic parameter during fatigue testing or during toughness experiments. While in specimens subjected to mechanical fatigue cycling the relative variation of nonlinear elasticity was significantly related to the relative variation of the number density of small cracks assessed with micro-computed tomography, in crack propagation experiments a significant relationship was found between the level of nonlinearity and total crack length. These results strongly suggest that NRUS measurements are sensitive to damage accumulation and can be used as a marker of bone damage.

2:20

4pBA2. Assessment of soft and mineralized tissue formation in a rat bone healing model using quantitative ultrasound (QUS). Daniel Rohrbach, Bernhard Hesse, Bernd Preininger, Carsten Perka, and Kay Raum (Charite, Julius Wolff Institut, Augustenburger Platz 1, 13353 Berlin, Germany, daniel.rohrbach@charite.de)

It is hypothesized that QUS is a promising candidate for the assessment of early stages of bone healing. 5-MHz QUS measurements in through transmission mode were conducted in vitro on a 2-mm osteotomy rat model (N=10). 2D parametric QUS images of speed of sound (SOS), ultrasound attenuation (UA), and broadband UA (BUA) were registered to histology sections and to projections obtained from 3D μ-CT images. Based on histology two groups of healing (N(A)=5: early healing stage and N(B)=5: early reparative phase) were defined. Parameter variations (medians and integrals) evaluated within the osteotomy gap region were compared between the healing groups. ANOVA revealed significantly higher attenuation and SOS values in group B (UA(A)=14.5$\pm$4.5 dB, UA(B)=35$\pm$10.1 dB, F=18.5; BUA(A)=4.09$\pm$1.7 dB/MHz, BUA(B)=9.6$\pm$4.4 dB/MHz, F=6.8; SOS(A)=1543$\pm$9 m/s, SOS(B)=1590$\pm$34 m/s, F=8.7). ROC analysis with AUC values between 0.84 and 1 confirm a good predictive power of US parameters. The bone mineral density (BMD) based variance assessed from μCT was less pronounced (F=9.5). Moderate correlations of UA and SOS with BMD were observed (R2<0.7). These results demonstrate that ultrasound parameter variations are sensitive to tissue alterations that are not depicted by BMD, but coincide with cartilage formation in the early reparative phase.

Invited Paper

2:40

4pBA3. Therapeutic ultrasound on bone cellular and in vivo adaptation. Yi-Xian Qin, Shu Zhang, Suzanne Ferreri, and Jacky Cheng (Department of Biomedical Engineering, Stony Brook University, Stony Brook, NY, Yi-Xian.Qin@sunysb.edu)

Objective: It is well documented that ultrasound, as a mechanical signal, can produce a wide variety of biological effects in vitro and in vivo. The purpose of the current study was to (1) develop a methodology to allow for in-vitro manipulating osteoblastic cells using acoustic radiation force generated by ultrasound, (2) use this methodology to determine the morphological and biological responses of bone cells to ultrasound, and (3) mitigate bone loss under estrogen deficient osteopenia. Methods: In Vitro Cellular Manipulation: We used a therapy focused transducer, which has spherical cap with 64 mm diameter and 62.64 mm focal length. A laser guide MC3T3-E1 osteoblastic cells were cultured in α-MEM containing 1% penicillin-streptomycin and 10% decomplemented newborn calf serum. In Vivo OVX Model: 72, 16 w.o. Sprague-Dawley rats were divided into six groups; baseline control, age-matched control, OVX control, OVX + 5 mW/cm^2 ultrasound (US), OVX + 30 mW/cm^2 US and OVX + 100 mW/cm^2 US. Low intensity pulsed ultrasound (LIPUS) was delivered transdermally at the L4/L5 vertebrae, using gel-coupled plane wave US transducers. The signal was applied 20 min/day, 5 days/week for 4 weeks. Results: In Vitro Cellular Response: The developed methodology allowed manipulation of MC3T3-E1 cells by acoustic radiation force. The deformation of cell membranes was observed by the US manipulation, which appeared after 15s treatment of pulsed ultrasound in 6W. We also imaged the movement of primary cilia, which showed corresponding movement when subjected to pulsed ultrasound. In Vivo Response: LIPUS treatment significantly increased BVF compared to OVX controls for the 100mW/cm^2 treated group. Interestingly, the 100mW/cm^2 treated groups showed a significant improvement over the 5mW/cm^2 treated group. Discussion and Conclusions: Pulsatile focused ultrasound can create local fluid flow nearby cells. The observed primary cilia can be triggered

to dynamic movement by the acoustic force as a mechanobiologic effect. In vivo results suggest that low-intensity pulse ultrasound can induced mechanical wave in tissue and initiate bone adaptation. These findings support the hypothesis that LIPUS can inhibit bone loss and preserve bone strength under conditions of estrogen deficient osteopenia. Keywords: quantitative ultrasound, therapeutic ultrasound, bone mechanotransduction, osteoporosis, bone remodeling

Contributed Papers

3:00

4pBA4. Effect of trabecular material property on ultrasonic backscattering in cancellous bone. Chengcheng Liu, Dean Ta, and Weiqi Wang (Department of Electronic Engineering, Fudan University, Shanghai 200433, China, estonelau@163.com)

Ultrasonic backscattering technique, used to assess cancellous bone status, is investigated using numerical simulations based on two-dimensional finite-difference time-domain (FDTD) method. High resolution microstructure mappings of bovine cancellous bone, provided by micro-CT, are used as the input geometry for simulations. This paper focused on the effects of material property parameters (density, lamé coefficients, viscosities, and resistance coefficients) of the trabecular on ultrasonic backscattering measurements at 1MHz. Simulations of ultrasonic backscattering in cancellous bone for different trabecular parameters were carried out and the backscatter coefficient (BSC) were measured and discussed. The results demonstrate that BSC is a nonlinear function of trabecular density and increases with trabecular density. While BSC is affected little by the viscosities, the first and second lamé coefficients have a complex effect on BSC. BSC is almost a linear function of normal resistance coefficient (NRC) and decreases with increases of NRC. On the other hand, BSC is practically independent of the shear resistance coefficient, just because there is little shear wave in backscattered signals. The results demonstrated that ultrasonic backscattering is affected by trabecular density and other material acoustic properties, as well as the bone mineral density and microarchitecture.

3:20

4pBA5. Correlation of ultrasonic backscatter parameters with transmission parameters and BMD in cancellous bone. Haijie Han and Dean Ta (Department of Electronic Engineering, Fudan University, No. 220, Handan Rd, Yangpu District, Shanghai 200433, China, haijie861017@126.com)

Quantitative ultrasound (QUS) has been suggested to have a performance equal to dual-energy X-ray absorptiometry (DXA) for the assessment of bone. In this paper, human cancellous bone is investigated in vivo using QUS and DXA. The ultrasonic backscatter method and its parameters (spectral centroid shift (SCS), apparent integrated backscatter (AIB)) are also introduced. The experimental ultrasonic backscatter signals are collected from 300 volunteers' calcanea in two hospitals, and the two parameters are calculated. Finally, correlation between ultrasonic backscatter parameters and transmission parameters (speed of sound (SOS), broadband ultrasonic attenuation (BUA) and stiffness index (SI)), as well as correlation between backscatter parameters and bone material density (BMD) are analyzed. The results showed that correlations between backscatter parameters and SI are better than that of SOS and BUA. SI correlates positively with SCS ($r=0.613$, $p<0.05$), as well with AIB ($r=0.463$, $p<0.05$). According to the size of SCS and AIB of ultrasonic backscattered signals, the status of cancellous bone may be assessed.

Invited Paper

3:40

4pBA6. Effects of boundary conditions on the two wave phenomenon in cancellous bone. Mami Matsukawa, Katsunori Mizuno (Doshisha University, Kyotanabe, 610-0321, Japan, mmatsuka@mail.doshisha.ac.jp), and Yoshiki Nagatani (Kobe City College of Technology, Kobe, 651-2194, Japan)

After the successful observation of two wave phenomenon in cancellous bone, wave characteristics have been investigated by in vitro studies. However, there still exists strong necessity to understand the effects of bone boundary conditions on the phenomenon, because the cancellous bone is always covered by a cortical layer in vivo. This paper is dedicated to the experimental and simulation studies of the effects on the two wave phenomenon. A sample of cancellous bone ($76.1 \times 45.2 \times 12.9$ mm^3) and two cortical plates (thickness: 1.2 and 1.9mm) were obtained from the left radius of a racehorse. Longitudinal pulse waves around 1MHz were measured by a conventional ultrasonic pulse technique using PVDF transducers. With 3-D CT image of the sample, wave propagation was also investigated by a 3-D elastic FDTD method. We then compared wave propagation in the sample covered with cortical layers or not. In both experimental and simulation approaches, two wave phenomenon was observed in the covered sample. However, the slow wave amplitude was very sensitive to the interface conditions between the cancellous bone and cortical layers. In addition, the waves often became small due to the conditions, telling the importance of sensitive wave observation in some cases.

4:00–4:20 Break

Contributed Papers

4:20

4pBA7. Bayesian-derived fast and slow waves correlate with porosity obtained from microCT. Joseph Hoffman, Amber Nelson, Mark Holland, and James Miller (Washington University in St. Louis, hoffman@wustl.edu)

It has previously been shown by our laboratory that Bayesian probability theory permits separation of ultrasonic fast and slow waves in cancellous bone even when the modes overlap substantially in time. The goal of the current study was to determine whether the fast and slow waves obtained from Bayesian separation of an apparently single mode signal individually correlate with porosity. The Bayesian technique was applied to data from cancellous bone samples from 8 human heels insonified with a broadband 500 kHz ultrasound pulse in the medial/lateral direction. The phase velocity (SOS), slope of attenuation (nBUA), and relative amplitude were determined for both the fast and slow waves. The porosity of the samples was measured by X-ray microCT. The phase velocity and slope of attenuation for both the fast and slow wave modes showed an inverse correlation with porosity. The fast wave amplitude decreased with increasing porosity. Conversely, the slow wave amplitude increased with increasing porosity. These results indicate that the properties of the individual fast and slow waves correlate with bone porosity, an important determinant of fracture risk. Supported, in part, by NIH/NIAMS grants R01AR057433 and P30AR057235.

4:40

4pBA8. Ultrasonic properties of fast and slow longitudinal waves propagating in the cancellous bone. Fuminori Fujita, Keisuke Yamashita, Katsunori Mizuno, Mami Matsukawa, and Takahiko Otani (Doshisha University, 1-3 Tatara Miyakodani, Kyotanabe City, Kyoto, Japan, bmi1009@mail4.doshisha.ne.jp)

Longitudinal wave in cancellous bone separates into fast and slow waves depending on the alignment of trabeculae. Here, the fast wave mainly propagates in trabeculae, whereas slow wave propagates in the soft tissue (bone marrow). Because these two waves usually overlap, the evaluation of each wave has still remained difficult. In this study, then, we have tried to evaluate the wave properties (attenuation and velocity), making use of the plane wave in an acoustic tube. 3-D image of the bone specimen was obtained by X-ray micro CT. In an acoustic tube, a cancellous bone specimen was set between home-made PVDF transducers. A function generator delivered a single sinusoidal signal in the range of 0.5 to 1.5 MHz to the transmitter. By filing a part of bone specimen away, we have tried to obtain the attenuation and velocity of the specimen. The two wave phenomenon clearly occurred in our specimen. In some trabecular bones of big animals, the wave separated perfectly. We have then tried to obtain ultrasonic properties of both waves and investigated the reflection and transmission at the interfaces.

5:00

4pBA9. Modeling ultrasound interaction with cancellous bone: investigation on the nature of the two compressional waves. Fabien Mézière, Marie Muller, Emmanuel Bossy, and Arnaud Derode (Institut Langevin, ESPCI ParisTech/CNRS UMR 7587/Université Paris 7/INSERM ERL U979 10 rue Vauquelin, 75005 Paris, France, fabien.meziere@espci.fr)

Although ultrasound might be useful to assess bone quality, the mechanisms of ultrasound propagation in trabecular bone are still poorly understood. For example the propagation of a short pulse that leads, under some conditions, to two transmitted longitudinal waves, a fast and a slow one, is not well explained yet. The objective of this work is to further investigate the nature of these two longitudinal waves in simplified bone model media. The approach is to determine if the fast wave could result from a guided propagation in the solid matrix, the slow wave being due to the propagation in the surrounding fluid (bone marrow). In this context, we study the propagation of the coherent waves through simplified and customizable binary structures, obtained by a random addition of scatterers, with characteristic dimensions, material properties and anisotropy similar to those of bone. The benefit of such simplified structures is that ultrasound propagation in the entire medium can be theoretically studied from the properties of a single scatterer.

5:20

4pBA10. Modeling and simulation of acoustic scattering in poroelastic materials. M. Yvonne Ou (408 Ewing Hall, University of Delaware, Newark, DE 19716, mou@math.udel.edu)

In this talk, we will present a model based on Biot's equation and numerical simulation of wave propagation in a fluid-elastic-poroelastic system, with scattering of a bone sample of cancellous bone surrounded by cortical layer and muscle layer in mind. The numerical methodology is based on the finite volume method and the operator splitting technique, combined with the grid-mappings technique and CLAWPACK. This is joint work with Randall LeVeque and Grady Lemoine form University of Washington.

5:40

4pBA11. Direct comparison of single mode versus fast and slow wave modes analyses of calcaneal bone data. Amber Nelson, Joseph Hoffman, Mark Holland, and James Miller (Washington University in St.Louis, Physics Dept., 1 Brookings Dr., St.Louis, MO 63130, nelsonam@wustl.edu)

Background: We previously demonstrated that a signal transmitted through cancellous bone might be comprised of two interfering (fast and slow wave) modes even though it appears to consist of only a single mode. Objective: The goal of this study was to compare the results of single mode analysis and two mode analysis to quantify the effects of interfering waves on the measured speed of sound (SOS) and broadband ultrasound attenuation (BUA). Methods: A series of human calcaneal samples (bone volume fractions = 0.09 to 0.21) were measured medial-laterally using 500kHz broadband focused transducers. Phase velocity was determined by phase spectroscopy and attenuation was determined by log-spectral subtraction. The original radiofrequency signal and the Bayesian-separated fast and slow wave radiofrequency signals were analyzed. Results: For each of the specimens, the slope of attenuation determined by single mode analysis was larger than that for either the fast or slow wave, and the speed of sound determined by single mode analysis lay between that of the fast and slow wave. Conclusion: The additional information provided by analyzing individually fast and slow wave modes might enhance the effectiveness of bone sonometry. Supported by NIH/NIAMS grants R01AR057433 and P30AR057235.

6:00

4pBA12. Bone ultrasound transducer. Masahiro Okino, Katsunori Mizuno, Daisuke Suga, and Mami Matsukawa (Doshisha University, Kyoto, Japan, dml0126@mail4.doshisha.ac.jp)

Low-intensity pulsed ultrasound (LIPUS) is used on bone-healing. One expected healing mechanism of this technique is the contribution of piezoelectricity. Actually, the piezoelectricity in dry bone at low frequencies has been reported by many investigators. However, the ultrasonic investigation of the piezoelectricity in bone has still remained very few. We have then made original transducers with bovine cortical bone samples, which were fabricated into plates (diameter: 10 mm, thickness: 0.5~1.0 mm). Using a conventional ultrasonic pulse technique in MHz range, we confirmed the observation of ultrasonic wave by the transducer. Here, the bone transducer was set in water at the focal position of a PVDF transmitter, which was excited by a single sinusoidal pulse of 70 Vp-p in the MHz range. The maximum voltage of the received wave was about 200 μVp-p. It was almost 1/2000 of the homemade PVDF transducer, which was made by the same procedure. The observed wave amplitude changed due to the sound wave frequency and bone thickness. Considering the inhomogeneous bone microstructure and the thickness dependence of the bone transducer, the piezoelectricity of bone in the MHz range is discussed.

Session 4pEAa

Engineering Acoustics and Physical Acoustics: Civil Non-Destructive Testing with Ultrasound or Other Non-Contact Methods I

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Invited Papers

2:00

4pEAa1. New designs of air-coupled ultrasonic transducers using micro-stereolithography. David Hutchins, Duncan Billson, Simon Leigh, and Chris Pursell (School of Engineering, University of Warwick, Coventry CV4 7AL, UK, D.A.Hutchins@warwick.ac.uk)

Micro-stereolithography, a form of high resolution rapid prototyping, has been used to design and construct novel designs of ultrasonic transducers for use in air. This approach allows for designs to be available that are difficult to produce by other means. The work will describe experiments that have been conducted with capacitive devices in two configurations: planar devices that are based on capacitive micromachined ultrasonic transducers (CMUTs), and those with a spiral geometry backplate for the generation of modified wavefronts. In addition, the fabrication technique has also been used to make structurally-modified materials (so-called metamaterials) to change the emitted characteristics in air. Examples are given of the type of applications for which these ultrasonic transducers could possibly be used.

2:20

4pEAa2. Air-coupled sensing of leaky rayleigh waves and ZGV modes in concrete slabs. Jinying Zhu (The University of Texas Austin, 1 University Sta., Austin, TX 78712, jyzhu@mail.utexas.edu), Yi-Te Tsai, Xiaowei Dai, and Michael R. Haberman (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, Texas 78712-1024)

Rayleigh waves and zero-group velocity (ZGV) Lamb modes (the impact-echo test) are commonly used for non-destructive evaluation (NDE) of concrete structures to extract information about material property and interior delaminations. Traditional methods in civil engineering employ point impactors to generate the waves while accelerometers measure the resulting out-of-plane motion. Though well-established, this methodology does not lend itself to efficient scanning of large areas, which is critical for monitoring the safety of infrastructure. The ability to detect these two wave types through in-air sensing of Leaky Rayleigh or ZGV Lamb waves can greatly accelerate NDE for large structures. Unfortunately, air-coupled sensing suffers from significantly decreased signal amplitude. This paper presents theoretical analysis and experimental validation efforts to amplify in-air signals resulting from Leaky Rayleigh waves and ZGV Lamb modes using a parabolic focusing mirror. A time-domain impulse response to the Kirchhoff-Helmholtz integral equation is presented that permits the analysis of discrete arrivals from each wave type and estimation of the expected focusing gain and depth of field based on the geometry of the parabolic dish. Analytical and numerical solutions in time-domain are compared with time-harmonic solutions taken from the literature and experimental results. This work is supported by NIST Technology Innovation Program (TIP).

2:40

4pEAa3. Toward a high power non-contact acoustic source using time reversal. Pierre-Yves Le Bas (Los Alamos National Laboratory, Geophysics group EES-17, MS D443, Los Alamos, NM 87545, pylb@lanl.gov), TJ Ulrich (Los Alamos National Laboratory, Geophysics group EES-17, MS D443, Los Alamos, NM 87545), Brian Anderson, and John Esplin (Department of Physics and Astronomy, Brigham Young University, N377 Eyring Science Center, Provo, UT 84601)

Over the last decades, nonlinear acoustic techniques have been developed to detect mechanical damage in solids. They have been proven to be far more sensitive to early damage than standard linear acoustic techniques. Unfortunately, they often require high amplitude waves to propagate within the sample. To be practical in industrial applications, signals have to be generated without contact. Currently available non-contact transducers are generally not powerful enough. A first step toward the creation of a high amplitude non-contact acoustic source will be described. This source is based on the principle of focusing energy on the surface of a sample using Time Reversal in a hollow cavity. By using a laser vibrometer for the necessary calibration of the system we are able to use a full non-contact process. New development in signal processing allows a much cleaner signal generation than usually achieve with time reversal. This source is proven to be much more energetic than current off the shelf non-contact transducers. It is a broad band source with an adjustable standoff distance and also has the capability to selectively generate in-plane waves. This work is supported by the U.S. Department of Energy through the LANL/LDRD Program.

3:00

4pEAa4. Non contact long distance exploration method for concrete using SLDV and LRAD. Tsuneyoshi Sugimoto, Ryo Akamatsu (Toin Univ. of Yokohama, tsugimot@cc.toin.ac.jp), Noriyuki Utagawa, and Syuichi Tsujino (Sato kougyo Co., Ltd.)

The hammering test is a representative method in inspection for cavities and delaminations at shallow area of concrete surface. Although this method is used widely because it is not expensive, efficiency of the defect-judging largely depends on the tester's experience and long measurement time is necessary for wide area inspection. Other methods have been developed, however, it is necessary to contact or approach to the inspection object during a measurement. Therefore, we propose a new non-contact acoustic imaging method for nondestructive inspection using scanning laser Doppler vibrometer (SLDV) and long range acoustic device (LRAD). In this method, Surface vibration, which is generated by air borne sound, is measured using SLDV. This time, the styrofoam board was buried at shallow depth in the concrete are used as a substitute of a cavity in the concrete. As an experimental result, a styrofoam board is clearly imaged by the vibration velocity of the concrete surface. Furthermore, we confirmed that our proposed method can apply even 10 m away distance, and the measurement distance is about within 20 m under the present conditions. It means that the non destructive inspection for concrete from a long distance is possible.

3:20

4pEAa5. Wavelength measurement of guided waves in plates with electromagnetic acoustic transducers. Guofu Zhai (Harbin Institute of Technology, P.O. Box 401, Harbin Institute of Technology, No. 92, West Dazhi Street, Harbin, China, gfzhai@hit.edu.cn), Tao Jiang, Jiapeng Gong, and Lei Kang (Harbin Institute of Technology, P.O. Box 404, Harbin Institute of Technology, No. 92, West Dazhi Street, Harbin, China)

For an EMAT (electromagnetic acoustic transducer) with a meander-line coil, the interval spacing between adjacent wires of the coil is generally designed to be the half wavelength of guided waves excited by the EMAT. However, the actual value of the wavelength deviates from the double interval spacing (double spacing) because of the variation spectrum of the spacing and the bandwidth of excitation frequency. So it is difficult to obtain the actual value of the wavelength of guided waves in plates. Based on dispersion equation and group velocity equation of guided waves, we find that the actual value is double spacing when the frequency of the excited signal is equal to the frequency of the received one; and the actual value deviates from the double spacing when the EMATs' operating points are not on the dispersion curves. Taking SH wave EMATs for instance, this paper proposes two wavelength measurement methods with different initial conditions for SH waves in plates. The validity of the proposed methods is verified by experiments.

3:40

4pEAa6. In-situ characterization of hedges with a parametric acoustic transducer. Kirill Horoshenkov, Amir Khan (School of Engineering, University of Bradford, Bradford, BD7 1DP, UK, k.horoshenkov@bradford.ac.uk), Hongseok Yang, Chris Cheal, Julija Smyrnowa, and Jian Kang (Sheffield School of Architecture, University of Sheffield, Arts Tower, Western Bank, Sheffield S10 2TN, Sheffield, UK)

This work reports on the results of outdoor measurements which were carried out using a highly directional parametric acoustic transducer and a 2-D array of microphones. The purpose of these measurements was to determine the frequency and angular dependence of the acoustic transmission loss of hedges (hedgerows). The experimental setup and procedure used in this work enabled us to simulate an approximately plane wave propagation regime and to reduce considerably the effect of ground on the recorded transmission loss data. The results show that a 1.5-2.5m thick hedge can provide a considerable (up to 20 dB) transmission loss which is comparable to that expected from an artificial noise barrier structure. The theory developed by Aylor (JASA 51(1), 197-205, 1972) is used to explain the observed dependencies as a function of the leaf density, size and hedge thickness. An alternative equivalent fluid theory for porous absorber (Horoshenkov et al, JASA 104(3), 1198-1209, 1998) is also used to explain the observed attenuation for sound propagation through the leaf mass.

4:00–4:20 Break

4:20

4pEAa7. Characterizing the properties of adhesive in bonded materials by laser ultrasonic method. Xiaodong Xu, Cong Liu, and Xiaojun Liu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, xdxu@nju.edu.cn)

In order to characterize the quality of adhesive in bonded materials, a metal/bonding/metal assembly model named as "sandwich" model is used to characterize the propagating properties of Lamb wave generated by laser ultrasonic method and studied in theory and experiment. In our experiment, a Nd:YAG ns laser is focused as line source and scanned on the surface of sample to supply signals in time and space, the signals of traveling lamb wave is detected by a beam deflection system. The dispersion curve of Lamb wave is deduced based on 2D FFT of obtained signals. According to the theoretical "sandwich" model, the real part and imaginary part, which is corresponded to the attenuation coefficients of sample, of elastic parameters of sample in wideband frequencies can be determined by theoretical fitting to the dispersion curve, and the determined parameters are reflected the changing properties of the adhesive in bonded materials.

4:40

4pEAa8. A quantitative imaging of bonding strength at the bonded solid-solid interface obtained by nonlinear test. Jianjun Chen (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, jjchen@nju.edu.cn)

As well known, when a longitudinal wave propagates through an interface with micro-cracks and micro-defects between solids, the contact acoustic nonlinearity (CAN) will be generated dramatically and the nonlinear parameter can be used to contour the bonding state of the interface. However the contour can only show the relative state of bonding strength and can not be used to judge weather the multilayered composite materials in use is safe because the safe judgment is not by the relative state while by the absolute value of the bonding strength. Therefore characterization of quantitative bonding strength at the interface is very important for judging a multilayered material in safe use. In this paper, how to get the quantitative bonding strength from the CAN parameter is studied. After the vibration amplitude of incident focusing wave at the bonded interface was calculated, the standard bonding strength with complete bonding state was established by tension test and CAN parameter is calibrated, the quantitative imaging of the bonding strength is obtained by CAN microscope in experiments. From the imaging, the positions with weak bonding strength could be easily located, which can be used to decide whether the material could be employed continuously.

5:00

4pEAa9. Nonlinear spring model for adhesive bonding interface. Zhiwu An (Beisihuanxi Road 21, Haidian District, Beijing, China, anzhiwu_111@163.com), Xiaomin Wang, Mingxi Deng, Jie Mao, and Mingxuan Li

The boundary conditions for an interface between two solids are analyzed to model second harmonic generation in a thin elastic adhesive layer. The approximate boundary condition models termed as "nonlinear spring models" are rigorously developed using an asymptotic expansion of the exact solutions in the limit of small ratio of interface layer thickness to wavelength. The applicability of such boundary conditions is analyzed by comparison with exact solutions for ultrasonic wave transmission. Numerical calculation indicate that as the acoustic nonlinearity increases, the model tends to be more accurate, meanwhile the second-harmonic amplitude increases in direct ratio. The present nonlinear spring models may provide a potential to evaluate the nonlinear mechanical behavior of bonding interface. Acknowledgment: This work is supported by the National Natural Science Foundation of China (Grant No.10834009)

5:20

4pEAa10. Noncontact MASW methods for near surface soil/infrastructure assessment. Zhiqu Lu (National Center for Physical Acoustics, The University of Mississippi, 1 Coliseum Dr. University, MS 38677, zhiqulu@olemiss.edu)

In near surface geophysics, a multi-channel analysis of surface wave (MASW) method has been increasingly applied for underground infrastructure assessments, in which conventional contact sensors like geophones and accelerometers were mostly employed to detect surface vibrations. In this study, noncontact sensors technology such as a laser Doppler vibrometer (LDV) and a microphone were used to measure Rayleigh wave and leaky Rayleigh waves respectively. These noncontact sensors were installed in a scanning platform that was driven by a stepper motor. The scanning MASW system consisted of two excitation sources: an electromechanical shaker and steel-balls to generate frequency sweeping (chirp) signals with frequency from 30 Hz to 500 Hz and high frequency (up to 40 kHz) impulsive signals respectively. The LDV-shaker-MASW was developed for near surface soil profile exploration and the microphone-steel ball-MASW was built for pavement assessment. The details of the system and several case studies will be addressed.

5:40

4pEAa11. Investigation of simultaneous signal transmission in non-destructive inspection of steel billet. Yoko Norose, Koichi Mizutani, and Naoto Wakatsuki (University of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8577, norose@aclab.esys.tsukuba.ac.jp)

A non-destructive method of steel billet, which uses time-of-flight of ultrasonic longitudinal wave and detects defects as decrement of pseudo sound velocity reconstructed by computerized tomography, has proposed. The remaining problem is long measurement time, owing to step-by-step measurement corresponds to each sound path whose number becomes huge. One of the solutions is simultaneous measurement, which can be achieved by transmitted signals simultaneously. To avoid interference among signals, choice of the transmitted signal becomes one of the most important points to be considered. In the case of non-destructive inspection of steel billet, many reflected waves from boundaries may cause adverse influence. Therefore, in this study, the parameters of the signal; modulation scheme, modulated signal, signal length, and signal frequency are investigated. Simulation-based study suggested that the signal is required to be short and independent each other. From those conditions, some experiments about defect detection by the several kinds of modulation schemes using pseudo-random noise were conducted. As a result, defects detection by 5ch simultaneous transmission was successful. It was suggested that the total measurement time could be decreased to 1/5.

6:00

4pEAa12. A grinding acoustic emission monitoring method using wavelet transforms and fuzzy logic. Yang Jing, Zhang Zhong-Ning, Cheng Jian-Chun (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, Jiangsu Province, China, yangj@nju.edu.cn), and Liu Xiang-Xiong (Hiecise Machines Co., Ltd, Kunshan 215337, Jiangsu Province, China)

It is known that the acoustic emission (AE) signals contain potentially valuable information for grinding wheel and work conditions during automatic grinding process. However, AE signals produced in the grinding zone are usually much more complicated and it is difficult to obtain enough effective information directly from raw AE signals. This paper presents an efficient grinding AE monitoring method based on a combination of wavelet transforms and fuzzy Logic techniques. First, the discrete wavelet transforms are used to extract the feature vectors from AE signals with different grinding wheel and work states. Secondly, a fuzzy classifier is employed to build a nonlinear relationship between the feature vectors of AE signals and the various grinding states. This method requires less computation and has a low sensitivity to the changes of grinding conditions. Experiment results show that the proposed intelligent monitoring system can identify the different grinding wheel and work states accurately and is valuable for industrial applications.

6:20

4pEAa13. Research on target-matching method in ultrasonic testing. Han Zhang, Zhiwu An, Xiaomin Wang, Jie Mao, and Mingxuan Li (Institute of Acoustics, Chinese Academy of Sciences, zhanghan81@126.com)

Ultrasonic echoes contain ample flaw information in nondestructive testing. However, bad coupling or small flaw often leads to low SNR (signal-to-noise ratio), so the result provided by conventional amplitude detection may not be accurate enough. To solve this problem, the finite element method is used to investigate the frequency domain characteristics of flaw echoes. Furthermore, a target-matching method based on the maximum output SNR criteria is proposed. The transducer is excited by a certain electrical signal obtained by adaptive filtering deconvolution algorithm for each flaw, so we get a maximum SNR when the excitation signal and flaw match. This is further verified in ultrasonic test for a series of samples containing different sizes of flat-bottom holes. This work is supported by the National Natural Science Foundation of China (Grant Nos. 10834009, 11074272)

Session 4pEAb

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

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Contributed Papers

2:00

4pEAb1. A passive dispersive wave amplifier for high-intensity broadband acoustic pulses. Steven Dion, Martin Brouillette, and Louis-Philippe Riel (Université de Sherbrooke, 2500 boul. de l'Université, Sherbrooke (Québec), Canada J1K 2R1, steven.dion@usherbrooke.ca)

The acoustical power output of piezoelectric ultrasonic transducer is limited by the material breakdown voltage or the available driving electrical power. While there are well known ways to passively amplify monofrequency acoustic waves generated by a single transducer, e.g., with an exponential horn, there is no obvious way to similarly pump energy into a structure to produce high-intensity broadband acoustic pulses. It was found that the frequency dependant phase velocity inherent to dispersive waveguides can be advantageously exploited to generate high intensity planar pulse waves using a single transducer. With this amplification concept, gain factors as high as 15 have been measured, which can be exploited to produce shock waves in water with a conventional ultrasonic transducer and low power electronics. The paper will present the theoretical underpinnings of this method, as well as its experimental validation. Some potential biomedical applications of this technology will also be discussed.

2:20

4pEAb2. Instantaneous tracking of a monopole sound source using algebraic localization method and circular array. Tsukassa Levy and Shigeru Ando (Dept. of Information Physics and Computing, University of Tokyo, Tokyo Bunkyou-ku Hongou 7-3-1, levy.t@alab.t.u-tokyo.ac.jp)

The purpose of this study is to localize instantaneously and to track sound source using circular array. Localization is done using a novel explicit inversion formula of direction and distance of a monopole source [S. Ando, ASA Seattle Meeting, 2011]. This method consists in applying the weighted integral method (IEEE Trans. SP, 57, 9, 2009, Inverse Problems, 26, 015011, 2010) on a partial differential equation (PDE) satisfied by the sound wave. In this study, we consider the location-constraint PDE (ASA/ASJ meeting, 2006) that describes the unilateral propagation of wide-band waves from a single source. We also take into account reverberation in this study by using a simplistic model of room reverberation. In the experiments, loci of instantaneous localization results are displayed on a motion image captured by a TV camera mounted on the 16-element circular array. Several experimental results are shown to demonstrate time-resolved localization capability of a multiply reflected source. Some theoretical and experimental comparisons with and without reverberation will be given.

2:40

4pEAb3. A mutual radiation impendence measurement method for 2-element transducer array considering nonlinear interaction. Xuesen Zhang, Peifeng Ji, and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, zhangxuesen@mail.ioa.ac.cn)

Mutual radiation impedances represent the acoustical interaction between transducers in an underwater array, which are inherent in the distance between transducers and the geometry of the array. The influence of mutual radiation impedances, which are important to evaluate the performance of the underwater array, especially needs to be intensively studied on condition that the power of transducers is high enough because of nonlinear interaction. A method for determining mutual radiation impedances of transducers in a 2-element array is proposed in this paper, which applies to estimation of mutual radiation impedances for an underwater array with high radiation power. This method derives from V method in which the mutual radiation impedances are considered as the acousto-motive force generated on the surface of each transducer. By using nonlinear acoustic analysis, theoretical foundations of this method based on equivalent circuits are described. Detailed experimental procedures to implement the method are illustrated.

3:00

4pEAb4. Mass sensing using a functionalized subordinate oscillator array. Joseph Vignola, John Judge, Aldo Glean, and Teresa Ryan (The Catholic University of America, 620 Michigan Ave, NE, Washington DC 20064, vignola@cua.edu)

Vibrometric based mass sensing for detection of trace levels of chemical vapors has been under investigation for a number of years. One implementation of such a sensor uses a vibrating MEMS cantilever that has been functionalized to bind with a specific chemical compound. The additional mass reduces the resonance frequency of the structure by an amount that can be related to the concentration of the compound. This work describes an analytic and numerical study of an array of differently functionalized cantilevers coupled to a primary structure. The dimensions of individual cantilevers are chosen such that there is a distribution of their isolated natural frequencies and the mass of the primary structure is substantially larger than the collective mass of the subordinate cantilevers. The isolated natural frequencies of the cantilevers must be packed around that of the primary structure such that the response curves of neighboring cantilevers cross near their half power points. This overlap ensures an exchange of energy between array elements. Results describing optimized distributions of cantilever

properties will be presented. Both fabrication variations and the sensed mass are considered as disorder in the system. This presentation reports how the relative magnitude of these disorders affects mass declarations.

3:20

4pEAb5. Performance of optimized sound field control techniques in simulated and real acoustic environments. Philip Coleman (CVSSP, University of Surrey, Guildford, Surrey GU2 7XH, United Kingdom, p.d.coleman@surrey.ac.uk), Martin Møller, Martin Olsen (Bang & Olufsen a/s, Peter Bangs Vej 15, DK-7600, Struer, Denmark), Marek Olik, Philip Jackson (CVSSP, University of Surrey, Guildford, Surrey GU2 7XH, United Kingdom), and Jan Abildgaard Pedersen (Bang & Olufsen a/s, Peter Bangs Vej 15, DK-7600, Struer, Denmark)

It is of interest to create regions of increased and reduced sound pressure ('sound zones') in an enclosure such that different audio programs can be simultaneously delivered over loudspeakers, thus allowing listeners sharing a space to receive independent audio without physical barriers or headphones. Where previous comparisons of sound zoning techniques exist, they have been conducted under favorable acoustic conditions, utilizing simulations based on theoretical transfer functions or anechoic measurements. Outside of these highly specified and controlled environments, real-world factors including reflections, measurement errors, matrix conditioning and practical filter design degrade the realizable performance. This study compares the performance of sound zoning techniques when applied to create two sound zones in simulated and real acoustic environments. In order to compare multiple methods in a common framework without unduly hindering performance, an optimization procedure for each method is first used to select the best loudspeaker positions in terms of robustness, efficiency and the acoustic contrast deliverable to both zones. The characteristics of each control technique are then studied, noting the contrast and the impact of acoustic conditions on performance.

3:40

4pEAb6. On frequency characteristics of bone conduction actuators—subjective and objective measurements of inner canal type and head-of-mandible type actuators. Yoshimi Fukuda, Yuki Yae, Fumihiro Nakahara (Kumamoto University, 2-39 Kurokami, Kumamoto City, Kumamoto, Japan, yoshimi@hicc.cskumamoto-u.ac.jp), Hidenori Nakatani (Goldendance Co., Ltd., 3-22-19 Furuichi, Jyoto-ku, Osaka, Japan), Koji Kobayashi (Toyota Tsusho Corporation, 4-9-8 Meieki, Nagoya, Aichi, Japan), Yoshifumi Chisaki, and Tsuyoshi Usagawa (Kumamoto University, 2-39 Kurokami, Kumamoto City, Kumamoto, Japan)

Recently, bone conduction actuators which can reproduce the high frequency signal around 10 kHz are released in the market and those are used not only for speech but also music reproduction. However, the detail characteristics of bone conduction actuators, such as threshold, loudness vibration and frequency characteristics are not yet clear. Because there are difficulties to make clear the relationship between input signal and loudness of perceived sound even if acceleration of actuator is given. In this paper, threshold and loudness of stimuli are measured by means of psychoacoustical test for two types of bone conduction actuators; inner canal type and head-of-mandible type. Frequency characteristics of actuators are measured as acceleration. Comparison between subjective and objective measurements is performed in order to make clear the characteristics of measured bone conduction actuators.

4:00–4:20 Break

4:20

4pEAb7. A micro-machined microphone based on field-effect-transistor and electrets. Kumjae Shin, Yub Je (Department of Mechanical Engineering, Pohang University of Science and Technology, PIRO 416 Hyoja-Dong, Nam-Gu, Pohang, Gyungbuk, 790-784, South Korea, forhim13@postech.ac.kr), Haksue Lee (Agency for Defence Development, P.O. Box 18, Jinhae, Changwon, Republic of Korea), James E. West (Department of Electrical & Computer Engineering, Johns Hopkins University, 3400 N. Charles St., Baltimore, MD 21218), and Wonkyu Moon (Department of Mechanical Engineering, Pohang University of Science and Technology, PIRO 405 Hyoja-Dong, Nam-Gu, Pohang, Gyungbuk, 790-784, South Korea)

Micro-machined microphones are attaching attention of industry because of their benefit of size over conventional ones. Since most of micro-machined microphones are capacitive sensors, the sizes of their electrodes determine the low frequency noise level that increases with inverse of frequency (1/f). Therefore, the size of microphone itself becomes larger than the one that can be fabricated. Here we introduce a micro-machined microphone that can overcome the limit of capacitive microphones. The proposed microphone is composed of a field-effect-transistor (FET) and an electret. The difference between the conventional electret capacitive microphones and the proposed microphone may be the transduction mechanism: the change in the position of an electret causes the change in electric field on the gate of FET. Compared with capacitive transduction, the resistive channel of FET can be designed to have low sensor impedance, and subsequntly have low impedance at low frequency. To make experimental specimen, FET onto membrane and electret were fabricated with conventional metal-oxide-semiconductor fabrication process and micromachining process, respectively. The FET membrane chip and the electret chip were assembled. Simple current to voltage converter was applied as a pre-amplifier. Its feasibility to apply low frequency acoustic sensor will be proved by simulation and experimental results.

4:40

4pEAb8. Study on improving piezoelectricity and PBLG/PMMA composite. Junsik Jeon, Yonghwan Hwang (Department of Mechanical Engineering, Pohang University of Science and Technology, PIRO 416 Hyoja-Dong, Nam-Gu, Pohang, Gyungbuk, 790-784, South Korea, jeonjs@postech.ac.kr), Micheal S Yu (Department of Materials Science & Engineering, Johns Hopkins University, 3400 N. Charles St., Baltimore, MD 21218), James E West (Department of Electrical & Computer Engineering, Johns Hopkins University, 3400 N. Charles St., Baltimore, MD 21218), and Wonkyu Moon (Department of Mechanical Engineering, Pohang University of Science and Technology, PIRO 405 Hyoja-Dong, Nam-Gu, Pohang, Gyungbuk, 790-784, South Korea)

Poly-Gamma-benzyl-L-glutamate Poly methyl methacrylate (PBLG-PMMA) composite polymer, a new piezoelectric transduction material, was invented by Yu et al. In our previous study, we have shown that the body composed of this new material can be produced by simultaneous poling and curing of a homogeneous solution comprising Poly-Gamma-benzyl-L-glutamate(PBLG) and methyl methacrylate(MMA) via detachable molding. And, it was also shown that the piezoelectric coefficient(d33) of PBLG-PMMA composite is dependent on the PBLG concentration and the intensity of electric field applied during curing in molds. In this study, we attempt to improve piezoelectricity of PBLG-PMMA composite because the piezoelectric coefficient of PBLG-PMMA was only about one twentieth of that of PVDF and measure the dynamic behaviors of PBLG-PMMA bodies in order to investigate the feasibility to apply this composite to acoustic transducers. In order to improve the piezoelectricity, the PBLG concentration and the electric field intensity applied during fabrication processes are increased. Our efforts result in making the piezoelectric coefficient about three times higher than it in previous study. We also measured the dynamic behaviors of a PBLG-PMMA disk. Then, a Tonpilz transducer was designed, fabricated using PBLG-PMMA composite disks, and tested experimentally. [Work supported by Grant No. ADDUD080004DD.]

5:00

4pEAb9. High Curie temperature relaxor piezoelectric single crystal broadband transducers. Jindong Xia (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing 100190, China, xiajindong@tom.com), Junbao Li (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing 100190, China), Guisheng Xu (Shanghai Institute of Ceramics, Chinese Academy of Sciences, No. 215, Chengbei Road, Jiading District, Shanghai 201800), Jianxin Xing, Gaolei Zhao, and Zhe Chen (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing 100190, China)

Lead indium niobate-lead magnesium niobate-lead titanate (PIMNT) crystals have been used to fabricate longitudinal vibrator broadband tonpilz transducers as active piezoelectric material. Compared to lead magnesium niobate-lead titanate (PMNT), rhombohedral-to-tetragonal phase transition temperature (Trt) of PIMNT increases to higher than 120° C and coercive field (Ec) can reach 5kV/cm, while possessing the similar electromechanical properties. The PIMNT single crystal transducer can work under high temperature exceeding 80° C and large drive voltage over 0.6MVpp/m with a

DC bias field of 0.27MV/m.Simultaneously, the device shows a smaller size, as much as 10dB higher transmitting voltage response and 8dB higher receiving voltage response than lead zirconate titanate (PZT) longtitudinal vibrator(tonpliz) transducer with a broad bandwidth over 150%. The PIMNT device demonstrates tremendous performance for sonar and ultrasonic technology.

5:20

4pEAb10. The study on vibratory tactile sensor using piezoelectric bimorph resonator. Subaru Kudo (Department of Information Technology and Electronics, Faculty of Science and Engineering, Ishinomaki Senshu University, 1 ShinnMito Minamisakai, Ishinomaki-shi, Miyagi 986-8580, Japan, kudou@isenshu-u.ac.jp)

Piezoelectric vibratory tactile sensors are used for measuring the softness and hardness of an object. In this work, a new construction of vibratory tactile sensor was investigated using a piezoelectric bimorph resonator. The bimorph resonator was driven in one side by constant voltage and constant driving frequency. When the tip of a resonator was contacted with a test piece, the output voltage in the other side of resonator was changed by contact impedance. Then, the contact impedances were calculated with experimental characteristics of output voltage using the equivalent circuit of the bimorph resonator. It was experimentally clarified that the electrical contact capacitance gradually decreased as the load added to test piece increased. The amount of decrease for hard test piece was larger than those of the soft test pieces. Next, the Young's moduli of test pieces were calculated from the experimental results in this method. The calculated values were compared with actual material constants of test pieces. It was examined that the possibility for detecting the softness and hardness of an object by this method on bimorph resonator. These results in this study may be useful for designing on the piezoelectric tactile sensor.

5:40

4pEAb11. Theoretical approach on SAW characteristics of layered structures for gas sensing. Xiao Xie (Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, xiexiao08@mails.gucas.ac.cn), Wen Wang, Minghua Liu, and Shitang He

Integrating the self-assemble sensing film with SAW devices has advantage for organic gas detection. It generally requires a thin metal layer like Au acting as a catalytic base for the sensing film formation. This leads to a double layer SAW device sensing area fabrication consisting of a metal layer deposited on the piezoelectric substrate. Therefore, the SAW characteristics will be affected when propagating through the sensing area. Various layer parameters such as stiffness constants and the thickness are needed for consideration. In this paper, the SAW phase velocities have been calculated theoretically for ST-X Quartz/Gold structure under different normalized thickness configurations.

6:00

4pEAb12. Deposition of ZnO films with c-axes lying in R-sapphire substrate planes. Yan Wang, Shu-yi Zhang (Laboratory of Modern Acoustics, Institute of Acoustics, Nanjing University, Nanjing 210093, China, wangyan8008@126.com), Kiyotaka Wasa (Department of Micro-Engineering, Kyoto University, Kyoto 606-8501, Japan), and Xiu-ji Shui (Laboratory of Modern Acoustics, Institute of Acoustics, Nanjing University, Nanjing 210093, China)

The (110) textured ZnO films with c-axes lying in R-sapphire substrate planes are deposited by RF magnetron sputtering. The focusing investigation is the effect of substrate position in the sputtering on structural and acoustic properties of the ZnO films on the sapphire. The crystallographic characteristics of the films are characterized by X-ray diffraction (XRD) analysis. It is found that the crystalline orientation of the ZnO film varies with the substrate position deflected from the center of the target. The XRD spectra of the films show that there is an optimized substrate position, at which the strongest (110) peak of the ZnO film can be obtained. However, with the decrease or further increase of the deflection, the (110) diffraction intensity decreases and additional diffraction peaks of (002), (102) and (103) emerge. In order to investigate the variations of acoustic properties of these films, the ZnO piezoelectric films are deposited on R-sapphire substrates with different deflection positions, and the bilayered structures are used to fabricate film bulk acoustic resonators (FBARs). The results show that the electromechanical coupling coefficients k_{15} of the shear mode acoustic waves excited by FBARs also vary with the different positions similar to that shown by the XRD spectra.

6:20

4pEAb13. Optimal design of surface wave emats for enhancing their ultrasonic signal strength. Lei Kang, Shujuan Wang, Zhichao Li, and Guofu Zhai (Harbin Institute of Technology, P.O. Box 404, No. 92, West Da-Zhi Street, Harbin, Heilongjiang 150001, P.R. China, victorkang11@126.com)

The strength of the ultrasonic signal transmitted by electromagnetic acoustic transducers (EMATs) is very weak, which severely confines the further expansion of the application scope of EMATs. To solve this problem, the features of the transmission process of a surface wave EMAT are studied based on a 3-D model. Simulation reveals that performances of the transducer are significantly affected by its parameters. Aiming at enhancing the strength of the ultrasonic signal, this paper investigates the influence of EMAT parameters on the ultrasound field of the transducer and accomplishes the optimal design of the EMAT by utilizing orthogonal test method. Experiments indicate that after optimization, the received signal of the EMAT has increased by 162.3%, which suggests the ultrasonic signal of EMATs can be effectively enhanced by utilizing orthogonal test method based on the 3-D EMAT model. Acknoweledgement: Supported by “the Fundamental Research Funds for Central Universities” (Grant No. HIT. NSRIF. 2012008)

Session 4pHT

Hot Topics: Aeroacoustics II

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Contributed Papers

2:00

4pHT1. A circular microphone array design approach for discrete noise suppression. Bo Yang, Jie Feng, and Ming Wen (The Third Research Institute of China Electronics Technology Group Corporation, Chaoyang District, Beijing 100015, China, byang@yahoo.cn)

A broadband detection performance of a circular microphone array has been degenerated by discrete noise interference. In this paper, a circular microphone array optimum design approach is proposed for discrete noise suppression in acoustic detection in the air. This approach utilizes the noise information of acoustic propagation's transfer function to mitigate its effect on broadband microphone array detection processing. It is demonstrated to be feasible to improve passive broadband detection ability of the circular microphone array through the adjustment of array shading weights. This is accomplished by numerically maximizing the deflection coefficient under the assumption of a small signal-to-noise ratio. Under this approach, the conventional beamformer is not redesigned, and only the shading weights of the conventional beamformer are adjusted.

2:20

4pHT2. Noise generation by a cylindrical cavity in subsonic flow. Olivier Marsden, Christophe Bogey, and Christophe Bailly (Laboratoire de M'ecanique des Fluides et d'Acoustique Ecole Centrale de Lyon & UMR CNRS 5509 69134 Ecully Cedex, France, olivier.marsden@ec-lyon.fr)

Following a previous experimental study of the noise generated by cylindrical cavities placed in subsonic flows, this work investigates mechanisms involved in the noise generation by a cylindrical cavity placed in a turbulent boundary layer, via numerical simulation. The cavity has a radius of r =5 cm and a depth of 10 cm. Flows are computed at Mach numbers of 0.2, 0.26 and 0.32, and the numerical boundary layer, of thickness 17 mm, is perturbed upstream of the cavity. Simulations are performed by solving the unsteady compressible Navier-Stokes equations using low-dispersion and low-dissipation finite-difference schemes. The LES approach is based on the explicit application of a selective filtering to the flow variables to take into account the dissipative effects of the subgrid scales. Numerical results are compared to the available experimental data presented in previous work, including mean flow aspects, flow statistics and acoustics. The numerical results are then examined in more detail in particular with regard to the origin of the well-defined tones observed in the acoustic far field. A simple feedback model coupling shear layer dynamics with depth mode acoustic resonance is shown to yield good frequency predictions.

2:40

4pHT3. A new acoustic target recognition method based on gaussian mixture model. Ming Wen, Jie Feng, and Bo Yang (The Third Research Institute of China Electronics Technology Group Corporation, Chaoyang District, Beijing 100015, China, wenming.wm@gmail.com)

Target recognition is one of the most important components of an acoustic detection system. Conventional acoustic target recognition algorithms base the classification results on features from a single frame of signal, suffering from inability to take the dynamic characteristics of acoustic signal into account, which can significantly increase the separability of acoustic targets. In this paper, we present a novel method for acoustic target recognition. Acoustic signal to be classified is divided into a sequence of signal segments. Gaussian mixture model is employed to model the sequence, and EM algorithm is utilized to learn the parameters. This idea is motivated by the interpretation that the Gaussian components can represent some general target-dependent spectral shapes. Experimental results show that the method achieves high recognition accuracy on different data sets, and is highly robust to ambient noise.

3:00

4pHT4. Aeroacoustics of flow merging at duct junction. Garret C. Y. Lam, S. K. Tang (Department of Building Services Engineering, The Hong Kong Polytechnic University Hung Hom, Kowloon, Hong Kong, P.R. China, garret.lam.hk@connect.polyu.hk), and Randolph C. K. Leung (Department of Mechanical Engineering, The Hong Kong Polytechnic University Hung Hom, Kowloon, Hong Kong, P.R. China)

Although merging flow at duct junction is always encountered in fluid-transporting systems, previous works were mainly devoted to study the acoustics of duct junction but the aeroacoustics of flow merging has received little attention. Therefore, this paper aims at revealing the mechanism of sound generation by a merging flow at a 30-degree duct junction. The flow problem is investigated numerically by solving the unsteady compressible Navier-Stokes equations and the gas equation of state simultaneously, thus allowing the acoustic field and the aerodynamic field to be determined without modelling the source terms in the wave equation. The Conservation Element and Solution Element (CE/SE) method is chosen as the solver, which has been successfully applied in tackling many aeroacoustic problems. The results show that a shear layer is created between the two flows at duct junction due to the velocity gradient across the flow. Furthermore, another shear layer is also formed at downstream to the edge of duct junction. However, only the latter one rolls up to form vortices, which are believed to be the acoustic source. The contributions of these flow dynamical processes to sound generation and the identification of dominant sources will be discussed.

3:20

4pHT5. Numerical research on the flow characteristics of confined sonic and supersonic air-modulated speakers. Zhao Yun, Zeng Xinwu, and Gong Changchao (Institute of Optical-Electronic Science and Engineering, National University of Defense Technology, Changsha 410073, P.R. China, sangelboy@sohu.com)

Air-modulated speaker is one of the most famous high-intensity sources. Sound waves with SPL above 160 dB are generated by the modulation of sonic air flow through a time-varying valve. To overcome the source saturation at high chamber pressures, the performance and flow characteristics of a new design, in which the flow speed increases from sonic to supersonic by a converging and diverging nozzle, were investigated. Based on the analytical results from the widely used quasi-steady model, the source level improvement was verified for the same pneumatic power. The SPL increment of more than 5 dB is obtained at Mach number 2.75. The transient flow inside the source was simulated using a recent developed numerical model in which the compressible flow is governed by the Navier-Stokes equation and the valve movement is modeled by the dynamic mesh method. Compared with the ordinary sonic flow modulation, the source mechanism on the supersonic case has shown several distinct features: the effect of the flow-acoustic coupling could not be neglected; some complex phenomena such as shock wave formation and propagation arise in the transient flow inside the nozzle; the establishment of supersonic flow and the resulting improvement are related to the valve modulation frequency.

3:40

4pHT6. On the adjoint problem in duct acoustics and its solution by the time domain wave packet method. Fang Hu, Ibrahim Kocaogul (Old Dominion University, Norfolk, VA 23529, fhu@odu.edu), and Xiaodong Li (Beihang University, Beijing 100191, China)

The Time Domain Wave Packet (TDWP) method has the advantage of obtaining radiated sound by a given duct mode at all frequencies in one computation. It also makes possible the separation of the acoustic and shear flow instability waves. In this paper, the TDWP method will be applied to the adjoint duct acoustics problem. As a microphone records the sound radiated by all duct modes combined, the adjoint approach has the advantage of obtaining the relative mode strengths of all the propagating modes at all frequencies in one computation. The novelty of the paper is on the formulation of adjoint equations in time domain and the numerical solution using the TDWP method. The theoretical formulation of the duct adjoint problem and derivation of various reciprocal relations are presented. The adjoint equations are then solved numerically in reversed time by the TDWP method, in which a point source in the far field is enforced with a Broadband Acoustic Test Pulse time function. PML absorbing boundary conditions for the adjoint equations are also discussed. By an FFT, the relative strengths of all duct modes at all frequencies to the far field point are computed at once.

4:00–4:20 Break

4:20

4pHT7. Preliminary analysis of acoustic measurement from ARJ21 flyover test. Zhifei Chen, Jianhua Yang, and Hong Hou (Northwestern Polytechnical University, 127 West Youyi Road, Xi'an, Shaanxi, China, chenzhifei@gmail.com)

A flight test was performed on the airport of Yanliang in June, 2010 for the noise source identiofication of ARJ21. The airframe noise was recorded using a ground-based multi-arm logarithmic spiral array with diameter as 30 meters. A data set of five taking off and eight approaching flight with extended landing gears has been acquired during four days. Conventional beamforming and deconvolution approach for the mapping of acoustic sources (DAMAS) are applied for the data analysis. The beamforming results could clearly distinguish two engines and the nose landing gear. Their spectrum characterizations are also presented together with a description of the experimental set-up.

4:40

4pHT8. A nodal discontinuous Galerkin method for computational aeroacoustics and comparison with finite difference schemes. Eryun Chen, Ailing Yang, and Gaiping Zhao (USST, chenеryun@usst.edu.cn)

A nodal discontinuous Galerkin formulation, which is based on Lagrange polynomials basis, is used to directly simulate the acoustic wave propagation. Two test problems of wave propagation with initial disturbance consisting of a Gaussian profile or rectangular pulse are investigated. We evaluate the performance of the schemes in short, intermediate, and long waves. Moreover, the comparisons of numerical results between the nodal discontinuous Galerkin method and finite difference type schemes are performed, which indicate that the numerical solution obtained using nodal discontinuous Galerkin method with a pure central flux has obvious high frequency oscillations for initial disturbance consisting of rectangular pulse, which is the same as those obtained using finite difference type schemes without artificial selective damping. If an upwind flux is adopted, spurious waves are eliminated effectively except for the location of discontinuities. When a limiter is used, obviously the spurious short waves are almost completely removed. The quality of the computed solution has greatly improved.

5:00

4pHT9. The enhancement of pulverized-coal combustion by using sound waves. Genshan Jiang, Yingchao Zheng (Department of Mathematics and Physics, North China Electric Power University, Baoding, Hebei Province 071003, P.R. China, gs_jiang@hotmail.com), Jie Pan (School of Mechanical Engineering, The University of Western Australia, Crawley, WA 6009, Australia), and Jing Tian (Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, P.R. China)

A model for the enhancement of pulverized coal combustion in high-intensity acoustic fields has been developed. In a power boiler pulverized coal particles and char are entrained in the main gas flow for a significant length of time. The slip velocity between the entrained particles and the gas is very low leading to low heat and mass transfer to and from the particles. This results in long combustion times for the particles, particularly if the combustion is diffusionally controlled. When a high-intensity acoustic field is applied to the gas flow, it superimposes high-velocity oscillations on the main flow. If the frequency and amplitude of this acoustic field is just right, the particles are not entrained in the acoustically induced flow oscillations. This creates a periodically varying slip velocity between the particles and the hot gas, leading to higher convective heat and mass transfer to and from the particles, thereby enhancing the combustion. The paper also presents the axial pressure gradient and shear stress in neighborhood of the particles placed in sound field.

5:20

4pHT10. CAA-based optimisation of a sensor array to characterize aft and forward fan noise in a 3D nacelle. D.-C. Mincu (Onera - Office National d'Etudes et de Recherches Aerospatiales, BP 72, 92322, Chatillon, France, daniel-ciprian.mincu@onera.fr), E. Manoha, and C. Polacsek

In the context of the on-going European project JTI-SFWA, the modal acoustic content inside a realistic fan duct, designed by Dassault-Aviation, must be characterized experimentally. The objective is to use Onera's CAA solver sAbrinA-V0 to design a non-intrusive sensor distribution inside the nacelle, upstream/downstream the fan plane, accounting for the heterogeneous mean flow and the complex internal geometry. By computing the propagation of several interacting modes in the duct, a region in which the acoustic pattern is close to the infinite duct analytical solution was characterized. Then several 3D distributions of pressure sensors were tested in this zone. Radial and azimuthal sensor arrays were located at the nacelle wall, whereas a radial sensor array was placed along the bifurcation leading edge, without additional intrusivity. The response of the 3D arrays to a given acoustic field, either analytical or propagated through CAA, was projected on an analytical modal basis. For several array configurations, this projection clearly showed the emergence of the modal injected content. From these simulations, the most efficient sensor distribution will be used to equip an experimental fan noise simulator, and the modal projection process will be applied on measurements to evaluate the actual modal content in the duct.

5:40

4pHT11. Trailing edge noise mitigation investigation for wind turbine blades. Michael J. Asheim, Dave Munoz (Colorado School of Mines, Golden, CO 80401, U.S.A., masheim@mines.edu), and Patrick Moriarty (NREL-NWTC, Golden, CO 80401, U.S.A.)

Wind turbines offer one of the most mature technologies for providing large scale renewable energy to society in an economically viable way. Although not on par with the price of conventional energy sources yet, the cost of energy has been steadily decreasing as the technology continues to develop. Unfortunately, like with all energy sources, there are some problems with this form of generation. Among these, sound emissions from wind turbines are one of the problems people who live close to the installed machines may be exposed to. Past studies show that these noise emissions are dominated by aeroacoustic noise and of the many mechanisms that lead

to aeroacoustic noise, the interaction between the unsteady flow and the trailing edge seems to constitute the largest portion of the overall sound spectrum. Modifications to the trailing edge geometry will change how the fluid interacts with the trailing edge and can be used to change the resulting noise emission. This study will investigate the effect passive trailing edge devices have on the overall noise emission from a wind turbine, in an attempt to reduce the aeroacoustic noise being generated by the turbine.

6:00

4pHT12. Aerodynamic noise prediction model of pantograph for high-speed train. Sukkeun Yi and Junhong Park (Hanyang University, 133-791, sarkkun@hanyang.ac.kr)

In this study, to analyze aerodynamic noise from the pantograph on high-speed train, numerical model was established. We study the mechanism of the noise generation by setting up the simple model and finally analyze entire pantograph model. Based on the Lattice Boltzmann Method, computational fluid analysis was carried out. Through noise analysis for near field, sound radiation for far field was estimated using Ffowcs Williams and Hawkings Equation. Simple model is the square and circular cross-sectional cylinder. This model makes noise by flow separation accompanied by vortex shedding. Considering pantograph have a series of the rods, we made a complex structure which have square and circular cross-sectional cylinder. By analyzing entire pantograph model we suggest the model for predicting the noise generation and radiation.

6:20

4pHT13. Prediction of aerodynamic sound radiation from pantograph of high-speed train. Sukkeun Yi and Junhong Park (Hanyang University, 133-791, sarkkun@hanyang.ac.kr)

Based on the Lattice Boltzmann method, aerodynamic noise from the pantograph of high-speed trains was investigated. The mechanism of the noise generation was analyzed using a simple panhead model, and the numerical procedures were extended to analyze sound radiation from the whole pantograph system. The square and circular cross-sectional cylinders were used to simulate the panheads. The primary noise was generated by flow separation accompanied by vortex shedding. Since the pantograph were consisted of a series of the rods, the primary noise generation characteristics were similar to the simple panhead. From the numerical model, the design of the pantograph for reduced aerodynamic noise generation is proposed and validated.

THURSDAY AFTERNOON, 17 MAY 2012 S228, 2:00 P.M. TO 3:00 P.M.

Session 4pMUa

Musical Acoustics and Psychological and Physiological Acoustics: Musical Timbre: Perception and Analysis/Synthesis II

James W. Beauchamp, Cochair
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Andrew B. Horner, Cochair
horner@cse.ust.hk

Contributed Papers

2:00

4pMUa1. Spatial variability of timbre for an electric guitar amplifier. Alexander Case (University of Massachusetts Lowell, 35 Wilder St., Lowell, MA 01854, alex_case@uml.edu), Agnieszka Roginska, and Jim Anderson (New York University, New York, NY 10012)

Impulse response measurements of an electric guitar amplifier at high spatial resolution reveal frequency response variabilities likely to drive key elements of timbre for the recording engineer. Data collection from the very near field of the driver to a distance of several feet away, gathered in a three-dimensional grid around the open-backed, single-driver, Fender Deluxe electric guitar amplifier quantify the variations in frequency response which may be exploited by the recording engineer using microphone placement to influence timbre.

2:20

4pMUa2. Synthesis of performance expression of bowed string instruments using "Expression Mark Functions". Yuma Koizumi and Katunobu Itou (Hosei University, 3-7-2 Kajino-cho Koganei, Tokyo, Japan, 08k1014@cis.k.hosei.ac.jp)

This paper proposes a method for synthesis of performance expression of bowed string instruments. In order to reflect a creator's intention on a synthetic sound, physical model is efficient because the sound can be synthesized by the performance imagination. The physics of the bowed string instruments are still some issues remain unresolved, bowed string sound synthesis using only the physical model is difficult. In this paper, a method using a transfer function for expression mark is proposed. "Expression mark functions" is estimated from recorded performance sounds, using spectrum of string motion and inverse filter of resonant properties of the instruments. Bowed String motion is a triangular wave called the Helmholtz wave, and it can be determined from bowed string position. Changes of waveform from expression performance were estimated by non-negative matrix factorization. Resonant properties of an instrument were measured as TSP response using a "direct conduction speaker". The expression mark functions of the performance is extracted by decomposing the actual performance. For evaluation of the quality of the synthesized sound, the scale by ten kinds of expression marks was synthesized using expression mark functions, and the questionnaire estimated the degree of agreement in five steps.

2:40

4pMUa3. Data collection for individuality analysis on subjective music similarity evaluation. Shota Kawabuchi, Chiyomi Miyajima, Norihide Kitaoka, and Kazuya Takeda (Nagoya University, Furo-cho Chikusa-ku, Nagoya 464-8603, Japan, shota.kawabuchi@g.sp.m.is.nagoya-u.ac.jp)

Recently there are many studies of subjective music similarity for music information retrieval. To quantify the subjective music similarity, there are many factors to take into account. In this study the individuality of the subjective similarity was focused on. To analyze the individuality of the subjective similarity, subjective evaluation data for 200 pairs of RWC popular music database was collected. 28 subjects listened to the pairs of tracks, and evaluated similarity of each pair by similar or dissimilar. They also selected the components of music (melody, tempo/rhythm, vocal, instruments) that was similar. Each subject evaluated the same 200 pairs, thus the individuality of the evaluation can be easily analyzed. The results of pilot analysis were reported.

Session 4pMUb

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

Thomas Moore, Chair
tmoore@rolling.edu

Contributed Papers

3:20

4pMUb1. Open-loop control of a robotized artificial mouth for brass instruments. Thomas Helie (IRCAM- CNRS UMR 9912, thomas.helie@ircam.fr), Nicolas Lopes, and René Causse (IRCAM - CNRS UMR 9912)

In order to have reproducible and controllable experiments, a robotized artificial mouth dedicated to brass instruments has been developed (CONSONNES project of the French National Research Agency). The actuators control: (A1) the airflow, (A2) the mouth position (monitoring the lips force applied to the mouthpiece), (A3-4) the water volume in each lip (water-filled latex chamber). The sensors measure: (S1-2) the pressure in the mouth and the mouthpiece, (S3: optical sensor) the area between the lips, (S4) the force lips/mouthpiece, (S5-6) the water pressure in each lip, and (S2bis-S4bis) the position of the moving coils (A2-4). We present an open-loop control obtained from measures, according to the following steps. First, a calibration for the lips control is performed, by analyzing signals (S4-6) w.r.t. positions (S2bis-S4bis), with no airflow. Second, slowly time-varying calibrated commands (A2-A4) are used to obtain quasi-stationary regimes (non oscillating, quasi-periodic, etc), for constant airflows. Third, a sound analysis is processed to generate cartographies of features (energy, fundamental frequency, etc) w.r.t. to the calibrated control parameters. This exploratory tool allows to build a dictionary of relevant control parameters from which basic sequences of notes can be played, and during which all the signals (S1-6) can be recorded.

3:40

4pMUb2. Minimum models for self-sustained oscillations of conical reed instruments. Jean Kergomard (LMA-CNRS 31 Chemin J. Aiguier, F-13402 Marseille Cedex 20, kergomard@lma.cnrs-mrs.fr), Philippe Guillemain, and Fabrice Silva (F-13402 Marseille Cedex 20)

It is now well known that a minimum model for self-sustained oscillations of clarinet-like instruments is the iterated map model, leading to square signals. The reed is assumed to be without dynamics, while losses are ignored (or assumed to be independent of frequency). The generalization to conical instruments is not straightforward. For the present work, the minimum model is used for a truncated cone instrument, but the missing part of the cone is not assumed to be small compared to the wavelength. Thus the result should be a signal without sharp corners. However, without any kind of mouthpiece, no periodic sound can be obtained in a steady-state regime. It will be explained that the choice of a model for the mouthpiece can be done without adding any supplementary parameter (therefore a conical resonator has one parameter only more than a cylindrical one). It is shown that several choices are possible, allowing either the use of: the same inverse nonlinear characteristic than for clarinet-like instruments, or any direct nonlinear characteristic, leading to a great simplicity. Advantages and drawbacks of several solutions are discussed.

4:00–4:20 Break

4:20

4pMUb3. Investigation of optimum shape for an acoustic guitar by electroanalysys of tone quality. Kazutaka Itako (Kanagawa Institute of Technology, 1030 Simo-Ogino, Atsugi, Kanagawa, Japan, itako@ele.kanagawa-it.ac.jp), and Satoshi Itako (B.K. Guitar Craft Center, 217-77 Mibutei, Mibumachi, Shimotugagun, Tochgi, Japan)

The shapes of acoustic guitars are strongly governed by the sensibilities of the craftsmen who make them, and thus, shapes vary widely. Unlike violins and other instruments, no ideal shape has yet been established for guitars. In our research, we conducted an electroanalysis of tone quality with the aim of manufacturing a guitar with enhanced tone quality. In this study, we examined the effects of varying guitar body thickness and chamber cubic volume on tone quality with various playing methods, and quantitatively identified the optimal thickness.

4:40

4pMUb4. Simulation of 'Suikinkutsu' sound considering sound radiation from water surface. Yuki Fujita, Naoto Wakatsuki, and Koichi Mizutani (University of Tsukuba, Tsukuba, Japan, fujita-y@aclab.esys.tsukuba.ac.jp)

This paper is about a 'Suikinkutsu', a kind of Japanese traditional musical instruments. Different from other musical instruments, 'Suikinkutsu' is composed of an upended jar buried underground in the garden. Dropping water into 'Suikinkutsu' works as sound source, and attracting harp-like sound inside the jar, 'Suikin-on', is generated. Although the sounding mechanism of 'Suikinkutsu' is interesting because there are complicated interactions among air, water, and jar made of ceramic to generate harp-like beautiful sound, its analysis has not been achieved yet. To investigate the sounding mechanism of 'Suikinkutsu', simulation of interaction between air and water is performed. The obtained results suggest that the water in the jar contributes to sound generation of 'Suikinkutsu'.

5:00

4pMUb5. Strike notes and their pitches of some kinds of bells. Shigeru Yoshikawa (Graduate School of Design, Kyushu University, 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, shig@design.kyushu-u.ac.jp), Wang-Ho Cho, and Takeshi Toi (Department of Precision Mechanics, Chuo University, 1-13-27 Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan)

The strike notes of western church bells have been studied very long. This paper considers the bells that are different from church bells: (1) simple tubular bells called "wind chime", (2) a downward-stretching tubular bell with hyperbolic surface, which the authors name "Gaudi bell" because Antoni Gaudi envisaged music of such bells coming from the belfries of the Sagrada Familia Church, and (3) a double Gaudi bell stretching downward and upward from the center with different lengths. Strike notes of these bells are yielded from the axial vibrations. However, their pitches do not necessarily correspond to the strongest spectrum (mode) of the bell vibrations. Aluminum tubular bells of our wind chime gave the "spectral pitch" determined by the spectral frequencies (around 1 kHz) of the fourth mode. On the other hand, our Gaudi bell (brass-made, 1.80 m, 50 kg) yielded the fourth mode at 750 Hz. However, many listeners perceived the pitch at 375 Hz. This is the "virtual pitch" based

on the missing fundamental (mode frequencies higher than the 4th were 1082, 1450, 1842, and 2258 Hz). A double Gaudi bell (bronze-made, 0.342 + 0.260 m) gave the "spectral pitch" at 1125 and 2046 Hz.

5:20

4pMUb6. Gabor domain analysis of mambranophones. Robert Ferguson (Department of Geoscience, University of Calgary, 2500 University Drive NW, Calgary, Alberta, Canada, rjfergus@ucalgary.ca)

Advanced analysis methods of exploration seismology are applied to the study of a class of membranophones that are well modelled as three spring damped oscillators. As a first attempt, damping is ignored in this treatment. Analysis using vibroseis-like source sweeps, cross-correlation, Fourier decomposition, and Gabor domain analysis provide insight into how this class is tuned for use in jazz and popular music. It is found that for a particularly good set of three example membranophones, they are tuned to pitches such that they resonate in combinations that are particularly harmonious. Combinations of octave, perfect fourth, and perfect fifth are found and, in particular, the Gabor domain is most useful for discrimination of resonant tones from the ambient noise of the recording system and surroundings.

5:40

4pMUb7. Simulation of 'Suikinkutsu' sound considering sound radiation from water surface. Yuki Fujita, Naoto Wakatsuki, and Koichi Mizutani (Univ. of Tsukuba, fujita-y@aclab.esys.tsukuba.ac.jp)

This paper is about a 'Suikinkutsu', a kind of Japanese traditional musical instruments. Different from other musical instruments, 'Suikinkutsu' is composed of an upended jar buried underground in the garden. Dropping water into 'Suikinkutsu' works as sound source, and attracting harp-like sound inside the jar, 'Suikin-on', is generated. Although the sounding mechanism of 'Suikinkutsu' is interesting because there are complicated interactions among air, water, and jar made of ceramic to generate harp-like beautiful sound, its analysis has not been achieved yet. To investigate the sounding mechanism of 'Suikinkutsu', simulation of interaction between air and water is performed. The obtained results suggest that the water in the jar contributes to sound generation of 'Suikinkutsu'.

THURSDAY AFTERNOON, 17 MAY 2012 — HALL B, 2:00 P.M. TO 5:40 P.M.

Session 4pNSa

Noise, Engineering Acoustics, Physical Acoustics, and Signal Processing in Acoustics: Active Noise Control III

Siu Kit Lau, Cochair
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Contributed Papers

2:00

4pNSa1. Active noise control in the exhaust port of a vacuum cleaner. Ping Wang, Jiancheng Tao, and Xiaojun Qiu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, wangpingfairy@163.com)

Exhaust port is one of the main sound transmission paths for noise radiation of a vacuum cleaner. Porous sound absorption materials are usually stuck at the inner surface of the exhaust port to absorb the noise; however the noise reduction performance, especially at low frequency, usually is limited due to the limited transmission path length to the exhaust port for airflow and the low acoustical absorption coefficients of porous materials at low frequency. In order to reduce the low-frequency noise which radiates outside through the exhaust port, active noise control technologies are applied in exhaust port, where the configurations of the active control units including the reference sensor, the control loudspeaker and the error microphone are investigated. Finally, experiments are carried out for performance validation.

2:20

4pNSa2. Active control on the scattered radiation by a rigid surface. Ning Han and Xiaojun Qiu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, hanning@nju.edu.cn)

An approach for predicting the scattered sound pressure in one- and three-dimensional sound field is proposed by measuring the total sound pressure on a rigid scatterer surface, which is further used as an error sensing strategy in an active noise control (ANC) system to reduce the scattered radiation. Experiments are carried out to validate the prediction method, and a single channel broadband feedforward ANC system is implemented to suppress the impulsive scatterance at the observation point. It is found that the ANC system based on the proposed prediction method is effective. About 12.2 dB reduction of the impulsive scattered radiation is obtained in one-dimensional sound field, and 8.2 dB reduction in three-dimensional sound field. ACKNOWLEDGMENTS The authors would like to acknowledge the support NSFC (Project 11104141).

2:40

4pNSa3. Case study of applying active noise control on communication chassis. Haishan Zou, Di Yu, and Xiaojun Qiu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, hszou@nju.edu.cn)

In some communication chassis, running of the cooling fan causes the noise emission level of the chassis exceed the upper limited level of the standards and regulations. In this case study, various sound transmission paths are indentified, such as the structure borne sound transmission, the air intake and the air outtake, etc. To reduce the direct noise, a metal enclosure which divided into several ducts and with new intake and outtake on its surface is used to cover the chassis. The wall of ducts is equipped with absorbing material to form mufflers, so the middle and high frequency noise is reduced successfully. The key point is that the active control systems are installed in the ducts to reduce the low frequency noise. A noise level reduction of 6-8 dB is achieved with this active-passive hybrid method. However, it is found that it is impossible to achieve such a reduction level without using active noise control because of the abundant low frequency component of the noise.

3:00

4pNSa4. A study on active noise control using multi-pole control sources. Hiroyuki Ichikawa (Kyushu University, 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, ichikawa.hiroyuki.365@s.kyushu-u.ac.jp)

In Active Noise Control (ANC), the difficulty of global control is one of the issues to be tackled. Basically, a lot of control sources and corresponding sensors are needed to realize effective control in large area. In the present study, a new type of secondary source arrangement, the multi-pole source array, is proposed. The idea is based on the Kempton's approach using multi-pole expansion, which was proposed in 1970's. This method is expected to have a capability of realizing more flexible shapes of wavefront compared with conventional control source arrays. The effect of proposed array sources is examined in the free field and in the diffracted field with thin semi-infinite barrier. The results of the numerical simulations and the preliminary experiment indicate the superiority of the proposed method.

3:20

4pNSa5. A robust acoustic feedback suppression algorithm based on spectral subtraction. Feiran 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, feirany.ioa@gmail.com), Ming Wu, Peifeng Ji, and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China)

Acoustic feedback is caused by the coupling between the loudspeaker and the microphone. In this paper a new acoustic feedback suppression (AFS) method is proposed to address this problem. The cross-spectrum of the loudspeaker and the microphone signal and power spectrum of the loudspeaker signal are estimated by the recursive averaging of periodograms. Then, the magnitude spectrum of the acoustic feedback path is obtained by dividing the cross-spectrum by the power spectrum of the input signal. The spectrum of the acoustic feedback signal is calculated using the above magnitude spectrum. Moreover, the spectral modification technique originally proposed for speech enhancement is adopted to remove the acoustic feedback from the microphone signal. The proposed algorithm is less sensitive to the acoustic feedback path changes than the existing adaptive filter approach. Simulation results demonstrate the effectiveness and the robustness of the new method.

3:40

4pNSa6. Assessment of the effectiveness of adaptive active vibration control for the minimisation of radiated sound from panels. Robin Wareing and John Pearse (University of Canterbury, 69 Creyke Road, Ilam Christchurch, robin.wareing@pg.canterbury.ac.nz)

In many practical situations noise is radiated from a noise or vibration source into adjacent areas through vibration of plates. This paper is concerned with the reduction of this radiated noise via active vibration control of such panels. An LMS based adaptive controller is implemented and the experimental results are compared to an ideal model of the noise reduction, this allows the effectiveness of the adaptive controller to be assessed. The plate used in tests is a 1546mm by 946mm simply supported panel. This has been modelled in Matlab allowing the sound field above the plate to be evaluated. Using the theories developed by Nelson et al [1] the theoretical optimal noise reduction can then be calculated. The panel is excited via a point force; the sound pressure at a number of points is then measured. The active control is actuated via a coil based inertial actuator. The sound pressure is measured following the implementation of active control, thus allowing the practical noise reduction to be calculated. 1. Fuller, C.R., S.J. Elliott, and P.A. nelson, Active Control of Vibration. Vol. 1. 1996, London: Academic Press Limited. 332.

4:00–4:20 Break

4:20

4pNSa7. A novel decentralized velocity feedback strategy of plate vibrations using piezoelectric patch actuators. Yin Cao, Hongling Sun, Xiaodong Li, and Jing Tian (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, caoyin@mail.ioa.ac.cn)

Previous studies have shown that, similar feedback gains are required to minimize the kinetic energy of a plate and to maximize the absorbed energy by each force actuator in a wide-band excitation. The energy absorption of a force actuator is easy to measure, but it is hard to get the energy absorption by a moment actuator such as a piezoelectric patch actuator which needs the angular velocity at the mounting point of the actuator. In this paper, the energy absorption of piezoelectric patch actuators for decentralized velocity feedback control of plate vibrations is investigated numerically. A kind of virtual energy absorption of the piezoelectric patch actuator is proposed, which is the multiplication of the torque produced by the piezoelectric patch actuator and the velocity at the mounting point of the actuator. Numerical investigations are performed to explore the relationship of the virtual energy absorption by the piezoelectric patch actuators and the kinetic energy of the structure. The results show that maximizing the wide-band virtual energy absorption is nearly equivalent to minimizing the kinetic energy.

4:40

4pNSa8. Variable step-size μ-law memorised improved proportionate affine projection algorithm for sparse system identification. Longshuai Xiao and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, xls.ioa@gmail.com)

System identification has wide applications in adaptive echo/feedback cancellation, active noise control, adaptive channel equalization, etc. When the system is sparse, recently, a μ-law memorised improved proportionate affine projection algorithm (MMIPAPA) has been proposed to improve the misalignment performance remarkably. However, the MMIPAPA with constant step-size has the conflicting requirement of fast convergence rate and low steady-state error. To solve this problem, a variable step-size version of the MMIPAPA (VSS-MMIPAPA) has been extended by setting each component of the a posterior error energy vector equal the system noise energy. Furthermore, through an alternative method to compute the variable step-size, when the estimated component of the a prior error energy vector is smaller than the system noise energy, further lower steady-state misalignment is achieved. This method leads to an improved VSS-MMIPAPA (IVSS-MMIPAPA). The computational complexity of the IVSS-MMIPAPA is $O(P^2L)$, which may be too high for many applications. Therefore, an approximate algorithm with little performance degradation has been implemented using recursive filtering and dichotomous coordinate descent iteration techniques, by which the complexity has been reduced to O(PL). Finally, the efficiency of both the exact and approximate algorithms developed here has been verified by simulation results.

5:00

4pNSa9. The Fortaleza noise mapping project—a tool for the definition of noise action plans for the airport located in the center of the municipality. Francisco Aurélio Chaves Brito (SER V and UNIFOR, aurelio.semam@hotmail.com), and Jose Luis Bento Coelho (CAPS, Instituto Superior Técnico, TULisbon, Lisboa, Portugal)

Studies of the impact of noise from Fortaleza International Airport, located well within the urban area, based on the noise mapping project that was created for the spatial representation of indicators of ambient noise in the city of Fortaleza, Brazil, which provided a tool essential to analyze and define strategies for the control of noise pollution in the city. This is the first noise map drawn to scale in a large city in Brazil. Noise emissions from major sources that contribute to the sound environment of the city, including road traffic, railway noise, aircraft noise, industrial noise, noise and entertainment areas were included. The method followed a hybrid approach, essentially calculating complemented with experimental measurements for validation and calibration. The study of airport noise allowed detailed results for equip lawsuit against the noise produced during operation of the airport.

5:20

4pNSa10. Building active noise control system decrease MRI driving sound. Shohei Nakayama (Shibaura Institute of Technology, f08091@shibaura-it.ac.jp)

Our purpose is to build Active Noise Control (ANC) system for MRI examination. It is problem for the patients that loud sound which is generates by Magnetic Resonance Imaging (MRI) equipment. Its maximum of the sound pressure level is over 100dB. It does not only make patients sickening, but has possibility of hearing loss. Patients' hearing ability is protected from the loud sound by the ear protectors. The ear protectors decrease around 20dB from the sound. Its performance is unsatisfactory to decrease the MRI driving sound for the patients. Because the patients hear the high level sound which the maximum sound pressure level is over 80dB if any wearing ear protectors. They hope for more the performance. We have an idea that we build the ear protectors whose ANC system controlling low frequency. It is important for the feed-forward ANC system to decide the reference point. We show that the result measured the MRI driving sound for finding the sound source. As a result, we found the sound source is located around the gradient coil. We show computer simulation result that ANC system whose reference point located on the sound source controls the MRI driving sound.

THURSDAY AFTERNOON, 17 MAY 2012 HALL C, 2:00 P.M. TO 6:20 P.M.

Session 4pNSb

Noise, Architectural Acoustics, Animal Bioacoustics, and ASA Committee on Standards: Soundscape and Its Application II

Brigitte Schulte-Fortkamp, Cochair
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K. C. Lam, Cochair
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Invited Papers

2:00

4pNSb1. Enhancement of soundscapes using natural sounds in urban spaces. Jooyoung Hong and Jin Yong Jeon (Hanyang University, Seongdong-gu, Seoul, st200045@hotmail.com)

Design elements of urban soundscape were investigated through individual soundwalk and laboratory experiments. Soundwalking was performed in Seoul; thirty subjects selected evaluation sites when they perceived positive and negative soundscape elements in the soundscape route. It was found that soundscape elements are closely related to visual elements. In order to investigate the effects of the design elements on soundscape perception, laboratory experiments were designed: birds' singing and water sounds were selected as acoustic stimuli enhancing the urban soundscapes. Trees and water features were also added as visual stimuli which were connected to the acoustic elements. The experiments consisted of three parts; 1) audio-only condition, 2) visual-only condition, and 3) audio-visual condition. As a result, the contributions of soundscape elements to urban soundscape perception were derived.

2:20

4pNSb2. Tranquility analysis by soundwalks in Pisa's green areas. Gaetano Licitra (CNR-IPCF and ARPAT Tuscany Environmental Protection Agency, g.licitra@arpat.toscana.it), Claudia Chiari (ARPAT, Tuscany environmental protection agency), Irene Menichini (University of Pisa), and Elena Ascari (CNR-IDASC, Institute of Acoustics "O.M. Corbino")

Identification and analysis of main quiet areas in Pisa agglomeration has been carried out by the University of Pisa and Tuscany Environmental Protection Agency (ARPAT). According to suggestions given by the Environmental Noise Directive (END) 2002/49/CE, authors measured noise levels (using different indicators) and perceived tranquility of users at four different park/green areas in the city (Monumental site – Miracle square, Botanic Garden, Riverside park "Le Piagge", City Park "Giardino Scotto"). In order to analyze tranquility, measurements and analysis on site has been carried out including noise levels acquisition, video recording, binaural audio acquisition and users surveys. At first some different methodologies has been tested and compared to identify most suitable measurement instruments and analysis procedures. The survey, based on similar studies carried out by G. Watts in UK, intended to identify most annoying sources and subjective users impression of the park. In the same sites, soundwalks have been recorded to allow a laboratory test to obtain a second evaluation of the park (video and binaural audio reproduction). First results highlighted the negative effects on objective and subjective evaluation of tranquility done by different type of transport (trains, cars, aircraft) in Pisa's parks.

2:40

4pNSb3. Revision of the new orleans noise code in regard to soundscape, enforcement, and cultural impact. David Woolworth (Oxford Acoustics, Inc. 356 CR 102 Oxford, MS 38655, dave@oxfordacoustics.com)

New Orleans is a city with districts of predominately residential historical structures that overlap with entertainment districts that play host to a tourist based economy, in addition to being the home of rich cultural traditions that include live music and Mardi Gras Carnival. As tourist and entertainment venues and their noise levels increase and expand, residents of mixed use areas are becoming less tolerant of the noise and are vacating the French Quarter that depends on their presence for the vitality of these districts. The complex layers and conflicting schedules of neighborhoods, entertainment zones, and frequent music festivals make a difficult case for enforcement. This paper outlines the methods for assessing the existing conditions and making the code revision to begin the process to strike a balance in the French Quarter and keep future development of the rest of the city in check.

3:00

4pNSb4. Validation of the Swedish soundscape quality protocol. Östen Axelsson, Mats E. Nilsson, and Birgitta Berglund (Department of Psychology, Stockholm University, SE-106 91 Stockholm, Sweden, oan@psychology.su.se)

The Swedish Soundscape-Quality Protocol was developed to help non-experts (e.g., officials working for municipalities rather than soundscape researchers) to make informed, accurate measurements of soundscape quality. The Protocol has hitherto been used in England, France, Italy, Spain, Sweden, and The Netherlands; a Korean version is being developed. Based on field studies – soundwalks in urban residential areas, recreational areas, and parks – the present paper reports on the psychometric properties of the scales of the Protocol. Participants were residents, or visitors to the areas and their results support the reliability and validity of the scales in the Protocol. Because high acoustic quality has a greater effect in visually attractive than in visually poor areas, the Swedish Soundscape-Quality Protocol includes scales for cross-sensory tabulation. These are sound source identification – sounds from humans, nature and technology – attribute scales (e.g., eventful, exciting, pleasant, and calm), overall soundscape quality, and concomitant visual impressions. In brief, the Swedish Soundscape-Quality Protocol is an easy to use and practical tool for measuring soundscape quality. It has the potential to help operationalize how soundscapes can be measured in “quiet areas” to meet a future guideline value of the World Health Organization.

3:20

4pNSb5. A soundwalk study on the relationship between soundscape and overall quality of urban outdoor places. Mats E. Nilsson (Department of Psychology, Stockholm University, SE-106 91 Stockholm, Sweden, mats.nilsson@psychology.su.se), Jin Young Jeon (Department of Architectual Engineering, Hanyang University), Maria Rådsten-Ekman, Östen Axelsson, Joo Young Hong, and Hyung Suk Jang

In a field study, we explored the relationship between the soundscape and the overall quality (good - bad) of outdoor open places. Thirty three residents in down town Stockholm participated in soundwalks near their homes. Along the soundwalk route, the participants assessed six places with respect to the soundscape, the visual environment and the overall quality of the place using a questionnaire. The six locations were preselected to vary in acoustic and visual quality. A regression model with pleasantness of the auditory and visual environment as predictors explained a substantial part of the variance in assessments of the six place’s overall quality. To disentangle the specific effects of auditory and visual aspects, the present study will be complemented with laboratory experiments in which visual and auditory aspects are independently manipulated.

3:40

4pNSb6. Soundscape: its theory and praxis in Chinese classical garden. Xiao-mei Yuan, Shuo-xian Wu (College of Architecture, South China Univ. of Tech, GuangZhou, GuangDong, 510640, xmyuan@scut.edu.cn), and Yan Wu (Faculty of Arts, Jinan University, GuangDong, 510642)

In this paper, an unique soundscape concept in Chinese classical garden is explicated from theory and praxis levers, which is based on the traditional “Harmony” culture, with the philosophical background of Confucianism, Taoism and Buddhism, and the character of the aesthetic refining and gardening construction experience together up. The general aim of which is creating a artistic conception with nature aesthetics and poetic culture to conduct people’s life to archive the highest state. This paper firstly introduces the formation and development of the concept of Chinese classical garden soundscape, and its related social and cultural reasons are investigated as well. And then, the technical logic contained in garden construction rules are sorted out. So the unique characteristic of soundscape theory and praxis in Chinese classical garden are explicated. Key Words: Chinese Classical Garden; Soundscape; Theory; Praxis

4:00–4:20 Break

Contributed Papers

4:20

4pNSb7. Thinking soundscapes into the management of country parks. Lawal Marafa, Kin Che Lam (The Chinese University of Hong Kong, lmmarafa@cuhk.edu.hk), and Lex Brown (Griffith University, Australia)

Countryside and Country Parks, commonly used as an area for recreation, conservation and education, accommodates sunlight and allows free air movement, thus providing visual relaxation. In recent years, ways to make country parks more attractive for use have been highlighted worldwide. Among a number of promoted common themes regarding the protection and enhancement of country parks, sound as a component of the biological and social environment should be utilized for improving the general environmental quality. Based on the large scale field recordings and questionnaire surveys carried out in 3 country parks in Hong Kong countryside, the relationship between subjective evaluation of soundscape and landscape quality as well as the particular contribution of sound in the enhancement of country parks have been statistically analyzed. Results of this study demonstrate a close relationship between subjective evaluation of landscape quality and acoustic quality. Identification of different sound sources in different functional zones, their particular properties and influence on subjective evaluation of the acoustic quality suggests an alternative approach for improving the acoustic quality and consequently the general environmental quality at the countryside location. Keywords: Country Park, Country side, Recreation, Soundscape

4:40

4pNSb8. Constructing the ideal soundscape: a practical study on closing the gaps between soundscape and urban designers. Daniel Steele (School of Architecture + CIRMMT, McGill University, 815 rue Sherbrooke Ouest, Montreal, Montreal, Quebec H3A 2K6, Canada, daniel.steele@mail.mcgill.ca), and Nik Luka (School of Architecture + School of Urban Planning, Macdonald-Harrington Building 815, rue Sherbrooke Ouest, Montréal, PQ H3A 2K6, Canada)

Calls are increasingly made for an urban land-use policy that takes non-vision sensory modalities into account, like hearing, but agents capable of making such changes often lack the expertise to do so. The best progress in acoustics so far has been through intentional soundscape design, which considers sound during the urban design process rather than after. Indeed, soundscape designers should understand how complicated factors play out in-situ, such as findings linking increased driving speeds with acoustically treated roads. Armed with this knowledge, they can take action to prevent further harm to the urban landscape. In practice, however, what can happens is 1) papers in soundscape are written in language not interesting to urban designers; 2) research studies examine the current environment without proposing design updates; and 3) different investigators fail to agree on what constitutes wanted and unwanted noise. Each of these shortcomings contributes to a built environment that reflects little of our sophisticated understanding. In response, this presentation will: demonstrate how soundscape research can fit into current urban design frameworks; review the literature to suggest some small and large acoustically-optimized urban designs; and encourage collaboration channels for the direct flow of soundscape research into urban design practice.

5:00

4pNSb9. An aural and visual study on the urban open space in Shenzhen Dongmen shopping district using ANN models. Lei Yu (HIT Shenzhen Graduate School, Room 425, E-Building, HIT Campus, Shenzhen University Town, Xili Nanshan, Shenzhen, leilayu@hitsz.edu.cn), Jiang Kang (School of Architecture, University of Sheffield, Western Bank, Sheffield, S10 2TN, UK), and Huan Liu (HIT Shenzhen Graduate School, Room 425, E-Building, HIT Campus, Shenzhen University Town, Xili Nanshan, Shenzhen)

The Dongmen shopping district is the oldest shopping market in Shenzhen. Since 1995, the district has become one of ten famous commercial pedestrian-streets in China. Located in a bustling area of Shenzhen, the Dongmen shopping district attracts a large number of people in its tight-street spaces. An open space is therefore designed, which is important for customers to have a rest after exhausted shopping. It is also a relaxing place for the nearby residents. However, the space quality does not match its recreation function. According to an on-site investigation, subjective evaluations of physical comfort in the space are rather poor, including the aural and visual evaluations. Therefore, this paper is going to investigate how subjective evaluations of aural and visual "scapes" influence physical comfort. Based on analyses of aural and visual evaluations and physical comfort evaluation, artificial neural network (ANN) models of predicting subjective evaluations of physical comfort of the open space in Dongmen are explored. Using the best-trained model, a design with various aural and visual "scapes" for the space is reviewed. It is expected that this study can provide an optimised design scheme for the Dongmen open space in its refurbishment programme in terms of the physical comfort.

5:20

4pNSb10. Changes of soundscape along rural-urban gradients and their influence on landscape preference: a case study from Xiamen, China. Yonghong Gan (Institute of Urban Environment, Chinese Academy of Sciences, Xiamen 361021, China; Department of Bioscience & Technology, Zhangzhou Normal University, Zhangzhou 363000, China, yhgan@iue.ac.cn), Tao Luo (Institute of Urban Environment, Chinese Academy of Sciences, 361021, Xiamen, China), Holger Behm (Landscape Planning and Landscape Design, Faculty of Agricultural and Environmental Sciences, University of Rostock, Rostock 18059, Germany), Timothy Coppack (Institute of Applied Ecology (IfAÖ), Alte Dorfstrasse 11, Neu Broderstorf 18184, Germany), and Jiang Liu (Institute of Urban Environment, Chinese Academy of Sciences, 361021, Xiamen, China; Landscape Planning and Landscape Design, Faculty of Agricultural and Environmental Sciences, University of Rostock, Rostock 18059, Germany)

Humans percept their environment mainly visually, but the acoustic background may have a strong effect on overall landscape preference. Through urbanization, not only the appearance, but also the acoustic background of a landscape is changed. In this paper, the following questions are addressed: (a) How does urbanization change soundscape, in terms of its composition and intension? (b) To what extent does soundscape perception affect the process of landscape preference? With audio and video data from field investigations carried out in Xiamen, China, the differences in composition and intensity of soundscapes along the rural-urban gradient were identified. In surveys with sixty test persons, human perception of visual landscape and soundscape, as well as the preference of overall landscape, were measured. Visual, acoustic, and overall landscape quality of the study area was mapped on the basis of a land-use map. The influence of visual and acoustic factors on overall landscape preference is discussed in view of potential methodological approaches for mapping overall landscape quality by integrating visual with acoustic qualities of landscape. This may provide the basis for an objective multivariate assessment tool in landscape planning. Acknowledgment:Project supported by National Natural Science Foundation of China(40971111) and Natural Science Foundation of Fujian Province, China (2011J01280).

5:40

4pNSb11. Noisy spring?—bird song as a component of urban soundscape perception. Jiang Liu, Holger Behm (University of Rostock, Justus-von-Liebig-weg-6, Rostock 18059, Germany, jiang.liu@uni-rostock.de), Timothy Coppack (Institute of Applied Ecology, Alte Dorfstrasse 11, Neu Broderstorf 18184, Germany), Yonghong Gan, and Tao Luo (Institute of Urban Environment, Chinese Academy of Sciences, 1799 Jimei Road, Xiamen 361021, China)

Where do we stand 50 years after Rachel Carson's (1962) scenario of "a strange stillness" that crept over "a town [...] where all life seemed to live in harmony with its surroundings"? Today, research on urban acoustics is focused more on noise control than on those factors that contribute positively to overall environmental quality. Thus, it seems that no significant progress has been made in our concept of an ideal urban environment. The soundscape approach, which considers not only unfavourable noises, but also desirable environmental sounds, could help to achieve this goal. Here, this hitherto neglected aspect of the urban soundscape was focused on. A field investigation was conducted on human perception of bird sounds in an urban area in Germany. First, the role of bird song as a positive soundscape element was confirmed. Then a series of analysis were performed, including characterization of daily spatial patterns of bird sounds, their contribution to overall soundscape, their relationship with other urban soundscape components, and the connection between avian sounds and urban landscape functions. At last, how bird song could improve urban environmental quality was discussed and possible strategies from an urban planning and a conservation perspective was suggested. Acknowledgement: Project supported by National Natural Science Foundation of China(40971111), Natural Science Foundation of Fujian Province, China (2011J01280), and German Academic Exchange Service(DAAD).

6:00

4pNSb12. The Swedish soundscape-quality protocol. Ãsten Axelsson (ISO/TC 43/SC 1/WG 54, osten.axelsson@comhem.se), Mats E. Nilsson, and Birgitta Berglund

The Swedish Soundscape-Quality Protocol was developed to help non-experts (e.g., officials working for municipalities rather than soundscape researchers) to make informed, accurate measurements of soundscape quality. The Protocol has hitherto been used in England, France, Italy, Spain, Sweden, and The Netherlands; a Korean version is being developed. Based on field studies – soundwalks in urban residential areas, recreational areas, and parks – the present paper reports on the psychometric properties of the scales of the Protocol. Participants were residents, or visitors to the areas and their results support the reliability and validity of the scales in the Protocol. Because high acoustic quality has a greater effect in visually attractive than in visually poor areas, the Swedish Soundscape-Quality Protocol includes scales for cross-sensory tabulation. These are sound source identification – sounds from humans, nature and technology – attribute scales (e.g., eventful, exciting, pleasant, and calm), overall soundscape quality, and concomitant visual impressions. In brief, the Swedish Soundscape-Quality Protocol is an easy to use and practical tool for measuring soundscape quality. It has the potential to help operationalize how soundscapes can be measured in quiet areas to meet a future guideline value of the World Health Organization.

THURSDAY AFTERNOON, 17 MAY 2012 S223, 2:00 P.M. TO 6:00 P.M.

Session 4pPA

Physical Acoustics: Laser Ultrasonics: Fundamentals and Applications

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Zhong Hua Shen, Cochair
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Contributed Papers

2:00

4pPA1. Laser ultrasonic inspection for water transport through membrane in a PEM fuel cell. Ching-Chung Yin (National Chiao Tung University, 1001 Ta Hsueh Road, Hsinchu 30010, Taiwan, ccyin@faculty.nctu.edu.tw), Min-Hsiu Wu, and Yu-Shyan Liu

The performance of a proton exchange membrane fuel cell (PEMFC) was reported to be strongly influenced by water management. This work presents laser ultrasonic inspection for in-situ detection of water across the membrane electrode assembly (MEA). A laser induced grating or an alternative interdigitated piezoelectric fiber composite based acoustic wave transducer was placed on an extension of the flow field plate to launch fixed-wavelength plate waves propagating parallel to the flow channels. The acoustic waves passing through the flow channels underlain by the moist MEA were detected at the opposite end using knife-edge technique. The phase velocities of the acoustic guided waves do not change much between both moist and moist-free states. The insight of water transport could be gained through the excess attenuation of guided waves. The synthetic aperture focusing technique was utilized to establish clear B-mode images of water transport through the membrane.

2:20

4pPA2. Laser ultrasound technique for determine the guided waves propagation in layered medium with temperature gradients. Sheng-Po Tseng and Che-Hua Yang (National Taipei University of Technology, tseng3392@gmail.com)

This paper focuses on the modeling and measurements on the propagation behaviors of guided waves propagating along plate-like wave guides with temperature gradients along their thickness direction. A theoretical model based on a recursive asymptotic stiffness matrix method (RASM) with recursion computation algorism is used to provide numerical calculations for the dispersion relations. A laser ultrasound technique is used to measure the dispersion relations. For all the experiments, the measured dispersion curves show good agreement with the theoretical calculation, indicating the reliabilities in the measurement and modeling. This study is useful to thermometry in different temperature gradient environment in a non-contact and non-destructive ways. This paper demonstrates a procedure employing a LUT introduced the dispersion curves for the steel plate with different temperature distribution. In the spectra, the higher mode dispersion curve could be observed more significant phase velocity difference by LUT system. The phenomenon of temperature gradient is more sensitive for the higher mode. This method is proved to have better accuracy for plate-like structures with lower thermal conductivity. However, applications for structures with higher thermal conductivity will be also applicable while the measurement accuracy is further improved.

2:40

4pPA3. Laser ultrasound technique for material characterization of thermal sprayed nickel aluminum coatings in elevated temperature environment. Cheng-Hung Yeh, Che-Hua Yang, Cheng-Yuh Su, and Wei-Tien Hsiao (National Taipei University of Technology, chyeh0706@gmail.com)

Thermal spraying processing usually use nickel-aluminum alloy system as major powder due to its strong adhesion to substrates. The contents of powder material and the processing parameters used in the spraying process cause material properties of coating exhibiting a wide variation. It is difficult to investigate mechanical properties of coating layer in nondestructive way. This research focuses on characterizing mechanical properties of thermal spraying coatings at high temperature environment up to 295° C in nondestructive way. A laser ultrasound technique (LUT) is used for the measurements of dispersion spectra of guided waves. Theoretical model for surface waves propagating along a multi-layered structure with coating and substrate is used to model the sprayed coatings. An inversion algorithm based on Shuffled Complex Evolution (SCE-UA) is used to extract mechanical properties from the measured dispersion spectra cooperating with theoretical model. In the results, surface wave dispersion spectra are measured for three different thickness coatings at different temperature environment. The difference thickness of coatings and temperature environment are reacted to dispersion spectrum of guided waves. It also related to mechanical properties of coating in theoretical model. This method is potentially useful to characterize the mechanical properties of thermal spraying coating in a nondestructive way.

3:00

4pPA4. A full-field mechanical property mapping with quantitative laser ultrasound visualization system. Chia-Han Wu and Che-Hua Yang (National Taipei University of Technology, chw0105@gmail.com)

This research employs a quantitative laser ultrasound visualization system (QLUVS) for the full-field mechanical property mapping in plate-like structures. The QLUVS has the advantage of fast, full-field and quantitative inspection. The QLUVS uses a pulsed laser generate acoustic waves with fast scanning mechanism to reach two dimensional scanning goal and then detected with a piezoelectric longitudinal transducer. By utilizing QLUVS, the spatial and temporal information of guided wave can be obtained with further signal processing, the velocity map can be extracted. Finally the analytical model and inversion algorithm will be integrated into QLUVS for the full-field mechanical property mapping purpose.

3:20

4pPA5. All-optical probing of laser-induced modulation of crack parameters by surface Rayleigh and surface skimming longitudinal bulk acoustic pulses. Chen-Yin NI (LAUM, UMR-CNRS 6613, Université du Maine; College of Science, Nanjing University of Aeronautics and Astronautics, Chenyin.Ni@univ-lemans.fr), Nikolay Chigarev, Vincent Tournat (LAUM, UMR-CNRS 6613, Université du Maine, Av. O. Messiaen, 72085 Le Mans, France), Nicolas Delorme (LPEC, UMR-CNRS 6087, Université du Maine, Av. O. Messiaen, 72085 Le Mans, France), Zhong-Hua Shen (School of Science, Nanjing University of Science and Technology, Nanjing 210094, P.R. China), and Vitalyi Gusev (LPEC, UMR-CNRS 6087, Université du Maine, Av. O. Messiaen, 72085 Le Mans, France)

All-optical monitoring of the nonlinear motion of surface-breaking cracks is reported. Crack closing is induced by quasi-continuous laser heating, while Rayleigh acoustic pulses and skimming longitudinal surface acoustic pulses are also generated and detected by lasers. By exploiting the strong dependence of the acoustic pulses reflection and transmission efficiency on the state – open or closed – of the contacts between crack faces, the parametric modulation of ultrasonic pulses is achieved. It is demonstrated that detection of the parametric modulation of the reflected and transmitted skimming longitudinal waves and Rayleigh waves mode converted by the crack from skimming longitudinal waves is a sensitive technique for the evaluation of crack modifications and local closure. It is observed that skimming longitudinal waves can be more sensitive to crack motion than Rayleigh waves, which are probing the crack motion without mode-conversion. In comparison with an all-optical frequency-domain technique, 1 the time-domain technique is potentially faster for the imaging applications. This research is supported by the grant ANR-10-BLAN-092302 and a postdoctoral fellowship from the Région des Pays de la Loire for Dr. C. Ni. Also, the support for travelling and participation to the conference from China Postdoctoral Science Foundation (No. 20110491409) is acknowledged. 1 S. Mezil, N. Chigarev, V. Tournat, V. Gusev, "All-optical probing of the nonlinear acoustic of a crack", Opt. Lett. 36, 3449-3451 (2011).

3:40

4pPA6. Numerical simulation and experimental study on surface acoustic waves interacting with cracks heated by scanning heating laser source. Zhonghua Shen, Jia Li (College of Science, Nanjing University of Science & Technology, Nanjing 210094, China, shenzh@mail.njust.edu.cn), Chenyin Ni (LAUM, UMR-CNRS 6613, Université du Maine, Av. O. Messiaen, 72085 Le Mans, France, College of Science, Nanjing University of Aeronautics and Astronautics, Nanjing 210016), and Vitali Gusev (LPEC, UMR-CNRS 6087, Université du Maine, Av. O. Messiaen, 72085 Le Mans, France)

The influence on surface acoustic waves (SAWs) propagate through the micro-crack with partial closure is presented in this paper. Heating brought by laser irradiation causes the thermal expansion of the sample, which leads to partial closure of the micro-crack. The partial closure of crack impacts the transmission efficiency of acoustic pulses strongly. Detected SAWs signals are different when heating laser irradiates different regions, when the middle region of the crack is heated, the amplitude of SAWs signals reach the maximum value. Based on this, the experimental system for detecting micro-crack is set up. The crack can be detected effectively by scanning the laser heating source. The finite element method is applied to simulate the temperature rise and relative displacement of crack edges caused by laser irradiation. The relative displacement change with different location of heating laser source is also calculated. The results of numerical simulation and experiment coincide with each other.

4:00–4:20 Break

4:20

4pPA7. Influence of acoustic leakage at liquid-solid interface on the interferometric detecting surface acoustic waves. Yan Zhao, Zhonghua Shen, Jian Lu, Xiaowu Ni (School of Science, Nanjing University of Science and Technology, Nanjing 210094, P.R. China, zhaoyan7906@mail.njust.edu.cn), and Yiping Cui (Advanced Photonics Center, Southeast University, Nanjing 210096, P.R. China)

Optical interferometry is an important method to detect the surface acoustic wave due to its advantages, such as without contact and a wide responsive bandwidth. So optical interferometric detecting the surface acoustic wave (SAWs) was widely used to diagnosis the surface and subsurface defection, which senses the SAWs by detecting the instantaneous surface displacement on the interfaces. Commonly, the material are placed in air or a special liquid and the surface acoustic wave is leaky and readiates energy into the liquid under the form of bulk waves. As a results, there has another factor to make the interferometric signal change, that is the refractive index variation of liquid induced by the leakage acoustic wave. In this paper, we will mainly analyze the influence of acoustic leakage at liquid-solid interface on the interferometric detecting surface acoustic waves and discuss the suitable condition of ignoring the influence of leakage wave in liquid on interferometric signal to ensure the measuring accuracy of interferometric sensor.

4:40

4pPA8. Theoretical and experimental study of wedge wave mode transformation and energy attenuation. Jing Jia, Zhonghua Shen, Ling Yuan, and Xiaowu Ni (Faculty of Science, Nanjing University of Science and Technology, Nanjing 210094, China, jiasujing2@126.com)

Wedge Waves are guided acoustic waves propagating along the tip of wedge. In this paper, both theoretical and experimental work are done to research the wedge wave modes transformation and energy attenuation. Firstly, finite-element method (FEM) was used to simulate the laser induced wedge waves and different orders of wedge wave modes and the mode transformation process were clearly observed. Then pulsed laser excitation and optical deflection beam method for detection was built to investigate these characteristics experimentally. Plenty of wedge waves at different locations were recorded by scanning the excitation laser along the wedge tip, and different orders of wedge waves were observed. By fixing the distance between the excitation and detection position and scanning the samples along the direction normal to the wedge tip, the modes transformation process was obtained. Finally, theoretical solution by the method of potentials is used to explain the principle of mode transformation and energy attenuation. By integrating the power spectra of the acoustic waves, the energy distribution of these acoustic waves is calculated. Both theoretical and experimental results were found that the energy of acoustic waves decrease exponentially to a steady energy of Rayleigh wave within the main wavelength range.

5:00

4pPA9. High quality photoacoustic tomography in scattering biological tissue. Dan Wu, Chao Tao, and Xiaojun Liu (Institute of Acoustics, Nanjing University, joywudan@gmail.com)

Photoacoustic tomography is an emerging biomedical imaging method, which has a high image contrast and good spatial resolution in deep tissues. Because of these merits, PAT has broad applications in biomedical imaging. However, acoustic scattering is unavoidable in tissue with inhomogeneous acoustic properties. Acoustic scattering brings a challenge for the applications of PAT. A new PAT method is presented to obtain the photoacoustic images in scattering tissue. This method is based on the time-reversal invariance of acoustic wave propagation during the acoustic scattering process.

Both numerical simulations and experiments are used to validate this method. The results show that the proposed method can provide better PAT images in scattering tissues than the traditional photoacoustic tomography. Therefore, this method could serve as an alternative for imaging inhomogeneous biological tissues with acoustic scattering.

5:20

4pPA10. Photoacoustic measurement of the optical absorption spectra of dark or turbid media. David Birtill, Anant Shah, Michael Jaeger, and Jeffrey Bamber (The Institute of Cancer Research, 15 Cotswold Road, Sutton, Surrey, SM2 5NG, UK, david.birtill@icr.ac.uk)

A photoacoustic (PA) spectroscopy system has been built to study small samples, particularly the differences between the PA spectra of oxygenated and deoxygenated blood, and various PA contrast agents, with view to optimising the indentifying these media, in clinical PA images. Short (ns) pulses of light from one of two OPO lasers are delivered into a 1mm diameter cylindrical sample holder. The wavelength is scanned over the range 400-700 nm or 690-950nm (depending on laser used) using a different pulse for each wavelength. Sensitive measurement of the thermoacoustic pressure wave energy emitted from the end of the sample, which acts like a disc-shaped piston source, is facilitated by placing it at the focus of a strongly focused ultrasound transducer. The resulting optical spectra are corrected for some system variables, such as the wavelength-dependent laser energy. Further corrections are planned, so that the measurement is truly of optical absorption coefficient at each wavelength. Even without these additional corrections however, the measured PA spectra of oxygenated blood and gold nano-rods strongly resemble their published optical absorption spectra. In addition to its intended use this system may have applications as a laboratory spectrophotometer, suitable for use with optically dark and turbid media.

5:40

4pPA11. Broadband propagation and photoacoustic time reversal imaging using k-space methods. Ben Cox (University College London, Gower Street, London, WC1E 6BT, UK, b.cox@ucl.ac.uk), and Bradley Treeby (Australian National University, Canberra, Australia)

Numerical, time domain, models of broadband acoustic propagation using k-space methods will be described and applied to the problem of image reconstruction in photoacoustic imaging. k-space methods are a subset of time-stepping pseudospectral methods, which use FFTs to calculate field gradients, with an additional correction to make the solutions exact in the homogeneous case. Furthermore, they are closely related to a number of methods in the mathematical, engineering and physical sciences literature, including non-standard finite difference schemes, exponential integrators, wavenumber domain reverse time migration, and wave propagator models developed to solve the time-dependent Schroedinger equation. These connections will be discussed and used to illuminate the advantages of the k-space approach for large scale modelling of broadband acoustic waves. To illustrate the method, a fluid-based k-space model will be applied to the inverse acoustic initial value problem encountered in tomographic photoacoustic imaging. It is well known that this can be solved, in the linear case, using a time-reversed numerical model. Extensions from this basic case to absorbing and non-linear media will be explored.

THURSDAY AFTERNOON, 17 MAY 2012 — S423, 2:00 P.M. TO 6:00 P.M.

Session 4pPP

Psychological and Physiological Acoustics and Signal Processing in Acoustics: Psychological and Physiological Basis of Tonal Language Processing

Fan-Gang Zeng, Cochair
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Michael Tong, Cochair
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Invited Papers

2:00

4pPP1. Relative contributions of temporal and spectral cues for Mandarin and Cantonese tone recognition. Ying-Yee Kong (Northeastern University, Speech Language Pathology and Audiology, Boston, MA 02115, yykong@neu.edu), Tan Lee (The Chinese University of Hong Kong, Department of Electronic Engineering, Shatin, Hong Kong), Meng Yuan (Bionic Ear and Sound Technology Laboratory, Shanghai, China), and Wilson Yu (The Chinese University of Hong Kong, Department of Electronic Engineering, Shatin, Hong Kong)

Despite good speech perception performance in quiet, pitch perception remains a challenge for cochlear-implant users. The current vocoder-based processing technique in cochlear implants preserves temporal envelope and coarse spectral information. This information, however, is insufficient to support high-level performance for tone perception in tonal languages (e.g., Mandarin and Cantonese), particularly in noise. Mandarin has four lexical tones and Cantonese has six lexical tones. Each lexical tone has different fundamental frequency contours. The temporal envelope of the signal also differs among different tones. In this talk, we will first discuss the relative contributions of temporal and spectral envelope and fine structure cues for Mandarin and Cantonese tone perception. We will then describe a new signal processing algorithm developed at the Chinese University of Hong Kong that aims to enhance temporal periodicity cues for Cantonese tone perception in noise. We will present findings from normal-hearing and cochlear-implant listeners, which demonstrate a significant improvement of Cantonese tone perception in noise with this new algorithm.

2:20

4pPP2. Recognition of Mandarin Chinese in noisy, reverberant environments. Liang Li (Department of Psychology, Peking University, Beijing 100871, China, liangli@pku.edu.cn), Xihong Wu (Peking University), and Bruce Schneider (University of Toronto, Mississuaga)

Cochlear-implant users partially recover their speech intelligibility in quiet but not in a noisy, reverberant environment, particularly for those speaking tonal languages, for which semantic information is also expressed by pitch contour. To improve cochlear-implant algorithms for tonal-language users, we have investigated speech recognition in Mandarin-Chinese speaking listeners under adverse listening condition to address four issues related to perceptual fusion and informational masking. First, to what extent do Chinese speech and non-speech sounds differ with respect to the tendency of perceptual fusion (between direct and reflected waves)? Second, why does perceptual separation provide a smaller release from informational masking in Mandarin Chinese than in English? Third, can the use of the recently developed simulated phase-locking stimulation strategy (SPLS, which extracts both phase and amplitude-envelope information) improve speech perception in Mandarin-speaking cochlear-implant patients compared to the continuous interleaved sampling strategy (CIS) currently in use? Fourth, does the prior presentation of a sentence spoken in quiet, by the same person who immediately afterwards produces a masked target sentence, improve identification of the masked target (voice priming) only for tonal-language speaking listeners?

2:40

4pPP3. The perceptual normalization of lexical tones: effects of surrounding tonal context. Valter Ciocca (School of Audiology and Speech Sciences, University of British Columbia, 2177 Wesbrook Mall, Vancouver, BC, V6T 1Z3, Canada, vciocca@audiospeech.ubc.ca), Alexander Francis (Department of Speech, Language, and Hearing Sciences, Purdue University, Heavilon Hall, 500 Oval Drive, West Lafayette, IN 47907), Elaine Eramela (Wonderworld Speech Therapy Clinic Unit 1703A-1705, Landmark North, Sheung Shui, New Territories, Hong Kong), Teresa Siu Kwan Yau (Professional Speech and Hearing Services Ltd., Room 1133, Pioneer Centre, 750 Nathan Road, Kwoloon, Hong Kong), and Wing Man Shum (Hong Chi Winifred Mary Cheung Morninghope School, 220 Lai King Hill Road, Kwai Chung, Kowloon, Hong Kong)

Previous research has shown that the perceptual categorization of a target lexical tone depends on its surrounding tonal context ("tone normalization"). Native speakers have been shown to take into account both preceding and following tonal context in order to carry out the normalization process. The present study focused on the effects of the duration and type of preceding tonal context on tone normalization. Listeners identified the tone of a syllable in final sentence position, as a function of the number of syllables (1, 2, 3, or 4) and of the tone type (level or contour) of a semantically-neutral precursor sentence. The results showed that: 1. The effect of the precursor sentence reached an asymptote once listeners heard two syllables; 2. The tone type of the precursor sentence did not affect performance. This evidence is consistent with a tone normalization process that operates on the basis of a running F0 average of the surrounding tonal context. When tonal context precedes the target tone, the running F0 average is effectively computed over a two-syllable interval.

3:00

4pPP4. Central nervous system markers for lexical tone learning. Patrick C.M. Wong (Knowles Center for Hearing, Northwestern University, Evanston, IL 60208, pwong@northwestern.edu)

A large degree of individual variability can be observed in language learning in adulthood. This variability is especially prominent in the learning of foreign sounds, including lexical tones. In this presentation, I will report a series of experiments from my research group that examines the central nervous system markers for lexical tone learning success. Our knowledge such as collicular sensitivity to frequency modulation and cortical basis of pitch perception can begin to assist in generating specific hypotheses for these markers. In our experiments, native English-speaking adults learned Mandarin-like tone patterns in a lexical context. Behavioral, electrophysiological, neural hemodynamic (fMRI), and neuroanatomical measures were collected to characterize variability in learning success. Among the markers of success we found were integrity of the frequency following response, activity in the posterior auditory cortex, gray matter volume in the Heschl's Gyrus, and white matter connectivity within the auditory cortex. Some of these markers were used to redesign training in order to optimize learning for all learners, including training strategies such as enhancing auditory pitch cues and reducing trial-by-trial stimulus variability. Taken together, these experiments provide information regarding the neural basis of lexical tone learning and demonstrate the feasibility of using biomarkers for designing training. [Work supported by NIH and NSF]

3:20

4pPP5. Acoustic cues for lexical tone perception in hearing-impaired listeners. Shuo Wang (Beijing Tongren Hospital, Capital Medical University, Beijing Institute of Otolaryngology, Beijing 100730, China, shannonwsh@yahoo.com.cn), Li Xu (Ohio University, Athens, OH 45701), Ruijuan Dong, Jing Li (Beijing Tongren Hospital, Capital Medical University, Beijing Institute of Otolaryngology, Beijing 100730, China), Robert Mannell (Macquarie University, Sydney, Australia), and Luo Zhang (Beijing Tongren Hospital, Capital Medical University, Beijing Institute of Otolaryngology, Beijing 100730, China)

This series of studies was aimed to investigate how listeners with sensorineural hearing loss (SNHL) and with auditory neuropathy syndrome disorder (ANSD) achieved lexical tone recognition using either the temporal envelope (E) or the fine structure (FS) cues. Five groups of Mandarin-speaking subjects, including (1) 22 normal-hearing subjects, (2) 8 moderate, (3) 13 moderate to severe, (4) 10 severe SNHL patients with various degrees of SNHL, and (5) 10 patients with ANSD, participated in the study. Monosyllabic words were processed through a 16-channel "auditory chimera" in which E from a monosyllabic word of one tone was paired with FS from the same monosyllable of other tones. On average, 92.0%, 67.4%, 58.1%, 37.5%, and 17.1% of the tone responses were consistent with FS cues, while 5.8%, 23.7%, 31.1%, 45.2%, 42.7% of the tone responses were consistent with E cues for the 5 groups of subjects mentioned above. Therefore, as the hearing loss becomes more severe, the ability of SNHL patients to use FS for tone recognition becomes more deteriorated. The ability of ANSD subjects to use FS is even poorer than patients with severe SNHL even though their pure-tone thresholds were only moderately elevated in the low and mid frequencies.

3:40

4pPP6. Lexical tone development in children with cochlear implants. Li Xu (Ohio University, Athens, OH 45701, xul@ohio.edu)

Most of the languages in the world are tonal. In a tonal language, voice pitch variation (i.e., tone) at the syllable level is a segmental structure that conveys lexical meaning of words. Multichannel cochlear implants (CIs) have shown great success in providing profoundly-deafened individuals with satisfactory speech perception in quiet. However, contemporary speech-processing strategies used in CIs do not explicitly code pitch information. This presentation will be focused on (1) acoustic cues for recognition of lexical tones, primarily the Mandarin Chinese tones, and the relative contributions of various cues to tone recognition, (2) results of tone recognition experiments in implant recipients in relation to their differences in demographics, devices, strategies, and psychoacoustic abilities, (3) relationship between music pitch perception and lexical tone recognition, and (4) results on tone production and vocal singing in prelingually-deafened, native tonal-language speaking children with CIs. It is concluded that there are marked deficits in tone development in tonal-language-speaking children with CIs. Results also indicate that early implantation and experience with using the device help improve tone development in prelingually-deafened tonal-language-speaking children. [Work supported by the NIH/NIDCD.]

4:00–4:20 Break

4:20

4pPP7. Importance of temporal periodicity on Mandarin tone perception of cochlear implant recipients. Meng Yuan, Jin Sun, Youyuan Chen, and Haihong Feng (Shanghai Acoustics Laboratory, Chinese Academy of Sciences, No. 456, Shanghai Xu Hui Qu Xiao Mu Qiao Rd., Shanghai, China, yuanmeng61@gmail.com)

Cochlear implants (CIs) were designed primarily to enhance the speech perception of western persons with bilateral sensorineural deafness. The CI technology used in the current commercial devices, in terms of both hardware and software, didn't focus on the delivery of tone or pitch related information. It's known that lexical tone is important to Mandarin-speaking people. To better encode pitch-related information, the current study will investigate the effect of temporal envelope periodicity to Mandarin tone perception of CI recipients. Temporal periodicity from each frequency band is manipulated corresponding to the fundamental frequency (F0) of the speech signal. Psychophysical experiment was carried out to evaluate the Mandarin tone perception performance of CI recipients in quiet and noisy conditions. Test materials were monosyllabic and disyllabic Chinese words with four different tones. Experimental results showed that F0-related coding on temporal periodicity can improve the Mandarin tone perception ability of CI recipients. It is indicated that new speech processing strategy should be developed for better tone perception of Chinese CI recipients. This work is supported by National Natural Science Foundation of China (11104316), and Shanghai Natural Science Foundation (11ZR1446000).

4:40

4pPP8. Combined electric and acoustic stimulation to improve tonal language processing in cochlear implant users. Fan-Gang Zeng and Hsin-I Yang (University of California, Irvine, CA 92697-5320, fzeng@uci.edu)

A unique characteristic of tonal language processing is the use of multiple acoustic cues that are distributed in both time and frequency domains to achieve robust recognition of tones. The temporal cues can be represented by simple acoustic differences in duration or amplitude contours that are correlated to tonal patterns. The spectral cues can be represented directly by fundamental frequency or indirectly by its harmonics. Because contemporary cochlear implants process only the less salient temporal cues, their users have generally poor tonal processing abilities. The present study takes advantage of residual low-frequency acoustic hearing (e.g., $<$ 500 Hz) that is often present in the cochlear implant users. Although this low-frequency hearing contributes little to speech intelligibility directly, it is sufficient to convey tonal information via fundamental frequency. The present study measures word recognition in noise in a group of Mandarin-speaking cochlear implant users who have significant residual low-frequency acoustic hearing. Preliminary data showed that combined electric and acoustic stimulation significantly improved word recognition in noise. The mechanism for this improvement will be discussed in terms of independent or synergetic contributions from the acoustic and electric cues.

Contributed Papers

5:00

4pPP9. Cochlear implant users' melody recognition with pitch and loudness cues. Xin Luo, Megan Masterson, and Ching-Chih Wu (Purdue University, luo5@purdue.edu)

Normal-hearing listeners can recognize familiar melodies from loudness changes, although more poorly than from pitch changes. Two hypotheses were tested here: (1) cochlear implant (CI) users can also use loudness changes to perceive melodic contours, and (2) their melody recognition can be enhanced with consistent pitch and loudness changes. In Experiment 1, melodic contours were created by changing the F0s of harmonic complex tones, by changing the intensities of broadband noise bursts, or both. In Experiment 2, familiar melodies were recorded by a pianist. The note intensities were kept as recorded, equalized at 70 dB SPL, or changed from 45 to 75 dB SPL to match the relative pitch changes in semitones. Loudness melodies were also generated with intensity-changing noise bursts. The results showed that CI users' melodic contour identification was better with both pitch and loudness changes than with either alone. Specifically, adding loudness changes significantly improved identification of melodic contours with 1-semitone intervals. CI users had similar familiar melody recognition with pitch or loudness changes alone. For all but one subject with the lowest performance, familiar melodies were not better recognized when pitch and loudness changed together, showing a lack of integration between the two cues.

5:20

4pPP10. Effects of two acoustic continua on the within-category perceptual structure of tones. Junru Wu (Phonetics Laboratory, Leiden University Centre for Linguistics, the Netherlands, Lipsiusgebouw, Cleveringaplaats 1, 2311 BD Leiden, Room Number 104, j.wu@hum.leidenuniv.nl)

The present study investigated effects of two acoustic continua on the within-category perceptual structure of Putonghua Tone 2 and Tone 3. These two tones were simulated with tokens varying along two acoustic continua about F0 contour: the timing of F0 turning point and falling of F0. Three different syllable durations were tested with voice quality under control. Multidimensional scaling analyses were applied to investigate relative influence of phonetic identification and category goodness on the perceptual dissimilarity of synthesized tonal tokens. The result revealed that Tone 3 has later F0 turning point and greater F0 falling than Tone 2, which confirms former findings. The new finding is that perceptual representation of these two tones categories is different in their internal structures. Best tokens disperse within categories and are usually not unique. Perceptual space involving Tone 2 tokens shrink but that involving Tone 3 doesn't. Goodness rating contributes significantly to the dissimilarity scaling across Tone 2 tokens but not Tone 3 tokens.

5:40

4pPP11. How is a five-level-tone contrast possible? Jianjing Kuang (UCLA Campbell Hall 3125, kuangjj@gmail.com)

Black Miao is the most famous five-level-tone language (reported in Kwan1966), which has been a challenge for tonal theories; however, the phonetic realizations of these five level tones have not been studied. This study thus examines the same Black Miao dialect as the one Fang-Kuei Li recorded. We conducted both production and perception experiments in the field to understand how this five-level-tone contrast is possible. Simultaneous EGG and acoustic recordings were made in the production experiment; the perception experiment consisted of an identification task and a discrimination task. Preliminary results show that these five level tones do not merely contrast in F0; non-modal phonations play an important role in the tonal contrasts. Vocal fry and tense voice help distinguish 11 and 55 from other tones, while breathy phonation distinguishes 33 from 22 and 44. Therefore, the tonal contrasts are optimal in dispersion in a 2-D (F0 and phonation) tone space.

THURSDAY AFTERNOON, 17 MAY 2012 — S425, 2:00 P.M. TO 7:00 P.M.

Session 4pSP

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

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Contributed Papers

2:00

4pSP1. Multiple maskers for speech masking in open-plan offices. Yue Wang (Institute of Acoustics, Chinese Academy of Sciences, enereswang@gmail.com), Horst Albert Drotleff (Fraunhofer Institute of Building Physics), and Ping Li (Institute of Acoustics, Chinese Academy of Sciences)

Sound masking is an effective method to reduce speech intelligibility for reaching acceptable speech privacy in open-plan offices. Speech-like masker is considered as a good choice to reduce speech intelligibility. Using one masker in masking system is the common way so far. In this paper, a new method was proposed, using randomly integrated speech frames (SF) of the speaker as masker. Because SF masker was based on speech of the same speaker, it is a good speech-like masker. We further studied the method of using multiple SF maskers. The performance of this new masking method was evaluated by objective measurement (STOI). The speech masked by one SF masker and multiple SF maskers were compared. Results showed that speech with SF masker could reduce speech intelligibility. Speech with multiple SF maskers led to poorer speech intelligibility than speech with only one SF masker. Masking speech by multiple SF maskers is an effective way to reduce speech intelligibility in open-plan offices. (Acknowledgement: This research was supported by the joint training PhD program of Chinese Academy of Sciences-Fraunhofer, and Fraunhofer Institute of Building Physics in Stuttgart, Germany.)

2:20

4pSP2. Acoustic privacy area generation based on simple summation of numerous loudspeaker signals. Takuma Okamoto, Yukio Iwaya, and Yoiti Suzuki (Tohoku University / 2-1-1 Katahira, Aoba-ku, Sendai, 980-8577, Japan, okamoto@ais.riec.tohoku.ac.jp)

We propose a new speech privacy technique based on simple summation of numerous signals using N-channel loudspeakers. A speech signal mixed with high-level white or pink noise and with a delay set appropriately for each channel is reproduced by each loudspeaker to be synchronized at a specified sweet spot. At the sweet spot, SNR increases proportionally to the square root of N. A speech signal at and around the sweet spot is enhanced significantly, making it easily intelligible. Computer simulations were conducted assuming a linear array, a circular array, and a surrounding 157-loudspeaker array to estimate SNR and the Speech Intelligibility Index (SII). Results show that SNR and intelligibility in terms of SII effectively increase only at and near the sweet spot when using a surrounding 157-loudspeaker array. The SII at the sweet spot increases to 0.46, although those at areas distant from the sweet spot are nearly 0.

2:40

4pSP3. Real-time sound source localization based on multi-resolution scanning in frequency domain. Kohei Hayashida, Masanori Morise, Takanobu Nishiura, and Yoichi Yamashita (Ritsumeikan University, 1-1-1 Noji-higashi, Kusatsu, Shiga, Japan, cm012063@ed.ritsumei.ac.jp)

Robust speech recognition is necessary for realizing useful speech interfaces. A microphone array is an effective item at capturing distant-talking speech with high-quality in noisy environments. It captures the target speech by localizing a talker and steering the directivity. These processing is usually executed in each frequency. For realizing useful speech interface, these processing must be finished in real-time. In the research into sound source localization, various methods have already been developed, and these methods localize a sound source based on acoustic space scanning by fixed resolution in each frequency. Therefore, calculation time is increased, and the real-time processing is difficult in higher spatial resolution. To overcome this problem, we proposed the localization method based on multi-resolution scanning in frequency domain. The lower frequency band has lower spatial resolution, and the higher frequency band has higher spatial resolution. The proposed method localizes the sound source by lower spatial resolution in lower frequency band and higher spatial resolution in higher frequency band. Therefore, the proposed method can reduce the calculation time without degrading the localization accuracy. The experimental results in noisy environment indicated that the proposed method could reduce the calculation time and could achieve the real-time sound source localization.

3:00

4pSP4. An identification of speaker-dependence in reverberant-robust speech recognition. Takahiro Fukumori, Masanori Morise, Takanobu Nishiura, and Yoichi Yamashita (Ritsumeikan University, 1-1-1, Nojihigashi, Kusatsu, Shiga 525-8577, Japan, cm013061@ed.ritsumei.ac.jp)

In recent years, a hands-free speech device has been developed with improving speech-recognition techniques. There is, however, a problem that the reverberant speech degrades the recognition performance in the field of distant-talking speech recognition. It is possibly addressed by taking preventive measures against the degradation of recognition performance with the reverberant criteria to estimate the recognition performance. We have already proposed the method to estimate recognition performance with ISO3382 acoustic parameter based on an impulse response. In this method, the recognition performance was estimated without speech features. Identification of the speaker with robust or weak features against reverberation makes it possible to adapt acoustic model for each speaker toward improving the recognition performance. In this research, we designed the speaker-dependence criteria in reverberant speech recognition. We first investigated existence of the speaker with robust or weak features against reverberation in various reverberant environments. After that, we compared clean and reverberant speech data in terms of speech features such as MFCC, delta MFCC, delta power, and utterance speed to evaluate the effects of reverberation on speech recognition. An experimental result showed the utterance speed was one of the effective candidates for the identification of speaker-dependence in reverberant-robust speech recognition.

3:20

4pSP5. Joint Doppler and time delay estimation by compressed sampling. Xuan Li (Institute of Acoustics, Chinese Academy of Sciences, 100190, lixuan.ioa@gmail.com), Xiaochuan Ma, Shefeng Yan, and Chaohuan Hou

Joint Doppler shift and time delay estimation is an important topic in radar, sonar and communication applications. Least square (LS) is a classical and effective method for solving the problem. However, the performance degrades severely in the scenario of low ratio of signal-noise (SNR), due to the instability of matrix inversing. In this paper, a high-resolution method is proposed basing on the compressive sampling theory. The 2-dimension channel response can be sparsely recovered, and high-resolution Doppler shift-time delay estimation can be described with an underdetermined equation solving problem. Three categories of algorithms, including diagonal loading least squares, ℓ_1 Regularization, and Greedy Pursuit, are adopt to solve the problem and show outstanding resolving capabilities. The three categories of algorithms are analyzed and compared in different conditions. For the dictionary appears not to have unit norm columns, Greedy pursuit is not good as ℓ_1 Regularization in general, and simulation results demonstrate it.

3:40

4pSP6. An efficient post-processing filter for multi-channel acoustic echo cancellation system. Henglizi Zhang, Kai Chen, and Jing Lu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, zhanghenglizi@gmail.com)

The acoustic echo usually cannot be totally removed by the adaptive filter due to the insufficient filter taps, nonlinearity of the echo path transfer function, and the background noise. A post-processing filter, which aims to suppress the residual echo, thus plays an important role in the acoustic echo cancellation system. In this paper, an efficient post filter is proposed based on the coherence between the error signal and the echo estimation of the adaptive filter. The proposed method can be seamlessly combined with the multi-delay frequency-domain adaptive filter. Furthermore, it can be easily extended to multi-channel processing. The instrumental evaluation and listening test both demonstrate the superiority of the proposed method.

4:00–4:20 Break

4:20

4pSP7. A study of beamforming in FDTD method to synthesize directional sound for sound field reproduction. Kota Nakano, Masanori Morise, Takanobu Nishiura, and Yoichi Yamashita (Ritsumeikan University, 1-1-1, Noji-higashi, Kusatsu City, Shiga prefecture, 525-8577, Japan, cm010064@ed.ritsumei.ac.jp)

Sound field reproduction is useful to represent sound contents with high-realistic sensation. We have proposed a sound field reproduction system to control the perceived angles of virtual sound sources. The system synthesizes the propagated sounds by using sound field simulation and multi-directionally represents them with head-enclosed loudspeakers. The multi-directional sound representation requires the sound field simulation to synthesize directional sounds. We therefore focus on beamforming by array processing in finite-difference time-domain (FDTD) method. FDTD is an effective method to simulate sound field. It discretizes the time-space with meshes and defines the distribution of sound pressure level (SPL) by calculation result of the SPL propagation between the each mesh. We accordingly propose a new approach for sound field simulation to synthesize directional sound. In the proposed approach, the beamforming is applied to the SPL propagation between FDTD meshes. The beamforming with FDTD meshes achieves to control the directional characteristics at each mesh for SPL propagation. With the directional characteristics controlled by beamforming, the directional sound is synthesized. According to the evaluation conducted to verify the performance, it was confirmed that the proposed approach enabled FDTD to control the directional characteristics for directional sound synthesis.

4:40

4pSP8. Persymmetric adaptive detection of distributed targets in homogeneous environment. Chengpeng Hao, Xiaochuan Ma (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, haochengp@mail.ioa.ac.cn), Shefeng Yan (Institute of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China; State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China), Qi Xu, and Chaohuan Hou (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China)

In this paper, we deal with the problem of adaptive detection of distributed targets in Gaussian disturbance with unknown but persymmetric structured covariance matrix. A homogeneous environment is considered at the design stage, and a receiver based on the generalized likelihood ratio test (GLRT) is derived. Remarkably the new receiver ensures the constant false alarm rate (CFAR) property with respect to the covariance matrix of the disturbance. Finally, a performance assessment, conducted by Monte Carlo simulation, has shown that the new receiver can significantly outperform its unstructured counterpart due to embedded a priori knowledge about the structure of the disturbance covariance matrix, especially in a severely heterogeneous scenario where a very small number of training data is available.

5:00

4pSP9. Design and implementation of an acoustic signal acquisition system based on Wi-Fi. Xin Li (No. 21, Bei-Si-huan-Xi Road, Beijing, China, lixincome@gmail.com)

This paper presents an acoustic signal acquisition system based on Wi-Fi aimed at acquiring real-time acoustic signals and vibration signals, and transferring them to a center server for further processing and analyzing. In this design, the system is a three-layer hierarchy consisting of wireless nodes, access point (AP) managers and a center server. The center server manages all AP managers and receives signals from them. Wireless nodes can search for the desired AP according to signal strength and are managed by the corresponding AP manager. And the robustness of the communication between AP managers and wireless nodes is enhanced by an improved heartbeat mechanism. The system has been applied into an existing acoustic signal acquisition system and desired performances have been achieved.

5:20

4pSP10. Determination of the physical parameters of systems with a time-domain approach. Yum-Ji Chan and Yumin Zhang (Mechanical Engineering Department, The University of Hong Kong, yjchan@hku.hk)

A time-domain method to analyze the effective stiffness, damping and mass of systems, such as a loudspeaker or a porous cavity, is presented. A time-domain approach is taken to show the phase variation of greater than 2π in transfer functions. With the help of the least-mean-square (LMS) algorithm, the impulse response can be sought by feeding random noise into the systems. The impulse response is then used to deduce the effective stiffness, damping and mass of the system. Experimental demonstration with shunt loudspeakers is presented, and the traditional approach of analyzing air cavities filled with porous material is evaluated.

5:40

4pSP11. A new soft decision feedback equalizer in underwater acoustic communications. Xiaoxia Yang, Jun Wang, and Haibin Wang (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, xiaoxiayang1987@gmail.com)

The tap coefficients of the decision-feedback equalizer (DFE) are associated with the multipath spread of the channel. For the underwater acoustic communication channel, the impulse response often covers tens to hundreds of symbols, requiring at least tens of taps in the feedback filter. Once the incorrect decisions are fed back, the error propagation will be severe. In this paper, we propose a new soft decision equalizer. The expected output symbols of the equalizer will be weighted according to their reliability, and summed up, then the result is fed back. This approach can alleviate the effect of those incorrect symbols. For evaluating the performance of the proposed method, a sea test in shallow water was carried out. The experimental results show that its BER decreases by 50% without channel codec, compared with the conventional DFE. Furthermore, the mean square error can also reduce about 2 dB when the error propagation is severe.

6:00

4pSP12. Design of wavelet frequency-division based signal processing algorithm for an implantable middle ear hearing device. Jiabin Tian, Zhushi Rao, and Na Ta (State Key Laboratory of Mechanical System and Vibration, Shanghai Jiao Tong University Dongchuan RD. 800, Shanghai 200240, China, tian201002@sjtu.edu.cn)

A signal processing algorithm was designed based on wavelet frequency division for an implantable middle ear hearing device(IMEHD) to make the piezoelectric actuator's vibration simulate normal middle ear transfer function. Firstly, the input digital signals were divided into seventeen bands based on Bark frequency scale using wavelet transform. Then, the assigned band gains were applied corresponding to middle ear transfer function. Finally, the algorithm was implemented and verified the performance through an experiment. The satisfactory agreements between the output of the piezoelectric stack and normal middle ear transfer function indicate that the designed algorithm is feasible. This work is supported by the National Natural Science Foundation of China (Grant No. 11072145 and No. 81170910) and the Science and Technology Commission of Shanghai Municipality Foundation (No. 08JC1404700). We are also thankful to Dr S. K. Yin, Department of Otolaryngology, Shanghai Sixth People's Hospital, China, for the facilities and encouragement given.

6:20

4pSP13. Study on mechanical feedback of hearing aids using maximum length sequence method. Jie Cui, Yongjia Ge, and Xiaoli Han (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-Huan-Xi Road, Haidian District, Beijing, China, cuij@mail.ioa.ac.cn)

Mechanical feedback is one of reasons causing hearing aids to howl, therefore the mechanical feedback problems of behind the ear (BTE) hearing aids was investigated by the maximum length sequence (MLS). The realtime data transmission was set up between a BTE digital hearing aid and a computer. The feedback path form the input port of receiver to the output port of microphone was measured with different input magnitude of receiver, in order to investigate the effect of mechanical feedback on the whole feedback path. The experience results verified the availability of the method.

6:40

4pSP14. A novel F0 estimator for pitch enhancement in cochlear implant. Haiyang Hu (Graduate University of Chinese Academy of Sciences and Shanghai Acoustics Laboratory, No. 456, Xiaomuqiao Road, Shanghai, huhaiyang87@gmail.com), Meng Yuan (Shanghai Acoustics Laboratory, No. 456, Xiaomuqiao Road, Shanghai), Qinglin Meng (Graduate University of Chinese Academy of Sciences and Shanghai Acoustics Laboratory, No. 456, Xiaomuqiao Road, Shanghai), and Haihong Feng (Shanghai Acoustics Laboratory, No. 456, Xiaomuqiao Road, Shanghai)

Current experimental Cochlear Implant (CI) sound processing strategies like F0mod [Laneau, et.al., 2006], which aim to improve the pitch perception of CI subjects, amplitude-modulate the envelope of each frequency channel by the fundamental frequency (F0) of the speech sound. Due to the hardware limitation of CI devices, typical F0 estimation algorithms are too complex to be implemented. Low-complexity and accurate F0 estimator will be necessary for such manipulation. This study uses the Summary Auto-Correlation Function (SACF) to extract F0. The new pitch estimator was simulated and compared with typical algorithms that estimate F0 before the sub-band filtering. By the use of SACF, the new estimator reduces the computation load significantly and preserves good F0 estimation performance. # This work is supported by National Natural Science Foundation of China (11104316), and Shanghai Natural Science Foundation (11ZR1446000).

Session 4pUWa

Underwater Acoustics and Signal Processing in Acoustics: Time Series Analysis and Data Processing in Underwater Acoustics II

Nicholas Chotiros, Cochair
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Contributed Papers

2:00

4pUWa1. Detection of broadband sound sources using a randomly spaced linear array. Mario Zampolli, Laurent Fillinger, Alan J. Hunter, and Martijn C. Clarijs (TNO Acoustics and Sonar Group, 2597 AK, The Hague, The Netherlands, mario.zampolli@tno.nl)

Arrays are commonly used in relatively narrow band applications. The array design and processing are usually optimized in order to maximize the gain in the look-direction, while minimizing the interference from sources located in other directions (but possibly not far off from the look-direction). This led for instance to the definition of minimum redundancy linear arrays that maximize the resolution for a given number of sensors. This maximization is, however, effective only at the design frequency of the array. For many active applications (such as active sonar or radar), which are narrow band in essence, this limitation is not an issue. However, if the signal to be detected in its entirety has a broadband frequency content, optimizing the performance of the array at a given frequency leads to possibly poor performance at other frequencies. Since optimal performance cannot be achieved at all frequencies simultaneously, one can aim for sub-optimal but useful performance throughout the entire frequency band. One possibility for achieving this is based on the use of a random hydrophone spacing. The effectiveness of the approach is investigated using simulations and is illustrated with experimental data collected using a randomly spaced linear array.

2:20

4pUWa2. Passive localization using flexible hydrophone array. Lijun Chen, Liang An, and Xiang Gao (School of Information Science and Engineering, Southeast University, ljchen@seu.edu.cn)

Localizing a radiated acoustic source in shallow water precisely is a difficult problem. To achieve this goal, a large aperture between hydrophones should be considered. But a large and rigid array is not convenient when used by ship. Here a flexible hydrophone array is designed to solve this problem. The position of each array hydrophone is randomly placed. Spherical interpolation method is used to estimate the position of acoustic source. This method need precise 3-D coordinates of each hydrophone to estimate 3-D coordinates of the source. In this paper a new method is derived to estimate the coordinates of each hydrophone. Computer simulation is conducted to analysis the performance of passive localization algorithm using the flexible array. Factors such as time-delay, hydrophone coordinates, and array shape, which affect the localization performance are discussed. Lake trial results are presented and show that the experiment results agree highly with the simulation results.

2:40

4pUWa3. Investigation of the horizontal directivity of underwater ambient noise with the circular hydrophone array. Shi Yang, Yang Yixin, and Ma Yuanliang (College of Marine, Northwestern Polytechnical University, Xi'an 710072, China, shiyangeagle@gmail.com)

The ambient noise field in shallow water is highly variable in time and space. The horizontal directivity pattern of ambient noise is anisotropic and can be exploited for source localization in sonar signal processing. In the present work, the 12-element circular hydrophone array with radius of 1 meter was designed and used to observe the ambient noise in the shallow water experiment in the Yellow Sea. The MVDR beamformer with diagonal loading was designed to get the horizontal directivity pattern in frequency band of 1kHz to 3kHz. The conventional beamformer was used to analyze the horizontal directivity of ambient noise above 3 kHz. The noise data were sampled at 25kHz, bandpass filtered between 100Hz and 5kHz in the experiment. The horizontal directivity patterns at the experiment location at different frequencies were obtained based on above methods and the noise data. This work was supported by the National Natural Science Foundation of China(10734030)

3:00

4pUWa4. Study of ocean ambient noise characteristics based on vector signal processing of acoustic. Jialiang Li (Chinese Academy of Sciences, No. 8, Shangqing Road, Shibei District, Qingdao 266023, Shandong Province, China, lijiaqingdao001@163.com), Jianheng Lin, and Xuejuan Yi

With the development of technology, vector sensors are more and more applied in underwater acoustics measurements. As one of the signal processing methods, vector signal processing method based on acoustic energy flow is gradually developed. Methods based on acoustic energy flow overcome some inherent shortcomings of traditional pressure signal processing methods and increase processing gain and DOA estimation accuracy. Methods based on acoustic energy flow have been verified to be practically useful by trials. The existing vector sensor frequency measurement model is modified in this paper. The anisotropy and other characteristics of ocean ambient noise are studied in the paper based on vector signal processing of acoustic energy flow, and the simulation results are reasonable.

3:20

4pUWa5. Directional pattern of a cross vector sensor array. Shi-e Yang (Acoustic Science and Technology Laboratory, Harbin Engineering University, Harbin, Heilongjiang Province, China, yangshie@hrbeu.edu.cn)

The directional pattern of array is a kind of spatial filter for signal processing. For the signal coming from different azimuth can be expressed as a periodic function of, it can be expanded in Fourier series. With the help of a cross vector sensor array, more directional patterns such as quadrupoles and octopoles can be obtained. By solving equations, the first few coefficients of Fourier series can be determined. A combine directional pattern of the vector sensor array, which has

a maximum value at the required direction and very small value at other directions, is properly chosen to improve the performance of the final directional pattern. The method of determining virtual directional pattern proposed in this paper greatly reduces calculation quantity rather than full-space scanning.

3:40

4pUWa6. Bispectrum and cross-bispectrum feature extraction based on vector hydrophone. Lanyue Zhang, Yang Wang, and Desen Yang (Harbin Engineering University, 150001, zhanglanyue@hrbeu.edu.cn)

A vector hydrophone co-locating and simultaneously measures pressure and particle, which obtains more entire information of acoustic field than pressure hydrophone, so that to render the more methods of signal processing. Bispectrum is a kind of high order spectrum analysis method, which impose a lot of information that two order statistics can't obtian. In this paper, the algorithms of bispectrum was researched based on vector hydrophone to obtain the feature of acoustic signals. Bispectrum of particle velocity and cross-bispectrum of pressure and particle velocity were analyzed. Different methods were used to compress the information measured of bispectrum and cross-bispectrum to get the signal feature quickly and effectively. The experiment was carried on based on theoretical research, and the bispectrum and cross-bispectrum were used to extract the features of radiation noise of two kinds of acoustic targets, and the pressure and vector features were obtiand. The researched results showed that Gaussian noise and no-correlated noise by bispectrum and cross-bispectrum could be compressed. The more information of sound field of targets was obtained by vector hydrophone and the distinguishing probability could be improved.

4:00–4:20 Break

4:20

4pUWa7. Research on active detection and direction finding method of autonomous underwater vehicle based on vector sensor. Qing Ling (Science and Technology on Complex Ship Systems Simulation Laboratory, Beijing 100000, shengxueli@yahoo.com.cn), Xue-li Sheng (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001), Yan-yi Yuan (Science and Technology on Complex Ship Systems Simulation Laboratory, Beijing 100000), Jia Lu, Chun-Yan Sun, and Ye Bai (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001)

To resolve the detection and orientation problem of autonomous underwater vehicle which platform is limited, single vector sensor with orientation ability is selected. The method of detection and bearing estimation based on vector sensor and its performance are researched in this paper. Vector sonar synthetical copy correlator and vector sonar synthetical adaptive correlator are put forward here, which use both pressure and velocity information. The composing principle and realizing method of the two processors above are studied and the processing gain, detecting probability and bearing estimation ability of the two are analyzed in acoustic multi-path condition. The theoretical results simulation and experiments show that all methods studied above have better performance than common copy correlation, and in multipath condition, the vector sonar synthetical adaptive correlation does better than vector sonar synthetical copy correlation in the aspect of processing gain and bearing estimation. Keywords: active detection; direction finding; autonomous underwater vehicle; vector sensor; synthetical copy correlator; synthetical adaptive correlator

4:40

4pUWa8. Fast prediction of underwater sound scattering based on Fourier diffraction theorem with modified born approximation. Peizhen Zhang (School of Communication and Information Engineering, Shanghai University, Shanghai 200072, China, and School of Information, Guangdong Ocean University, Zhanjiang 524088, China, zpzhen7242@163.com), Shuozhong Wang (School of Communication and Information Engineering, Shanghai University, Shanghai 200072, China), Runtian Wang (Shanghai Acoustics Laboratory, Chinese Academy of Sciences, Shanghai 200032, China), Yunfei Chen (760 Research Institute, Dalian 116013, China), and Luxian Wang (School of Communication and Information Engineering, Shanghai University, Shanghai 200072, China)

It has been shown in a previous work that the directional pattern of underwater sound scattered from an object insonified by an incident plane wave can be efficiently predicted based on a reversal of diffraction tomography. The method uses the Fourier diffraction theorem with the first order Born approximation. In this paper, modification to the Born approximation is proposed by taking into account the difference in acoustic impedance between the object and water. The impedance difference was ignored in the original derivation of the Fourier diffraction theorem, leading to inaccuracy in the computation. In a two-dimensional case, the proposed modification causes a shift of a circle in the 2D Fourier transform domain, from which spectral samples are taken to give a more accurate generalized projection in the sound field. This leads to improved prediction of acoustic scattering. Extension to 3D is straightforward. The Fourier diffraction theorem with modified Born approximation is applied to produce far-field directional patterns of scattered sounds from several objects of different shapes. Comparison with the original method shows effectiveness of the proposed method. The work was supported by the Natural Science Foundation of China under the Grant No. 61071187.

5:00

4pUWa9. Lake test of 70 kHz correlation velocity log for AUV. Changhong Wang, Long Chen, Wei Qiu, and Yuling Wang (Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, wangch@mail.ioa.ac.cn)

A 70 kHz correlation velocity log (CVL) for AUV has recently been developed. The hardware design and signal processing were discussed. Localized least mean squares (LLMS) algorithm was proposed as the criterion of velocity estimation. The CVL underwent lake trials in Qiandao Lake Sea both on a ship and an AUV in Aug 2011. The CVL was calibrated. Trackline test, varying speed cruise test and drifting test were carried out to examine the performance of the CVL. The uncertainty statistics from the lake trial showed that the velocity accuracy of the CVL was about 0.8%V. This work was supported by Knowledge Innovation Project of Chinese Academy of Sciences.

5:20

4pUWa10. Examination of the complex signals distribution conditions influence. V.A. Akulichev (V.I.Il'ichev Pacific Oceanological Institute, FEB RAS Vladivostok 690041, Russia, akulich@poi.dvo.ru), A.A. Golov, S.I. Kamenev (V.I.Il'ichev Pacific Oceanological Institute, FEB RAS Vladivostok 690041, Russia), and Yu.N. Morgunov

Solving tasks of underwater acoustic communication and navigation for controlling underwater objects much depends on right hydrology and acoustic media condition estimation in operation area. Technically and economically it is worth to deploy on a operation area a stationary source of navigational and communication signals system with the range of functioning equal to operation area maximum size. For navigational system tasks each source in direct period of time beam a unique signal, which is recognized by underwater object and then propagation time and distance is been calculated. In this examination the using of a complex phase manipulated signals with carrier frequency 2000 Hz and 6000 Hz as sounding signals is been tested. These signals were used for transmitting information and navigational data. Beside their beam let to measure and examine waveguide impulse characteristics on the acoustic tracks. In current work the experiment results of informational signal transfer in case of shelf sea is shown.

5:40

4pUWa11. improved methods and results of hrbsss for mapping the deep sea bottom. Xiaodong Liu, Weiqing Zhu, Fangsheng Zhang, Dongsheng Zhang, and Gaofeng Xu (Lab of Ocean Acoustic Technology, Institute of Acoustics, Chinese Academy of Sciences, 100190, liuzhiyu@mail.ioa.ac.cn)

High resolution bathymetric sidescan sonar (HRBSSS) is designed by the Institute of Acoustic, Chinese Academic of Sciences (IACAS). HRBSSS have used the technique of direction of arrival (DOA) estimation. It can work near the sea bottom and mapping the deep sea bottom, and get the high resolution bathymetry and sidescan map, concurrently. HRBSSSs have been mounted on the DTA-6000 deep towed body (maximum work depth 6000m) and the Jiaolong HOV (maximum work depth 7000m). In order to upgrading the performance of the system, a new method for improving the depth accuracy on the nadir of the vehicle and an error correct method for minimizing the influence of the inconsistency among the channels have been developed by IACAS. With these method, we have got the good survey results on the area of the seamount (>2000m depth) and on the area of the sea plain (>5000m depth) in the Pacific Ocean in 2011. These methods and results will be introduced in this article.

6:00

4pUWa12. Recognition of ship echo signal applying wigner distribution feature. Yuan Peng, Guijuan Li, Xin Wang, and Zhengqing Lin (Dalian Scientific Test and Control Technology Institute, Dalian 116013, China, 03081514@163.com)

Recognition of ship echo signal's difficulty lies in extracting effective feature from target reflection signal. Researches show that target echo signal is double grading with time and frequency, so time-frequency is a effective way in the field of echo signal recognition. Wigner distribution is a real means of time-frequency analysis, but it is limited by its false time-frequency spectrum called interference item. In the paper, on the basis of studying the theory of interference item generation and reduction, Choi-Williams kernel function and self frequency-window methods are applied to implement Wigner's interference reduction. Through simulation test using typical signal such as multicomponent signal, Merits and drawbacks of the two means are also analyzed. Then to merchant ship, reef and reverberation, three kinds of CW echo signal, the interesting time-frequency features are extracted. At last, sample sets of above three kinds of echo signals are divided into training and testing sample sets. The number of training samples to the number of testing samples ratio is 1 to 4. The training samples are regarded as typical sample and input to Fuzzy Adaption Resonance Theory (FART) network to train. According to typical samples, the testing samples are tested by the same network. The results show that self frequency-window is a better means to reduce interference item. But high recognition rate can be achieved if interference items exist.

THURSDAY AFTERNOON, 17 MAY 2012 S428, 2:00 P.M. TO 5:00 P.M.

Session 4pUWb

Underwater Acoustics: Underwater Acoustics Poster Session (Poster Session)

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Contributed Papers

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

4pUWb1. Performance simulation and implementation of gold sequences in underwater acoustic CDMA. Yi Tao, Zheguang Zou, and Xiaomei Xu (College of Ocean and Earth, Xiamen University, 361005, China, taoyi@xmu.edu.cn)

Underwater Acoustic Network (UAN) is becoming a hotter research area of underwater acoustic communication recently. Underwater acoustic channels are restricted by narrow bandwidth, long delay and high ambient noise, so Code Division Multiple Access (CDMA) appears to be a promising multiple access technique for UAN. However, due to low underwater data rate, only short pseudo-random sequences are available. Comparing to m sequences, Gold sequences with the same length show the same cross-correlation properties but enable networks to support more users. This paper examines the performance of short Gold sequence in underwater acoustic channel, to determine whether it is a good choice for UAN. Specifically, the paper makes contributions on the following fronts: (1) Based on a Rayleigh fading shallow water acoustic channel model, the performance of bit error rate (BER) of a 63-bit Gold-sequence up to 65 users CDMA is simulated and evaluated in MATLAB platform. (2) Underwater CDMA nodes using Gold-sequence Direct Sequence Spread Spectrum (DSSS) modulation is implemented with National Instruments' new embedded device (cRIO9076). An experiment at sea will be conducted at Xiamen Port to confirm the better performance of short Gold sequences in underwater CDMA.

4pUWb2. Technical scheme and optimization of precision timing and synchronization for underwater systems. Yu Chen (IOA, CAS, chenyu@mail.ioa.ac.cn), Haibin Wang, Xi Chen, and Dejun Wang

Real time base is one of the key features for underwater systems. However, time bias accumulates inevitably in a long-term unattended underwater system, even if high precision oscillators are used. Since it's difficult for underwater equipments to obtain timing information directly from GPS satellites, there's demanding need for effective ways of timing services. In this paper, a technical scheme based on Precision Time Protocol and virtual instruments is given. Timing and synchronization services are provided from remote time server to underwater systems. Technical optimizations are made considering the requirements of practical applications. The optimized solution has been equipped in a practical system. Experiments show it's easy to implement and with high precision, good stability and efficiency.

4pUWb3. Deployment analysis of underwater acoustic wireless sensor networks. Xia Li and Shiliang Fang (School of Information Science and Engineering, Southeast University, Si Pailou 2# Nanjing, zzhlixia@seu.edu.cn)

In this paper, underwater sensor nodes and gateway nodes deployment strategies for two-dimensional communication architecture in Underwater Acoustic Wireless Sensor Networks(UWSNs) are proposed. In the sensor nodes deployment strategy, underwater sensor nodes are deployed in two rows along the coastline, which is of complete coverage and connectivity, localization available and scalable. In the gateway deployment strategy, the gateway deployment is modelled as an optimization problem, by finding the locations of underwater gateway nodes required to achieve a given design objective, which can be minimal expected delay and minimal expected energy consumption. The OPNET network simulator is used to measure the performance of the strategies we studied. Acknowledgment of support: This work was supported by the key laboratory of underwater acoustic signal processing of ministry of education.

4pUWb4. Performance of least mean square equalizer in shallow water acoustic communication channel. Jong Rak Yoon, Kyu-Chil Park, Jihyun Park (Pukyong National University, Department of Information and Communication Engineering, Daeyon-3dong, Namgu, Busan 608-737, Korea, jryoon@pknu.ac.kr), Jungchae Shin, and Seung-Wook Lee (Hanwha Corporation Gumi Plant, 258 Kongdandong, Gumi Kyunbuk 730-030, Korea)

In shallow water, a transmitted signal is severely influenced by sea surface and bottom boundaries. Every signal to receiver except the signals through the water medium experiences a time-variant scattering in the sea surface and grazing-angle-dependent bottom reflection loss in the bottom. Consequently, the performance of underwater acoustic communication systems is degraded, and high-speed digital communication is disrupted by inter-symbol interference (ISI) effect. In this study, least mean square (LMS) algorithm equalizer is adopted to cancel out ISI effect. The equalizer is implemented in underwater acoustic communication system. Experiment is conducted in 26 m depth littoral ocean and the standard Lenna image which consists of 50x50 pixels and 8 bits per pixel is transmitted. It is verified that equalization is an efficient way to achieve high transmission rate in shallow water acoustic communication channel. This work was supported by Research Programs of Hanwha Corporation 2011.

4pUWb5. Performance analysis of anti-multipath fading underwater acoustic communication (AMF-UAC) system in the ocean environmental variability. Jangeun Kim, Taebo Shim, Euicheol Jeong (Soongsil Univ, sentije@ssu.ac.kr), and Youngnam Na (Agency for Defense Development)

Due to surface and bottom space constraints of the underwater acoustic channel, multi-path fading occurs and causes degradation of communication. Multi-path fading depending on the maximum delay time(Tm) and symbol period(Ts) can be divided into two kinds of channels. In this paper, we propose Anti-Multipath Fading Underwater Acoustic Communication(AMF-UAC) System. This system estimates the kind of channel and distinguishes flat fading (Tm<Ts) from frequency selective fading (Tm>Ts) under the ocean environmental variability. After checking the channel status, this system selects a mitigation technique depending on the type of multipath fading. In order to verify AMF-UAC system performance, we test transmission of image using 43.2kbit of gray image through the multi-path fading channel. Test results show that the number of bit errors is reduced from 300 to 10 under flat fading channel and from 20,000 to 90 under frequency selective fading channel when the reference SNR is 14dB.

4pUWb6. A simple distributed networking protocol for underwater acoustic networks. Yongfeng Wang, Yongqing Wu, Chunhua Zhang, and Huizhi Cai (Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, wyf@mail.ioa.ac.cn)

A simple distributed networking protocol for underwater acoustic networks is proposed. Each node in the network learns about the degrees of its separation to a certain destination node by passively analysis the packets transmitted from its immediate neighbors. The address of neighboring node with the least hops to a certain destination node will then be saved in a route table as the relay address for later communication with that destination. The route table is initialized with broadcast address and the relay address to a destination node will be reset to broadcast address if no packets from the destination have been heard through the corresponding neighbor for a given time. By applying such a decay mechanism, the routing information is locally updated to keep up with the topology changes of the network due to mobility, node failure, and channel variations. There is little overhead traffic for networking and the network-wide flooding only happens in the initialization process, the proposed protocol is energy efficient. The details of the protocol and evaluation and simulation results are given in the paper.

4pUWb7. Geoacoustic characteristics of P-wave velocity in Donghae City–Ulleung Island line, East Sea (Sea of Japan). Woo-Hun Ryang (Chonbuk National University, Jeonju Jeonbuk 561-756, Republic of Korea, ryang@jbnu.ac.kr), Seong-Pil Kim (Korea Institute of Geoscience and Mining Resources, KIGAM), and Jin-Hyuk Choi (Agency for Defense Development, ADD)

Donghae City–Ulleung Island Line (DC-UI Line) is a representative line for underwater and geoacoustic modeling in the middle western East Sea (Sea of Japan). In this line, an integrated model of P-wave velocity is proposed for a low-frequency range target (<200 Hz), based on high-resolution seismic profiles (2–7 kHz sonar and air-gun), shallow and deep cores (grab, piston, and Portable Remote Operated Drilling), and outcrop geology (Tertiary rocks and the basement on land). The basement comprises 3 geoacoustic layers of P-wave velocity ranging from 3750 to 5550 m/s. The overlying sediments consist of 7 layers of P-wave velocities ranging from 1500 to 1900 m/s. The bottom model shows that the structure is very irregular and the velocity is also variable with both vertical and lateral extension. In this area, seabed and underwater acousticians should consider that low-frequency acoustic modeling is very range-dependent and a detailed geoacoustic model is necessary for better modeling of acoustic propagation such as long-range surveillance of submarines and monitoring of currents. This research was supported by Basic Science Research Program through the National Research Foundation of Korea (NRF) funded by the Ministry of Education, Science and Technology (2010-0025733) and by the Ministry of Knowledge Economy through the grant of Marine Geology and Geophysical Mapping Project (GP2010-013).

4pUWb8. Hybrid geoacoustic inversion scheme with an equivalent seabed model. Zhenglin Li and Renhe Zhang (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, lzhl@mail.ioa.ac.cn)

Acoustic propagation in shallow water is greatly influenced by the properties of the bottom. The purpose of geoacoustic inversion is estimation of ocean bottom acoustic parameters such as sediment sound-speeds, densities, and attenuations from measured acoustic fields. It is a helpful supplement to direct measurements. Especially, geoacoustic inversion could give low frequency attenuation, which cannot be measured by coring the sediment. Therefore, it has been paid much attention in recent years. A hybrid geoacoustic inversion scheme, which combines several inversion methods together to invert for the bottom parameters, has been proposed based on the fact that the bottom acoustic parameters have different sensitivities to the different physical parameters of acoustic field. This inversion scheme could avoid the problem of the multiple solutions, which are often companied with some geoacoustic inversion methods. The validity of the inversion scheme is verified in a series of sea experiments at different sites. The inverted bottom parameters could be used to forecast the sound propagations and distinguish the atlas marked bottom type quite well. The mapping relation between the sediment types and the acoustic parameters are also given. [Work supported by the National Natural Science Foundation of China under Grant No. 10974218 and Grant No. 10734100]

4pUWb9. Robust wideband adaptive beamforming using waveguide invariant focusing method. Biao Jiang (Hangzhou Applied Acoustics Research Institute, 96 Huaxing Road, Hangzhou, 310012, China, jiangbiao@sina.cn)

The waveguide invariant describing the dispersive propagation in underwater environment can provide useful information for signal processing. In this paper, waveguide invariant focusing is exploited to preprocess the received horizontal array signal, such that the moving target is aligned in a single rank-one signal subspace over the bandwidth, results a reduction of the number of the snapshots necessary for the adaptive beamforming, and the diagonal loading is optimized using the robust Capon method to further improve the robustness. Numerical results show that the proposed method can improve the detection performance with limited observation time. Moreover, passive ranging is accomplished when the beamforming output achieves the maximum over a scanned target range limit.

4pUWb10. The relationship between normalmode interference structure and waveguide invariant. Liu Fuchen (Huaxing Road, Hangzhou, China, fuchen-liu@163.com)

In shallow water, waveguide invariant describes the dispersive characteristics of the filed.It relates the normal group slowness and phase slowness of normal modes.In this paper, waveguide invariant changes with normal mode and interference patterns difference are researched. the relationship formula between waveguide invariant and horizontal wavenumber of normal mode is deduced, which is valided by comparision between theorical value with theorical formula with the value from new formula. Correspondingly, the conclusion is drawn that waveguide invariant is decreasing with horizontal wavenumber decreasing, with is also valided by simulation.

4pUWb11. Fluctuation of acoustic signals due to internal waves in the East Sea of Korea. Jooyoung Hahn, Joung-Soo Park, HyoungRok Kim, Woogeun Chon, Haksue Lee, and Young-Nam Na (Agency for Defense Development, hahnjy@add.re.kr)

This study attempts to investigate the fluctuation of underwater acoustic signals due to internal waves (IWs) off the east coast of Donghae, Korea. Sea experiment was performed with thermistor strings, a sound source, and an array of hydrophones. Based on the thermistor string data, the IWs have characteristics of typical periods of 10-20 minutes, amplitudes of 10-20m, and a duration of 1-2 hours. The IWs were analyzed as they moved from offshore to the coast at a speed of 70 cm/sec. Underwater acoustic signals (CW 80 - 800Hz) also show obvious energy fluctuations with the IWs. Through an analysis of these acoustic signals, fluctuations of periods of 15 minutes are located in time domain. As mixed layer depth varies with time, it may cause travel time difference of acoustic signals. This travel time differences causes fluctuation of acoustic signals in range-independent stratified ocean structure. The spectrum characteristics of the acoustic signals show the possibility that acoustic waves may react to the IWs through mechanisms such as mode coupling and travel time fluctuation.

4pUWb12. Experimental study for short-range acoustic field fluctuations and time-space related characteristics in shallow. ZhongCheng Ma and QingGang Cao (Dalian Scientific Test and Control Technology Institute, Dalian 116013, China, ma_zc@yahoo.com.cn)

Short-range sound propagation characteristic in shallow is an important basis for the underwater test parameters settings, and it is also an important basis for the data summarized and analysis. The interference of the interface reflection and the direct wave in water makes shallow water acoustic show complex time-space variability. By a series of experimental design, fluctuations and time-space characteristics variation in short-range sound field were analysis. Statistical characteristics of the sound field fluctuations and variability characteristics were obtained.

4pUWb13. The effect on the propagation in shallow water with a mega-ripple. Sungho Cho, Donhyug Kang, Young-Kuk Lee (Korea Ocean Research and Development Institute, Ansan, Korea, shcho@kordi.re.kr), and Jee Woong Choi (Department of Environmental Marine Sciences, Hanyang University, Ansan, Korea)

It is well known that acoustic propagation in shallow water is greatly influenced by an interaction of reflection, transmission, and scattering with the boundary condition. Especially, acoustic propagation of low frequency is depended on the structures of seabed and sub-botttom layer. In this study, acoustic propagations for low-frequency in shallow water were measured from field experiments under the condition of mega-ripple. The study area formed various mega-ripple caused by strong tidal current. Precisely bathymetry and sub-bottom structure were obtained from multi-beam and sparker system. In the experiments, acoustic source and receiver for propagation measurement were single bulb system and multiple self recording hydrophone system, respectively. From the measurement, we described characteristics of acoustic transmission for low frequency around the mega ripple regions. [Supported by KORDI (Korea Ocean Research & Development Institute)]

4pUWb14. Research on waveform design of sinusoidal frequency-modulated to extract target phase feature. Wu Yongqing (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, wyq@mail.ioa.ac.cn)

The majority of active sonar systems to detect and classify a target based on the amplitude of the received echo strength or the induced Doppler shift. However, additional classification information is available from the phase shift introduced by some targets as a result of the acoustic boundary conditions. In this paper, waveform design of sinusoidal frequency-modulated (SFM) based on the use of sub-band correlators is presented for measuring the phase shifts associated with certain stationary and moving targets when insonified by broadband transmissions. With the aim of providing improved range resolution, it maintains the amplitude of the transmission constant to maximize energy efficiency and compatibility with existing nonlinear power amplifiers. And the influences on target echo phase measurement are analyzed from reverberation gain achievable with broadband, Doppler sensitive SFM transmissions in littoral waters. At last, field trial results are given for a mine-like classification sonar system and a forward-looking sonar system designed for operation near the sea surface.

4pUWb15. Baseband code estimation for underwater acoustic phase encoding signal. Xiaoyan Wang, Shiliang Fang, and Li Wang (Key Laboratory of Underwater Acoustic Signal Processing of Ministry of Education, Southeast University, No. 2, Sipailou, Nanjing 210096, China, xyanwseu@gmail.com)

A method based on phase jump is proposed in this paper to solve the problem of the baseband code extraction from the underwater acoustic phase encoding signal under non-cooperative conditions. Phase estimation is conducted to the baseband signal, which is demodulated from the received signal according to the estimated carrier frequency. Besides, data smoothing processing is performed twice to reduce the noise effect. Then the baseband code can be estimated by using phase jump of the baseband signal. In practical application, the code sequence is difficult to directly estimate by using the baseband signal waveform due to the distinct amplitude distortion caused by the complexity of underwater acoustic channel. However, the approach based on phase jump is not sensitive to the amplitude fluctuation. So it is especially suitable for the estimation of baseband code form the underwater acoustic phase encoding signal. Computer simulations and experiments in the lake verify the feasibility of this method.

4pUWb16. Sonar detection performance analysis with environmental uncertainty using vertical array data. Xinmin Ren (College of Information Science and Technology, Ocean University of China, 238 Song-Ling Rd, Qingdao 266100, China; The Institute of Acoustics, Chinese Academy of Sciences, 21 West Road of Beisihuan, Beijing 100190, China, qdpeople-2008@163.com), Guofeng Zheng (College of Information Science and Technology, Ocean University of China, 238 Song-Ling Rd, Qingdao 266100, China), Qihu LI, Guiqing Sun, Haining Huang (The Institute of Acoustics, Chinese Academy of Sciences, 21 West Road of Beisihuan, Beijing 100190, China), and Ju Lin (College of Information Science and Technology, Ocean University of China, 238 Song-Ling Rd, Qingdao 266100, China)

Sonar detection performance is related to ocean environmental parameters, such as the source position, the ocean depth, the sound speed profile and geoacoustic parameters, etc. These parameters have strong spatial and temporal variability, which result to environmental uncertainty. The sonar detection system can be limited by the presence of environmental uncertainty. Based on a statistical model of the environmental uncertainty, the optimal Bayesian predictor by L. Sha has been applied in this paper to analyze the effects of environmental uncertainty on detection performance using vertical array data collected in two experiments. The first experiment took place in shallow water off the Italian west coast by the NATO SACLANT Center in 1993(SACLANT Sonar Data). The second experiment took place in shallow water in China in 2008(LOFAR'08 data). Quantitative effects of various uncertain parameters on detection performance have been illustrated to evaluate which one is the most sensitive and which one is insignificant. The present work is supported by the National Defense Fundamental Fund of China (No.613xxxxx).

4pUWb17. Analysis of influence of line array beamforming on the target modulation feature. Xinwei Luo (2# Sipailou, Nanjing 210096, China, luoxinwei@seu.edu.cn)

In the line array sonar, the target signal was obtained after beamforming processing. The beamforming processing could improve the Signal to Noise Ratio (SNR) by using array signal's spatial processing gain. But the distortion would be brought in signal characteristics under non-ideal conditions during line array beamforming in practical application, which increased the difficulty of target characteristics extraction. As an important basis for target classification and recognition, the performance of modulation characteristics detection and extraction is affected by SNR, signal bandwidth, modulation depth and other factors. In this paper, the influence of the time delay error on target characteristics was analyzed based on the conventional beamforming of linear array. And then a method with the combination of theroy and simulation is proposed to analyze the influence of plane wave assumption, azimuth estimation error and disturbance of array shape on modulation characteristics.

4pUWb18. Simulation of time reversed acoustic inversion in shallow water. Bok Kyoung Choi, Byoung-Nam Kim, Bong-Chae Kim, Seom-Kyu Jung, and Donhyug Kang (Korea Ocean Research and Development Institute, Ansan, Korea, bkchoi@kordi.re.kr)

For simulation to time reversed acoustic inversion in any enclosure system as shallow water, we must know the impulse response of the system. Throughout this process we can reproduce an original signal at focal point by convolution of time reversed inversion. To apply this TRA simulation to shallow water environment, after extracting the impulse response about simple condition as shallow water, we analyze the acoustic focusing by TRA inversion. In result, the spatial focusing and time signal patterns can be built by TRA simulation for shallow water condition. [Supported by KORDI (Korea Ocean Research & Development Institute)]

4pUWb19. Spherical Acoustic Lens for 3D Imaging SONAR. Shinta Takano, Teiichiro Ikeda, Kunio Hashiba (Hitachi Ltd., Central Research Laboratory, 1-280 Higashi-koigagubo, Kokubunji-shi, Tokyo 185-8601, Japan, shinta.takano.zo@hitachi.com), Shinsuke Sato, Kentaro Kato, and Mitsuhiko Nanri (Hitachi Ltd., Defense Systems Company, 216 Totsuka-cho, Totsuka-ku, Yokohama 244-8567, Japan)

The states of the structures within a harbor after underwater construction or after a disaster are mainly visually checked and confirmed by divers. A new method that is independent of the water turbidity is desired to replace this direct diver necessity. In this study, we propose a spherical acoustic lens consisting of two or more concentric layers for use in a real-time three-dimensional imaging SONAR, which has many advantages. It does not require electronic circuitry for the beam forming, and it has a wide field of view and can simultaneously collect signals from various directions. The acoustical characteristics of the spherical lens are analyzed under conditions in which the speed of sound of the lens materials by using a finite difference time domain (FDTD) method. The results show that the spherical lens has a narrower receive beam profile and a higher receive gain than conventional compound lenses consisting of three aspheric lens.

4pUWb20. Time-reversal passive localization for underwater radiated noise source. Liang An, Lijun Chen, and Shiliang Fang (Key Laboratory of Underwater Acoustic Signal Processing of Ministry of Education, Southeast University, Nanjing 210096, China, an_liang@seu.edu.cn)

Time reversal (TR) is a technique of passive acoustical sources localization using a time reversal mirror (TRM) and is especially useful in multipath environments. TR is commonly used to localize noise sources in aero acoustics and in many cases pulsed waveforms are used. A TR passive localization method for long duration waveforms was presented in this paper. The TR is applied to actual radiated noise recorded by only one hydrophone to estimate the source distance and depth. Compared to a standard time reversal approach, the proposed technique is based on the virtual time reversal mirror which can match the acoustic channel automatically and leads to an adaptive spatial focusing and temporal compressing as the conventional TRM. The processing result of simulated data and experiment data show this method could mitigate the multi-path interference on passive localization.

4pUWb21. A delay-sensitive gts allocation scheme in ieee 802.15.4 for wireless audio applications. Ying Wang, Peng Zhang, and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, wangying_lab@yahoo.com)

The IEEE 802.15.4 standard is designed for low rate wireless personal area networks, targeting at low cost and low power communications. It also provides a guaranteed time slot (GTS) mechanism to support time sensitive wireless audio applications. The coordinator allocates specific durations within a superframe to guarantee reliability and performance of data deliveries. However the existing first-come-first-served (FCFS) GTS allocation policy can result in the scheduling inflexibility in low-latency data transmission, since it doesn't take into account the traffic specification, delay requirements and the energy resources. In this paper, an adaptive GTS allocation scheme is developed to satisfy the delay constraints of wireless audio applications. Based on recent GTS usage feedback, each device in the WPAN is dynamically allocated a priority number. Then the devices are allocated specific slots according to this priority information. The devices with higher priorities indicate more recent traffic and have higher probabilities to transmit their data in the subsequent superframe. The performance of the scheme is analyzed using Markov chain. The analytical model is validated through simulation results, indicating that the proposed adaptive scheme has better performance for audio transmissions.

4pUWb22. The improved methods and results of HRBSSS for Mapping the Deep Sea Bottom. Xiaodong Liu, Weiqing Zhu, Fangsheng Zhang, Dongsheng Zhang, and Gaofeng Xu (100190, liuxd@mail.ioa.ac.cn)

High resolution bathymetric sidescan sonar (HRBSSS) is designed by the Institute of Acoustic, Chinese Academic of Sciences (IACAS). HRBSSS have used the technique of direction of arrival (DOA) estimation. It can work near the sea bottom and mapping the deep sea bottom, and get the high resolution bathymetry and sidescan map, concurrently. HRBSSSs have been mounted on the DTA-6000 deep towed body (maximum work depth 6000m) and the Jiaolong HOV (maximum work depth 7000m). In order to upgrading the performance of the system, a new method for improving the depth accuracy on the nadir of the vehicle and an error correct method for minimizing the influence of the inconsistency among the channels have been developed by IACAS. With these method, we have got the good survey results on the area of the seamount (>2000m depth) and on the area of the sea plain (>5000m depth) in the Pacific Ocean in 2011. These methods and results will be introduced in this article.

4pUWb23. Measurement of propeller-induced cavitation noise for ship identification. Endang Widjiati, Eko Budi Djatmiko, Wisnu Wardhana, and Wirawan (Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia, ewidjiati@na.its.ac.id)

Acoustic cavitation noise caused by propeller is used in many underwater applications as one of the ships noise signatures. This paper reports characterization process done by measuring the cavitation noise generated by a propeller in a cavitation tunnel. The cavitation tunnel used for the measurement is of the K16B type belonging to the Indonesian Hydrodynamic Laboratory, Surabaya, Indonesia. The experiment is done using a B-series four-blade bronze propeller (diameter 23cm) using a hydrophone placed in the tunnel at window section (60x90)cm^2 of the measurement section (400x85x85)cm^3. Some of the solutions to get accurate and reliable measurement results include calibration measurement and minimizing acoustic noise system and environment. Preliminary process has been done to analyse the characteristics of the measurement results in the time-frequency domain, with the objective being to detect when and which type of cavitation noise occurs in any kind of condition. Measurement outcomes in the form of acoustic data signals are obtained in different conditions by varying the water pressure, flow velocity and propeller speed rotation.

4pUWb24. Underwater signals' characteristics of power spectrum based on data mining. Qingfu Wang (Science and Technology on Sonar Laboratory, Hangzhou Applied Acoustic Research Institute Hangzhou 310012, P.R. China, wangqfu@sohu.com), Shuanping Du, and Huiliang Ge

Power spectrum was always utilized for underwater signal recognition. Distance based data mining was applied to feature extraction of power spectrum. The distances within and between classes were first calculated, then, the weight for each frequency bin was get according to its discrimination, finally, the weighted power spectrum was used to feature extraction and classification. All kinds of signals were generated with underwater acoustic propagation model through wave guide. Experiments and tests were conducted and performance of the method was verified.

4pUWb25. Research on the reverberation measuring technique of underwater directional sound sources. Lee Chi, Shang Dajing, and Zhang Lin (Harbin Engineering University, leechi819@yahoo.com)

The radiated noise of underwater directional sound sources was generally measured in the corresponding ocean environmental condition. Influenced by the ocean environmental noise, the reflection of ocean bottom and ocean surface, it is very difficult to get the radiated sound power of underwater directional sound sources. Reverberation method used in the Architectural acoustics was taken in this paper for measuring the radiated sound power of underwater directional sound sources. It has been used in the radiated sound power measurement of underwater spherical sound sources, but never been used in underwater directional sound sources. The reverberant theoretical formula of directional sound sources was derived by introducing the directivity factor. The radiated sound power of directional sound sources such as dipole and the two same-phase sphere was measured. The measuring and analyzing results show that the reverberant radius of directional sound sources is related to directivity factor. With the directivity factor turns strong, the reverberant radius will become larger but the reverberant controlled area will become smaller, so the measurement range of spatial averaging will become small. The radiated sound power of directional sound sources such as dipole and two same-phase spheres can be measured by large-range of spatial averaging. The measurement techniques can apply to measure the radiated sound power of underwater complex sound sources.

Keynote Lecture

Invited Paper

8:20

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

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

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

Session 5aAA

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

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

Invited Papers

9:20

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

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

9:40

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

11:00

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

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

11:20

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

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

11:40

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

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

12:00

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

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

12:20

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

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

Session 5aAB

Animal Bioacoustics: Acoustic Monitoring of Animals

Douglas Cato, Cochair
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Kotaro Ichikawa, Cochair
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Contributed Papers

9:20

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

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

9:40

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

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

10:00

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

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

10:20

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

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

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

10:40–11:00 Break

11:00

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

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

11:20

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

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

11:40

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

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

12:00

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

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

Session 5aBA

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

Jonathan Mamou, Cochair
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Tadashi Yamaguchi, Cochair
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Invited Papers

9:20

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

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

9:40

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

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

10:00

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

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

10:20

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

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

10:40–11:00 Break

11:00

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

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

11:20

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

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

11:40

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

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

12:00

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

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

Contributed Paper

12:20

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

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

Session 5aEAa

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

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Michael Haberman, Cochair
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Contributed Papers

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Session 5aEAb

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

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

11:00

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

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

11:20

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

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

11:40

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

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

Contributed Papers

12:00

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

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

12:20

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

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

Session 5aEAc

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

Klaus Genuit, Cochair
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Invited Papers

9:20

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

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

9:40

5aEAc2. An experimental and analytical application of vehicle sound quality target cascading. Koo Tae Kang (Hyundai Motor Company, kanggood@hyundai.com)

To achieve pre-defined sound quality target in a cabin, the interior sound is decomposed into air-borne and structure-borne sound by using transfer path analysis. First, the air-borne sound target is cascaded to the target of each sound package component in the form of the transmission loss mainly in the analytical way. Second, for the structure-bone sound, the experimental noise transfer path analysis is extended to identify the sensitive structural path for specific frequency contents as opposed to the traditional methods with which the critical structural components are hard to identify. As a result, the vehicle sound target proved to be successfully cascaded to the component level target by applying aforementioned methods.

Contributed Papers

10:00

5aEAc3. Assessment of annoyance caused by different types of construction noises. Sung Chan Lee, Pyoung Jik Lee, and Jin Yong Jeon (Hanyang University, sungchan@hanyang.ac.kr)

In the present study, annoyance caused by diverse construction noises was evaluated through surveys and laboratory experiments. A survey with a total of 100 construction workers was carried out to investigate annoyance from construction noises at different construction phases. Then, a number of noises from machinery that were evaluated in the survey as highly annoying were recorded from construction sites in Korea. Recorded construction noises were classified into four groups: stationary, fluctuating, intermittent, and impulsive, according to the temporal, psychoacoustical and spectral characteristics of the noises. A laboratory auditory experiment was then performed in order to quantify the total annoyance caused by individual construction noise and multiple construction noises. From the experiment, synthesis curves were derived for the relationship between noise levels and the percentage of highly-annoyed (%HA) and the percentage of annoyed (%A) for the combined noise sources.

10:20

5aEAc4. A-weighting the equal loudness contours. Jeremy Charbonneau, Colin Novak, Robert Gaspar, and Helen Ule (University of Windsor, charbo6@uwindsor.ca)

The standardized equal loudness contours identify the non-linearities of the human auditory system using simple sinusoidal input signals. The graphical illustration of auditory performance trends provides a visual representation of these non-linearities with respect to both frequency and amplitude across the range of auditory perception. Metrics such as the A-Weighting filter approximate one generalized curve shape, in an effort to quantify measured values in a manner that represents the perception of the measured

sound. With the release of the ISO226:2003 version of the standard, the most recent version of the equal loudness contours provide an improved contour set with more refined shapes and steeper slopes. The purpose of this study is to investigate the performance of the A-weighting function compared to the updated curves of the equal loudness contours. Included is an examination and discussion of the appropriateness of the continued use of the existing A-Weighting filter. Given the overall un-hyperbolic shape and flattening of the equal loudness contours as the amplitude of the noise increases, the A-weighted results visually identify the areas of weakness associated with a constant filter approach. This visual examination easily identifies the strengths of each approach as well as the deviations from anticipated outcomes.

10:40–11:00 Break

11:00

5aEAc5. Comparative study of unsteady loudness models for mechanical and real sounds. Colin Novak, Helen Ule, Jeremy Charbonneau (University of Windsor, 401 Sunset Avenue, Windsor, ON, N8Y 4E5, Canada, novak1@uwindsor.ca), and Tomasz Letowski (U.S. Army Research Laboratory, 520 Mulberry Point Road / R-39, Aberdeen Proving Ground, MD 21005-5425)

While noise levels are most often quantified using physical quantities including A-weighted sound pressure level, these metrics do not adequately represent the human perception of the noise. For this, loudness is a more appropriate acoustic metric as it describes the perceived acoustic intensity of a sound. Given that real sounds are often unsteady, a most useful loudness calculation will also account for the perceptional phenomena of time and temporal masking. Studies have been done which demonstrate the performance of several loudness models for pure tone sounds and compare these results to the ISO 226 equal loudness curves. This investigation goes beyond that and evaluates the performance of two unsteady loudness models, the Glasberg and Moore model and the DIN 45631-A1 method, using mechanical and real life sounds. Through implementation ofa jury, the differences of the two loudness calculation methods are demonstrated by plotting the perceived loudness of the sounds for two different levels compared to two different reference levels.

11:20

5aEAc6. Proposed hybrid multiple look approach for calculating unsteady loudness. Helen Ule, Colin Novak, and Robert Gaspar (University of Windsor, 401 Sunset Ave., Windsor, ON, N9B 3P4, ule@uwindsor.ca)

Experimental studies have shown that for short gaps between 2 to 5 ms, the perceived loudness is higher than for uninterrupted noise presented to the ear. Other studies have also shown that the present temporal integration models for the calculation of time varying loudness do not adequately account for short duration phenomena. It has been proposed that the multiple look approach is a more applicable method for describing these short term circumstances. This approach breaks a sound into very small durations which allows for the intelligent processing of the looks and decision making depending on the nature of the stimulus. However, present technologies, such as the Fast Fourier Transform, are not adequate to deal with short duration sounds across the entire frequency spectra. A compromised approach using a proposed hybrid model is presented to account for perceived loudness levels for sounds in the presence of gaps while using an integration model. This hybrid multiple look model was tested using several sounds including mechanical and speech sounds and was found to perform as intended.

11:40

5aEAc7. Annoyance evaluation of interior vehicle motor noise by impulse response method. Sin-Yeob Lee and Buhm Park (Hanyang University, melonavel@hanyang.ac.kr)

In this study, the annoyance of EPS(Electronic Power Steering system) motor noise in a vehicle was evaluated through the anechoic chamber test. Evaluation during actual operation installed in vehicles allows in-situ annoyance tests, but requires a great deal of labor and time. Therefore anechoic chamber test is required to automate the evaluation process under consistent test condition. However vehicle noise test is significantly different from anechoic chamber test from different sound generation and propagation. Due to these discrepancies, the evaluated results among motors open differ between two test results. For consistent test results compared to those obtained from the actual vehicle environment, impulse response of both the vehicle interior and anechoic chamber was measured, and produced a simulated motor noise by using recorded signal in the anechoic chamber. The important peak noises at 1/3 octave bands to take into account of the tonal components were selected for evaluation of its noisiness. From the multiple regression model, the annoyance of motor noise in vehicle using the anechoic chamber test was evaluated.

12:00

5aEAc8. Development of a high quality wireless sound reinforcement system. Li Liu, Peng Zhang, and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, liuli7700@gmail.com)

This paper describes the development and initial application of a wireless sound reinforcement system. This system consists of wireless communication module, digital signal processor (DSP) and audio codec. Signal processing algorithms are used to achieve high quality wireless voice communications and audio playback. To this end, howling suppression algorithm is implemented using DSP. To date, there are three widely used methods for acoustic feedback control: phase-modulating feedback control (PFC), adaptive feedback cancellation (AFC), and notch-filter-based howling suppression (NHS). In this system, we used an adaptive notch filter (ANF) to perform feedback controlling. Simulation Results indicates the effect of this algorithm in suppressing howling. Furthermore, the control strategy of the algorithm is optimized by experiments in a conference room environment. The experimental results show the performance of this system in the real environment.

12:20

5aEAc9. The complexity of sound quality engineering—only a technical issue? Genuit Klaus (HEAD acoustics GmbH, Ebertstr. 30a, 52134 Herzogenrath, Germany, klaus.genuit@head-acoustics.de)

It is an absolutely essential task to enhance the perceived quality of technical products in order to stand out against competitors in times of a highly competitive market. It is clear that sound quality engineering has changed over time from basic level reduction needs to more complex sound quality design processes taking into account several psychoacoustic quantities. Today, it is very evident that sound level indicators are completely insufficient in describing and predicting perceived sound quality. Nevertheless, it is still an ambitious and challenging task to design product sound adequately, which indicates functionality, high quality, and corporate identity at the same time. To fulfil this task sound quality engineering work must change from a simple technical consideration to an interdisciplinary perspective, where knowledge from engineers, psychologies and sociologists is required. Sound quality is not an inherent product property, it develops when listeners are exposed to and interact with a technical product finally judging on the basis of their experience, expectation and context. Sound quality engineering has to consider these "confounding" variables. The paper will highlight the newest developments in the field of sound quality engineering of technical products. For it, experimental data collected with different test methods and its respective analysis results will be shown and discussed.

Session 5aNSa

Noise and ASA Committee on Standards: Environmental Noise and Regulations I

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Invited Papers

9:20

5aNSa1. Annoyance from railway vibration in residential environments: factors of importance when considering exposure-response relationships. Eulalia Peris, James Woodcock, Gennaro Sica, Andy Moorhouse, and David Waddington (The University of Salford, Acoustics Research Centre, M5 4WT, Salford, UK, E.Peris@edu.salford.ac.uk)

Railway induced vibration is an important source of annoyance in residential environments. Annoyance increases with vibration magnitude. However, these correlations between annoyance and physical ratings are weak. This suggests that vibration-induced annoyance is governed by more than just vibration level, and that simple exposure-response relationships alone sometimes do not provide sufficient information for understanding the wide variation in annoyance reactions. Results of investigations made on factors coming into play when considering an exposure-response relationship between level of vibration and annoyance are presented here. Examples of these factors are time of day, situational factors, personal and attitudinal factors. This was achieved using data from case studies comprised of face-to-face interviews (N=931) and internal vibration measurements collected within the study "Human Response to Vibration in Residential Environments" by the University of Salford. This work will be of interest to researchers and environmental health practitioners involved in the assessment of vibration complaints, as well as to planners and consultants involved in the design of buildings and railways. [Work funded by the Department for Environment, Food and Rural Affairs (Defra) UK]

9:40

5aNSa2. A local noise ordinance for relatively small community, population 16,000. Paul Schomer (Schomer and Associate, 2117 Robert Dr., Chammpaign, IL 61821, schomer@SchomerAndAssociates.com)

This paper presents a local noise ordinance developed for a relatively small community with a population of about 16,000. Key features are ease of enforcement by the use, 95 to 99 percent of the time, of simplified methods, with more detailed, rigorous procedures used only when needed (less than 5 percent of the time). When the more complex procedures are used, they are to be performed with qualified personnel and equipment that are external to the township staff. Use of sound level meters is minimized by employment of noise control techniques that depend on more easily measured factors than sound pressure level. Specifically, the ordinance includes such techniques as speed limits (for noise control), distance criteria, time of day, and the criteria "plainly audible," to reduce reliance on traditional measurements using a sound level meter. The ordinance includes enforcement procedures and qualifications for both the "every day" actions that should be 95 to 99 percent of all of the actions, and the more detailed and precise measurements procedures for the infrequent times that they are needed

10:00

5aNSa3. Characteristics of effective noise regulations. Nancy Timmerman (Nancy S. Timmerman, P.E., 25 Upton Street, Boston, MA 02118-1609, nstpe@hotmail.com)

In considering what is an "effective" noise regulation, the definition used will be that which actually reduces the noise regulated. Parameters which are expected to have an effect on the ability to reduce the noise include control over the noise source regulated, a way to measure results, and consequences or enforceability. Examples will be taken from community noise and airport noise regulations in the United States.

10:20

5aNSa4. Noise regulations and control policy in Taiwan. Li-chung Chou (EPA, eric@oe.com.tw), Chung-ho Yu, and Chien-wei Chen (Victory Scientech)

The number of noise complaint cases increased gradually every year in Taiwan. Refer to the annual report, the major noise sources were such as entertainment and business noise, following with noise from construction sites. The high population density and mixed residential and commercial zones caused a lot of noise complaint cases in urban areas. There were also a lot of neighbor noise complaints. In addition, traffic noise with no effective buffer zone caused complaints from the residents along the wayside. In order to solve the various types of noise problems, the Environmental Protection Administration proposed a noise control policy for short-term, mid-term and long-term. This paper will describe the noise control regulation structure and introduce the control policy in Taiwan.

10:40–11:00 Break

11:00

5aNSa5. A study of the effectiveness of the key environmental protection policies for road traffic noise control. Jiping Zhang (Zhejiang Research & Design Institute of Environmental Protection, 109 Tian Mu Shan Road, Hangzhou 310007, China, jpzhang@mail.hz.zj.cn), Paul D. Schomer (Schomer and Associates, Inc., 2117 Robert Drive Champaign, IL 61821), Maurice Yeung (Hong Kong Institute of Acoustics, China), Anguo Zhou, Hui Ming, Juan Chai, and Lu Sun (Zhejiang Research & Design Institute of Environmental Protection, Hangzhou 310007, China)

This paper introduces the history and current status of key environmental policies implemented for road traffic noise control in China, based in part on lessons learned both from international experiences and from China's own experiences. The general framework for traffic noise control in China is similar to what can be found in many foreign countries, and it has been playing an important role in environmental management of the rapidly-growing Chinese economy. The key international environmental policies for road traffic noise control in developed countries are very mature from the technological, economic, legal, and sociological standpoint, and these policies indicate a sharp conflict between road traffic noise and land uses along the road for distance up to about 200m from the road. But China faces greater limitations in using and enforcing these policies to control road traffic noise completely. This paper reconsiders and evaluates China's environmental policies for road traffic noise by collecting and comparing difficult cases of road traffic noise management in other countries to the Chinese experience so as to learn from these experiences now and minimize problems for the future. The research supported from the State Key Lab. of Subtropical Building Science South China University of Technology.

11:20

5aNSa6. Differences in responses to vibration induced in residential environments by railway and construction activities. James Woodcock, Eulalia Peris, Gennaro Sica, Andy Moorhouse, and David Waddington (University of Salford, Salford, M5 4WT, UK, j.s.woodcock@edu.salford.ac.uk)

This paper summarises the results of the Defra (UK) funded project 'NANR209: Human response to vibration in residential environments'. The main aim of this project was to develop exposure-response relationships for the human response to environmental vibration as experienced by residents in their own homes. The sources of vibration considered were railway, construction, and internal sources outside of the resident's control. In this study, 1431 questionnaires were completed with UK residents in their own homes to determine self reported annoyance. Measurements of vibration inside and outside residences were conducted to determine each resident's vibration exposure. Presented in this paper are the exposure-response relationships derived from these data indicating the percentage of people expressing annoyance above a given threshold for a given vibration exposure. In particular, this paper reports the differences in responses to vibration induced by railway and construction activities. [Work funded by the Department for Environment, Food and Rural Affairs (Defra) UK]

11:40

5aNSa7. Noise policy development in Italy and the EU. Gaetano Licitra (CNR-IPCF and ARPAT Tuscany, Environmental Protection Agency, g.licitra@arpat.toscana.it), and Elena Ascari (IDASC-CNR, Institute of Acoustics "O.M.Corbino")

Since the approval of European Directive 49/2005, in Europe many studies and developments have been done to improve quality and accuracy of noise mapping and action planning. With the end of the first round of agglomerations mapping, a comparison between mapping methods and exposure of cities is available. In Italy, tests have been done to compare procedures and results of different cities to understand how to improve quality and effectiveness of noise maps. In the meantime, EU Community worked to change calculation methods in order to find a procedure adaptable for each country but with a common structure. In fact, we have already seen that each country, also using an interim method has produced very different results. In particular, example within Italian maps will be shown and also between ones of different countries in Europe. The level assignment method to population will change too, in order to move towards a more realistic photograph of health conditions: maximum level was too preventive, when considering large building, so a new method has been taken from the German procedure such that each dwelling/room has his own level. Some example of the effects of this change will be evidenced in this paper.

12:00

5aNSa8. Study on the legislation and management system of community noise. Yiping Chu, Yude Zhou, and Wenying Zhu (Shanghai Academy of Environmental Sciences, 508 Qinzhou Rd., Shanghai, chuyp@saes.sh.cn)

Community noise is very common and numerous in urban life. According to the investigation in Shanghai, most of community noise is accidental, and its impact is different from industrial noise, traffic noise and construction noise. Measured by the one-hour or daily equivalent sound pressure level of current standard, the community noise will not exceed usually. On the other hand, based on the current Law of China on Prevention and Control of Environmental Noise Pollution, 'noise pollution' consists of two factors: one is to impact some person, and the other is to exceed the standard. To this extent, most of the community noise is not 'pollution'. This paper argues that the majority of complaints on community noise should be ranged to the area of social public management, and discusses the legislation and management system of community noise from following aspects: 1) collecting of the existing complaints on community noise; 2) study and investigation on domestic and overseas management system of community noise; 3) the connotative meaning of community noise' impact; 4) the position of community noise in urban management, and the management system be forwarded, including the objects of management, law-enforcing department, and the content of management, etc.

12:20

5aNSa9. Research of aircraft noise evaluation and limits. Wenying Zhu, Yude Zhou, and Yiping Chu (Shanghai Academy of Environmental Sciences, 508 Qinzhou Rd., Shanghai 200233, zwing258@yahoo.com.cn)

Along with the European Union to promote the day - and - night equivalent sound level DENL, a growing number of countries have adopted the equivalent sound level DNL or DENL instead of the traditional evaluation for airport noise evaluation, and in sensitive areas the maximum sound level limits was used, some also use the TALA (Time-above an A-weighted sound level threshold). China promulgated the "GB9660 - 88 measurement of aircraft noise around airport" and "GB9661-88 environment standard of aircraft noise around airport", rating for the weighted equivalent continuous perceived noise level WECPNL, and it is independent of other acoustic environmental standard. The state environmental protection department proposed to revise the criteria. In this paper, based on the large amount of measured data, under the aircraft noise statistical analysis, trying to build the relationship between WECPNL, Leq, Lmax; combined with the research results of relationships between noise level and annoyance; investigating the suitability of the current criteria (evaluation and limits), and explore the establishment of a set of aircraft noise evaluation method, which can not only reflect the characteristics of aircraft noise, but also better associated with living requirements.

FRIDAY MORNING, 18 MAY 2012 — HALL C, 9:20 A.M. TO 12:40 P.M.

Session 5aNSb

Noise: Ship Noise and Vibration I

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Contributed Papers

9:20

5aNSb1. Air spring vibration isolation technology for ship propulsion engine. He Lin, Xu Wei, and Shuai Changgeng (Institute of Noise & Vibration, Naval University of Engineering, Wuhan 430033, P.R. China; Science and Technology on Ship Vibration and Noise Laboratory, 430033, P.R. China, helin401@yeah.net)

Propulsion engine (PE) is one of the most dominant noise sources of ship. Due to the imposed requirement of keeping alignment with propulsion shaft during operation, the effective vibration isolation of PE using low frequency mount is difficult to implement as is often adopted by other onboard machinery. In this paper, a low frequency air spring vibration isolation system (ASVIS) with alignment control strategy for PE is conceived and introduced. The application of ASVIS to PE presents both advantages and challenges, which are discussed detailedly in the paper, as well as the feasibility of the ASVIS concept. A systematic design method of ASVIS for PE is established, with focus on the system mechanical behavior optimization and automatic alignment control algorithm development. An ASVIS prototype is designed and manufactured using the proposed method. The performance of the prototype is tested by a series of experiments, including alignment control precision and isolation efficiency. Experimental results show that using ASVIS, the vibration of PE can be attenuated to a satisfactory level, with the alignment between PE and shaft being maintained in the safe range.

9:40

5aNSb2. Experimental investigation of fluctuation characteristics caused by the interaction between turbulent boundary layer and the cavity on the trailing edge of cavity. Yezhen Pang and Mengsa Yu (China Ship Scientific Research Center, No. 222, East Shanshui Road, Wuxi Jiangsu 214082, China, chaos123@sohu.com)

The fluctuation characteristics caused by the interaction between turbulent boundary layer and the cavity on the trailing edge of cavity were investigated. Experiment results showed the fluctuation on the trailing edge of cavity is reinforced. Because the depth of cavity is smaller than the length of opening, the sheartone frequency will not approach the resonance frequency of the cavity, however the sheartone will reinforce the acoustic radiation of the structure downstream the cavity.

10:00

5aNSb3. The study of sono-elasticity of floating bodies in the acoustical environment of shallow sea. Ming-Song Zou (China Ship Scientific Research Center, No. 222, ShanShui East Road, BinHu District, WuXi, JiangSu, zoumingsong1234@yahoo.com.cn)

A concept combining sono-elasticity research field with oceanic sound propagation research field in developed in this paper. The paper formulates in details about how to transfer the sono-elasticity dynamic equations

together with oceanic sound propagation equations in theoretical model into related numerical program by using the PEKERIS neritic acoustic environment as an example. The further verification of this approach and related computing program, and analyses on characteristics of the structural vibration and acoustic radiation in neritic acoustic environment are also provided in the paper by analyzing the problem of a elastic spherical shell as an example. Keywords:Sono-elasticity; Sound propagation; Shallow sea; Spherical shell.

10:20

5aNSb4. Limitations caused by radiation damping and water viscosity on power delivered by ocean wave energy conversion devices. Amadou G. Thiam and Allan D. Pierce (Boston University, 110 Cummington Str., Boston, MA 02215, thiam@bu.edu)

Any generic ocean wave energy conversion device can be suitably "tuned," given various design constraints, to extract the maximum amount of power from a given incoming ocean wave. Just how large this maximum extracted power can be depends to some extent on the magnitudes associated with the various natural damping mechanisms that inhibit the driven oscillations of the device. Such mechanisms include (i) the excitation of ocean waves by the oscillating body and (ii) the friction-induced drag when the body's surface moves relative to the neighboring water. One can generally quantity the joint influence of these mechanisms with a "bobbing test," where an object is initially displaced downward and allowed to freely oscillate, the oscillations dying out exponentially. The authors' general experience is that such a decay typically corresponds to a Q of about 5. Simple analytical models are described to quantify the two mechanisms. For viscous damping, the applicable fundamental model is that which yields a friction force on a flat plate oscillating below a nominally stationary incompressible fluid. For the radiation damping, the theory uses the Green's function solution for a point volume source just below the free surface of an incompressible fluid under the influence of gravity.

10:40–11:00 Break

11:00

5aNSb5. Line-spectra extraction of ship-radiated noise based on harmonic wavelet. Lu Wang and Yuanliang Ma (Institute of Acoustic Engineering, Northwestern Polytechnical University, ylma@nwpu.edu.cn)

The ship-radiated noise line-spectra contain a large number of characteristic information of the ship, which has great significance for detection and classification of underwater targets. This paper presents a method for extraction of the line-spectra of the ship-radiated noise under strong background noise using harmonic wavelet transformation. Harmonic wavelet is a class of wavelet, which has specific expression for the box-like shape of spectrum and frequency-domain continuous distribution. These features make it more accurate in time-frequency positioning with no frequency leakage in the decomposition process which results in ultra-narrow-band and high resolution. Using harmonic wavelet transformation, the ship-radiated noise is decomposed into separate orthogonal frequency bands which are non-redundant and non-leaking. Based on this, the strong background noise can be suppressed and the line-spectra can be extracted in frequency domain and reconstructed in time domain successfully. Simulation and experimental results show that the harmonic wavelet transform is more effective in the background noise suppression, and its ability for weak line-spectra extraction is superior to Fourier analysis and FIR filter methods.

11:20

5aNSb6. Study on effect of dynamic lubricated bearing oil film stiffness on underwater structure acoustic radiation induced by propeller force. Jiayou Yang, Yipeng Cao, and Liaoyuan Li (College of Power and Energy Engineering, Harbin Engineering University, Harbin 150001, China, yangjiayou1986@163.com)

When propeller runs in the unsteady wake flow field, the exciting force can form in the propeller blades. Such propeller exciting force transfers from the shafting, bearing supporting to the structure, causes the underwater structure strong vibration and noise. The vibration of propeller-shafting-structure coupled system is one of the main sources of underwater structure radiated noise now. Different oil film stiffness will affect the input power transmission characteristics of coupled system severely, and then affect the vibration and noise analysis of underwater structure. In this paper, the finite element model of double cylindrical shell structure with shafting is built. The oil film stiffness is got by solving two-dimensional Reynolds equation based on finite difference method. The vibration and underwater noise radiation of the shell structure induced by propeller excite force are calculated by FEM/BEM. Finally, the effect of dynamic lubricated bearing oil film stiffness on underwater structure acoustic radiation induced by propeller force is analyzed.

11:40

5aNSb7. Study on hull vibration and underwater radiation noise induced by propeller excitation considering shafting alignment. Liaoyuan Li (College of Power and Energy Engineering, Harbin Engineering University, Harbin 150001, Heilongjiang, P.R. China, simple.life.lly@gmail.com), Yipeng Cao, and Wenping Zhang (College of Power and Energy Engineering, Harbin Engineering University, Harbin, Heilongjiang)

The recent increases in the number, size, speed and horsepower of commercial ships led the ocean ambient noise levels at low frequencies increased. Accordingly, the analysis of ship hull vibration control and noise reduction needs to be paid more attention now. In this paper, a finite element model of bulk carrier including shafting is built. Considering the shafting alignment, the vibration and underwater radiation noise characteristics of hull structure caused by propeller excite force are calculated. Finite element method (FEM) is employed to simulate the vibration response of the hull structure due to the propeller excitation in consideration of fluid-structure interaction. Then the outer surface of the finite element model is treated as a boundary element model. Finite element solutions of the hull surface vibratory velocity are further used as velocity boundary condition of the hull boundary element model for consequent underwater radiated noise calculation with boundary element method (BEM). In addition, the characteristics of transfer force and radiated sound power of hull structure are obtained. Comparing with the numerical results of considering shafting alignment and without, the effect of shafting alignment on propeller-induced ship hull vibration and underwater radiation noise is analyzed.

12:00

5aNSb8. A method for predicting the vibrations of underwater cylindrical shells by using the equivalent beam model. Tang Rui (Harbin Engineering University, Underwater Acousitc Building, Room 1205, tangrui@hrbeu.edu.cn), Li Qi (Harbin Engineering University, Underwater Acousitc Building, Room 301), and Shang Dejiang (Harbin Engineering University, Underwater Acousitc Building, Room 1206)

The vibrations of cylindrical shells with large length-to-radius radio are similar to those of beams in very low frequency band. One-dimensional beam theoretical model is only considered about the transverse displacement, neglecting the coupling effects of displacements in other directions. Only the first order beam-type modal frequency, not the other higher orders, of cylindrical shells can be predicted precisely by one-dimensional beam theoretical model. To solve this problem, an equivalent beam theoretical model is established, based on Euler-Bernoulli beam theory, in this paper. In this model, a cylindrical shell model is considered as a beam model with the same structural parameters and boundary conditions, the interaction between the structures and water is approximated to added mass. Different Young's modulus values have been searched to make the calculated results identical to those obtained by cylindrical shell theoretical model. The results show that the beam-type modal frequencies are mainly dominated by the length-to-radius radio for shells with length-to-radius radio $L/a>10$ and radius-to-thickness radio $a/h>20$, and the effect of the shell thickness to the modal frequencies can be neglected in such conditions. The equivalent Young's modulus curves for the first five order beam-type modal frequencies of cylindrical shells with different length-to-radius radio have been calculated.

12:20

5aNSb9. Mitigation of low-frequency underwater noise generated by rotating machinery on a mobile work barge using large tethered encapsulated bubbles. Kevin M. Lee, Mark S. Wochner (Applied Res. Labs., The University of Texas at Austin, Austin, TX 78713-8029, klee@physics.utexas.edu), and Preston S. Wilson (Mech. Eng. Dept. and Applied Res. Labs., The University of Texas at Austin, Austin, TX 78712-0292)

Collections of bubbles cause significant dispersion and attenuation of underwater sound at frequencies near the individual bubble resonance and can potentially be used to mitigate low-frequency anthropogenic underwater

noise. Such effects have been reported for large encapsulated bubbles with resonance frequencies below 100 Hz [J. Acoust. Soc. Am. **130**:3325-3332 (2011)] and significant attenuation due to bubble resonance phenomena and acoustic impedance mismatching was observed in experiments using a compact electromechanical acoustic source [J. Acoust. Soc. Am. **128**:2279 (2010); J. Acoust. Soc. Am. **129**:2462 (2011)]. In the present study, screens of tethered resonant encapsulated air bubbles were used to surround a mechanically-vibrated mobile work barge in a lake experiment to demonstrate their potential as a mitigation strategy for such noise sources. Conventional screens of freely-rising bubbles were also deployed for comparison. The radiated acoustic pressure was measured at various water depths and ranges to determine the effect of the bubble screens on the noise field. Compared to the freely rising bubbles, the tethered encapsulated bubbles yielded a significant increase in noise reduction below 1 kHz, demonstrating their efficacy for abatement of low-frequency underwater rotating machinery noise. [Work supported by Shell Global Solutions.]

FRIDAY MORNING, 18 MAY 2012 — HALL B, 9:20 A.M. TO 12:40 P.M.

Session 5aNSc

Noise and Architectural Acoustics: Noise Effects on Occupant Comfort and Performance in Buildings I

Lily Wang, Cochair
lwang4@unl.edu

C. M. Mak, Cochair
becmmak@polyu.edu.hk

Invited Papers

9:20

5aNSc1. The long-term effects of aircraft noise exposure on children's cognition: findings from the UK RANCH follow-up study. Charlotte Clark (Barts & the London School of Medicine, Queen Mary University of London, London, EC1M 6BQ, UK, c.clark@qmul.ac.uk), Jenny Head (University College London 1-19 Torrington Place London, WC1E 7HB, UK), and Stephen Stansfeld (Barts & the London School of Medicine, Queen Mary University of London, London, EC1M 6BQ, UK)

Many studies have demonstrated that environmental noise exposure at school is associated with poorer reading ability. However, studies have yet to determine the long-term consequences of environmental noise exposure during school for later developmental outcomes. This paper presents a longitudinal follow-up of the UK RANCH cohort assessing whether aircraft noise exposure in primary school is associated with poorer reading comprehension in secondary school. The original RANCH study found a relationship between aircraft noise exposure at primary school and children's reading comprehension for 9-10 year old children attending schools around London Heathrow, Amsterdam Schiphol and Madrid Barajas airports. Six years after the original study, 461 participants aged 15-16y (48% of the UK baseline sample) completed a standardised reading test during group testing in their classroom. Aircraft noise exposure at school was assessed using Leaq16 contour data. Analyses found no significant effect of aircraft noise exposure at primary school or secondary school on reading comprehension assessed at secondary school, although there was a trend for both types of noise exposure to be associated with poorer reading comprehension. The study is limited by its small sample but the findings indicate that aircraft noise exposure at school could have long-term implications for children's cognitive development.

9:40

5aNSc2. The effects of intelligibility and variability on cognitive performance and acoustic comfort. Andreas Liebl (Fraunhofer Institute for Building Physics, Nobelstrasse 12, 70569 Suttgart, Germany, andreas.liebl@ibp.fraunhofer.de)

Open-plan offices are very popular due to some assumed economic and organizational advantages. These benefits are confronted with typical drawbacks, i.e. office noise and lack of speech privacy. Ambient speech caused by conversations of colleagues is assumed to impair office workers' individual task performance at silent, concentrated work. The Speech Transmission Index (STI) is proposed to be a predictor of how much performance is reduced due to ambient speech in dependency of its intelligibility (Hongisto, 2005). Target values for the acoustic quality of open-plan offices were defined which account for the special importance of speech intelligibility (e.g. Virjonen et al., 2009). Fluctuation strength, a hearing impression due to slow modulations of amplitude or frequency, has also been shown to predict the loss of individual task performance (Schlittmeier et al. 2011). The relevance of both variables is systematically explored with regard to cognitive performance and acoustic comfort. Based on these results target values are discussed.

10:00

5aNSc3. Temporal and spatial characteristics of noise in train stations. Yoshiharu Soeta (Health Research Institute, National Institute of Advanced Industrial Science and Technology (AIST), 563-8577, y.soeta@aist.go.jp), and Ryota Shimokura (Nara Medical University, 634-8522, Japan)

The train noise in station (TNIS) can annoy passengers and reduce the speech intelligibility of the public address system in a station. Thus clarifying the acoustical characteristics of TNIS is important for the comfort and safety. Noise level was evaluated the equivalent continuous sound pressure level (LAeq). The sound diffuseness was evaluated by the interaural cross-correlation coefficient (IACC) from the interaural cross-correlation function. The pitch and pitch strength was evaluated by the delay time (t1) and amplitude (f1) of the first peak of the autocorrelation function. The LAeq and IACC of the TNIS in the underground station were higher and lower, respectively, than those in the ground station. The t1 and f1 of the TNIS in the underground station were stable and lower, respectively, than those in the ground station. When the train came into the station, the LAeq of the TNIS at the island platform were higher than that at the side platform. The result suggests that the underground island platform, which is one platform at the center and the rail tracks on the both sides, needs improvement in particular although it is most frequently adopted as newly constructed stations in Japan.

10:20

5aNSc4. The quiet office—noise abatement in office buildings by means of smart structures. Thilo Bein (Fraunhofer LBF, Bartningstr. 47, 64289 Darmstadt, thilo.bein@lbf.fraunhofer.de), and Joachim Bös (TU Darmstadt, Chair of System Relibaility and Machine Acoustics, Magdalenenstr. 4, 64289 Darmstadt)

Noise is a serious form of environmental pollution believed to affect the lives of some 100 million European citizens. The cost of the associated damage is estimated at more than ten billion euro per year. Noise leads to serious health problems, limits the capability to learn and effects the occupant comfort and performance in buildings. In this context, advanced noise abatement concepts are being developed in the demonstration project "Quiet Office" within the LOEWE-Center AdRIA (Adaptronics – Research, Innovation, Application), a large interdisciplinary research project funded by the German federal state of Hessia. As underlying principle for noise reduction concepts, Active Structural Acoustic Control (ASAC) is primarily being considered. Applying ASAC concepts, the noise radiation is controlled either by controlling the structural vibration of the radiating structure or by controlling the structure borne sound path. This paper will present the most recent concepts and results from the "Quiet Office". Starting from more generic concepts, concepts specifically designed for office applications including smart windows will be discussed. Examples presented will include distributed absorbers, shunt technologies with piezoelectric ceramics, smart Helmholtz resonators and fully active noise and vibration control.

10:40–11:00 Break

11:00

5aNSc5. Noise reduction by eaves/louvers attached on façade of high-rise buildings. Shinichi Sakamoto (Institute of Industrial Science, The University of Tokyo, sakamo@iis.u-tokyo.ac.jp), and Takumi Asakura

The authors have been investigating noise reduction effects of several types of eaves/louvers attached on façade of high-rise buildings. The eaves are originally used for the aim of shading of solar radiation or as structures for fire-proofing in Japan, but the devices can also provide high effectiveness of noise reduction against road traffic. The authors have revealed such effects through wave-based numerical analyses, scale model experiments and in-situ experiment. The eaves attached on the façade work as either noise shielding materials or noise reflectors, so inclined eaves at higher rooms have more effectiveness of noise reduction. The effectiveness varies with protrusion length of the eave, inclination angle, height of the room and so on. In this paper, such results of our investigation on noise reduction by eaves and louvers will be presented.

11:20

5aNSc6. On site performances of ventilation windows. H.K. Wong, C.N. Yung (Housing Department, Hong Kong SAR Government, hungkeung.wong@housingauthority.gov.hk), and S.K. Tang (Department of Building Services Engineering, The Hong Kong Polytechnic University)

The present study investigates the performance, in term of sound attenuation, of ventilation windows when applied to a real situation. A full scale model consisted of two identical modular public housing residential flats was built up next to a very busy truck road with traffic noise level LA10 exceeding 80 dBA in general. One of these model flats was equipped with two ventilation windows, while the other with conventional windows having side-hung window panes. Noise measurements were carried out simultaneously inside both flats and at the façades of these model flats. The total conventional window opening size was kept at that of minimum prescribed area for ventilation (1/16 of floor area in Hong Kong). The measurements at the façades outside provided the necessary adjustments if necessary whereas the sound attenuation is the difference between the insertion loss of the ventilation window and the conventional window. The present results show that the current setup of the ventilation window can provide sound attenuation of 6.6 dB without any sound absorption treatment inside the window. The sound attenuation rose up to 8.1dBA when the two vertical interior sides and the interior top surface of each ventilation window were lined with such materials.

11:40

5aNSc7. The acoustic and ventilation performance of new ventilated window design. Michelle Wang (Dept. of CSE, H.K. Polytechnic University, Humghom, Hong Kong, michelle.wong@connect.polyu.edu.hk), Catherine Hui, and C.F. Ng

Noise and air pollution problems become significantly in dense city such as Hong Kong, since buildings are usual located close to the heavy traffic lines. Traditional openable window cannot fulfill all the functions of noise reduction, lighting and natural ventilation. A new ventilated window, which combines the multiple quarter-wave resonators (silencer) with the new wing wall design, is designed to balance between acoustic and ventilation performances. Furthermore, the use of multiple-wave resonators to replace absorption material can enhance the durability; avoid small particle emission and toxic gas due to fire. The acoustic and ventilation performance of new ventilated window were examined in this study. Noise attenuation of the new ventilated window design is improved significantly by combining flexible absorber and quarter-wave resonator effects. The test methods in acoustics are in accordance with relevant ISO 140 procedures.Transmission loss of 10 dB to 22 dB can be achieved in the frequency range of 500 Hz to 4k Hz band. The best ventilation performance of the wing wall is at an incident angle of 45°. Outlet air flow velocity of ventilated window design is double of the velocity of "open window". Thus, both the acoustics and ventilation performance of the new ventilated window is effective. Wind-driven natural ventilation is an effective way in maintaining the comfort and health of indoor environment.

Contributed Papers

12:00

5aNSc8. Influence of music performance on Sun Yat-sen Memorial Hall. Linqiang Gu and Weixin Lin (Department of Architecture and Urban Planning, Guangzhou University, 510006, China, gulinqiang@gmail.com)

Currently, many heritage buildings have been converted into music performance venues. Performing at ultra high sound pressure has induced coupled vibration of building elements which should be of protected, then caused irreversible damage of them after a certain time, such that application requirements and cultural relic protection formed an inevitable contradiction. Research on a case study of Guangzhou's Sun Yat - sen Memorial Hall in China demonstrated that music performance has little impact on the structure safety but a passive influence on some heritage building elements like colored glass and some murals, gave a reference for selection and design of damping measures, and at the same time pointed out that some issues required further study in this area.

12:20

5aNSc9. Acoustic design criteria in green building rating systems. May Han Grace Kwok (Allied Environmental Consultants Ltd. 19/F, Kwan Chart Tower, 6 Tonnochy Road, Wanchai, Hong Kong, gk@aechk.com)

Acoustic design aspect has been considered in various green building rating systems as part of the sustainable building design approach in an attempt to ensure a desirable acoustic environment for the future occupants. Primarily, the focus is on speech privacy, background noise control and noise isolation in a general building setting. In this paper, the acoustic design criteria adopted by different green building rating systems, including LEED in the United States, China Green Building Label in China, BREEAM in Britain, Green Star in Australia, CASBEE in Japan, BEAM Plus in Hong Kong, are evaluated in a comparative study with pros and cons highlighted. Selected criteria are investigated against respective regulatory requirements and common acoustic design standards and practices. The potential implications on building design and construction as well as the associated benefits to building users are also discussed. Possible conflicts and synergy with other sustainable design aspects in the green building rating systems are addressed. Lastly, acoustic design strategies and post-occupancy acoustic surveys for green buildings are recommended.

FRIDAY MORNING, 18 MAY 2012 S224 + S225, 9:20 A.M. TO 12:40 P.M.

Session 5aPA

Physical Acoustics: Sound Generation, Propagation, and Scattering (Lecture/Poster Session)

Constantin Coussios, Cochair
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Contributed Papers

9:20

5aPA1. Formulation and selected applications of a regularized elastodynamics integral equation approach to large scale scattering problems involving inhomogeneous objects. Elizabeth Bleszynski, Marek Bleszynski, and Thomas Jaroszewicz (Monopole Research, Thousand Oaks, CA 91360, elizabeth@monopoleresearch.com)

Elements are presented of the formulation and selected applications of the combined surface and volume integral-equation approach for finding pressure, displacement, and traction fields in composite objects consisting of piece-wise homogeneous regions characterized by different Lame's material parameters, as well as of highly inhomogeneous regions. The proposed elastodynamics surface/volume-based integral equations, in which the unknowns are volumetric pressure or displacement variables and surface displacement and traction fields on the interfaces, involve only weakly singular integral-equation kernels with at most 1/r singularity. The proposed method is found to provide well-convergent and accurate solutions for a large spectrum of problems, in particular those involving high-contrast material interfaces. The proposed method's accuracy and its capability (achieved through the (non-lossy FFT-based) Adaptive Integral Method matrix compression) of handling large-scale problems involving several millions of unknowns, is demonstrated through comparison with analytically constructed solution for a layered material sphere and in selected numerical simulations of propagation of acousto-elastic waves in human head and inner-ear region through air- and bone-conduction pathways. The proposed approach can be used in assessing effectiveness of noise-protection devices. This work is supported by a grant from the Air Force Office of Scientific Research.

9:40

5aPA2. Diffuse ultrasonic backscatter in two-phase media. Dalie Liu (Dept. of Mechanical Engineering, Zhejiang A&F University, 88 Huancheng North Road, Lin'an, Hangzhou, Zhejiang Province, 311300, China, dalieliu@msn.com), and Joseph Turner (Dept. of Mechanical & Materials Engineering, W 317.4 Nebraska Hall, Univ. of Nebraska-Lincoln, Lincoln, NE 68588-0526)

The microstructure of sintered metals and ceramics may be modeled as two phases consisting of the particles and a surrounding matrix during certain processing stages. Diffuse ultrasonic backscatter has been widely used to extract the microstructural parameters and also to detect flaws in such materials. In this research, a singly-scattered response model is used to predict the grain noise for the two-phase model. The expression of backscatter coefficient is derived and found to be dependent on the spatial correlations of the material properties. The geometric two-point correlation function plays an important role in this model and can be determined from numerical correlation statistics. Good agreement is found between the results determined from the model and the experimental results.

10:00

5aPA3. Mass flow and surface wave in a vertically vibrated granular system. Hui Cai, Weizhong Chen, and Guoqing Miao (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, mg0723001@smail.nju.edu.cn)

We report an experimental finding on coexistence of the horizontal mass flow and the vertical surface wave in a vertically vibrated granular system. The container is an annular channel with a sawtooth-shaped base which is excited and controlled by a vibration system. The particles are copper spheres with diameter of 3 mm. The driving frequency f and dimensionless acceleration amplitude $\Gamma=4\pi^2 f^2 A/g$ (A is the driving amplitude and g the gravitational acceleration) are used as two control parameters. A high-speed image recording system is used to record the movements of all spheres, and an image processing technique is used to track the locations of all particles. As Γ increases beyond a critical value (about 1.8), the granular layer fluidizes from top to lower parts, and the horizontal mass flow appears. The velocity of the mass flow increases with Γ, and from bottom to top decreases monotonously. In the meanwhile, if Γ exceeds 2.5, the vertical surface wave occurs. The amplitude of the surface wave increases with driving amplitude. The mechanism of the mass flow is the ratchet effect of the sawtooth, while the correlation between the mass flow and the surface wave is to be explored.

10:20

5aPA4. Acoustic-gravity waves in ocean and atmosphere generated by an underwater source. Iosif Fuks (NOAA Earth System Research Laboratory and Zel Technologies LLC, Mail Code R/PSD-99, 325 Broadway, Boulder, CO 80305, Iosif.Fuks@noaa.gov), and Oleg A. Godin (NOAA Earth System Research Laboratory, Physical Sciences Division and CIRES, University of Colorado at Boulder, Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305)

Air-water interface becomes anomalously transparent, and the power flux in the wave transmitted into the air increases dramatically, when a compact sound source in water approaches the interface within a fraction of wavelength [O.A. Godin, Phys. Rev. Lett. **97**, 164301 (2006)]. The anomalous transparency of the ocean-atmosphere interface has important implications for detection of underwater explosions and monitoring of compliance with the Comprehensive Nuclear Test Ban Treaty. At wave frequencies below 0.1 Hz, it becomes necessary to take gravity into account. Then fluid buoyancy and compressibility simultaneously serve as restoring forces, and mechanical perturbations in the water and in the air propagate as acoustic-gravity waves (AGW). It was previously shown [I. Fuks and O.A. Godin, Proc. OCEANS'11, MTS/IEEE, Kona, HI, Sept. 2011] that, in the case of a shallow source in an ocean of an infinite depth, a sharp peak in the power flux into air appears at frequencies close to a cutoff frequency of about 4mHz of a surface acoustic-gravity wave. In this paper, we extend these results to the ocean of a finite depth where the AGWs interact with an elastic bottom.

10:40–11:00 Break

11:00

5aPA5. Micorcrack localization in pipelines using nonlinear guided waves combined with time reversal. Xiasheng Guo, Di Yang, and Dong Zhang (Key Laboratory of Modern Acoustics (MOE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, guoxs@nju.edu.cn)

A three-dimensional (3D) micro-crack imaging technique is proposed for pipeline inspections. With non-classical nonlinear guided waves generated from micro-cracks recorded, the third harmonic waves are used to image the fatigued crack, with the help of time reversal process. A finite-difference time-domain (FDTD) code is programmed to solve the wave equations under cylindrical coordinates, and simulate the wave propagation process. A defect with hysteretic stress-strain behavior is embedded in the model; its interaction with guided waves generates nonlinear components, the harmonic components are time reversed and played back into the pipe, hence produce the image of the cracked area. The results show excellent spatial retrofocusing capability. The accuracy of localization depends on crack orientation angle and an adopted guided wave mode. [This work is supported by the National Natural Science Foundation of China (11104140) and Natural Science Foundation of Jiangsu Province (BK2011543)]

11:20

5aPA6. Characteristics of thickness-shear modes excited by two layers of piezoelectric films in acoustic sensors. Hui Zhang, Shu-yi Zhang, Li Fan, and Yu-ran Wang (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, paslabw@nju.edu.cn)

The thickness-shear mode (TSM) excited by two layers of (112(–)0) textured hexagonal piezoelectric films are studied to improve the performances of acoustic sensors. The corresponding researches are carried out by the acoustic wave propagation model which considers the effects of the acoustic attenuations. When the acoustic field and electric field satisfy to a special match condition, i.e. the phase variation of TSM at each film equal to π, both piezoelectric layers with opposite polarization directions reduce the first TSM and generate the second TSM with higher frequency and higher quality factor than the first TSM excited by two-layers of piezoelectric films with identical polarization direction. Furthermore, the second enhanced TSM holds a moderate the ratio of the operating frequency to the quality factor. These properties of the TSM are appropriate for improving the mass sensitivity and sensitive resolution of the acoustic sensors in liquid medium. Acknowledgment: This work is supported by National Natural Science Foundation of China (No. 11004099 and 11174142), State Key Laboratory of Acoustics of Chinese Academy of Sciences, and also PAPD of Jiangsu Higher Education Institutions.

11:40

5aPA7. Theoretical study of interaction of the acoustic waves in different crystals. Xiaozhou Liu, Tingting Zhang, and Xiufen Gong (Key Lab of Modern Acoustics, Institute of Acoustics, Nanjing University, Nanjing 210093, China, xzliu@nju.edu.cn)

In isotropic solids, transverse acoustic waves propagate undistorted and the interaction of collinear transverse waves is forbidden. In this study, It is found that transverse and longitude waves behave differently in anisotropic solids and the different formulas are given for the interaction of transverse and longitude acoustic waves propagating in different kinds of crystals including cubic, trigonal, tetragonal, hexagonal crystals. The formulas are expressed in terms of the derivatives of the strain energy. The formulas are used to calculate the interaction coefficients in the general anisotropic medium emphasizing the influence of nonlinearities. The evolution equations are derived for the wave amplitudes and find analytical formulas for all interaction coefficients of quadratically nonlinear interacting waves. The equations of motion show that transverse and longitude waves are distort as they propagate and interact. It is observed that two acoustic waves with frequency can produce harmonic waves with sum and difference frequency. This shows evidence that the intrinsic nonlinearities of an anisotropic solid can induce an interaction of acoustic waves. This project is supported the National Basic Research Program of China (Grant Nos. 2012CB921504, 2011CB707902), financial support of the National Natural Science Foundation of China (Grant No. 11074122), fundamental Research Funds for the Central Universities (Grant Nos. 1113020403, 1101020402), State Key Laboratory of Acoustics, Chinese Academy of China (Grant No. SKLOA201005) and A project funded by the priority academic program development of Jiangsu higher education institutions.

The following abstracts will be presented in poster format. The posters will be on display and the authors will be at their posters from 12:00 noon to 12:40 p.m.

5aPA8. Optimization and limitations of a preconditioned multi-level fast multipole algorithm for acoustical calculations. Ralf Burgschweiger, Martin Ochmann (Beuth Hochschule für Technik Berlin, University of Applied Sciences, Luxemburger Str. 10, 13353 Berlin, Germany, burgi@beuth-hochschule.de), Ingo Schäfer, and Bodo Nolte (Federal Armed Forces Underwater Acoustics and Marine Geophysics Research Institute, Klausdorfer Weg 2-24, 24148 Kiel, Germany)

The Multi-Level Fast Multipole Method (MLFMM) allows the computation of acoustical problems based on the Boundary Element Method (BEM) where the discretized models of the corresponding structures may consist of a huge number of elements. The required calculation time and the memory requirements are much less when compared with conventional methods because the algorithm uses a level-based composition of the potentials from different point sources to acoustic multipoles, which highly accelerates the computation of the matrix-vector-products required for iterative solvers. A multi-level single-order variation of the algorithm was extended to a multi-level adaptive-order version, which was analyzed and optimized with respect to quality, performance and parallelization issues. The iterative solvers used with the MLFMM will be combined with appropriate preconditioners for reducing the number of iterations and improving the performance. The insights gained will be presented using different test cases and the results achieved will be compared with analytical solutions and results of conventional BEM- and FEM-based calculations.

5aPA9. Reduced acoustic cloaks: theoretical analysis and numerical simulation. Yuxian Fan (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, fanyx@mail.ioa.ac.cn), Wenlin Hu, Peifeng Ji, and Jun Yang

Acoustic cloaking is a new technique which can make an object invisible in acoustic waves. This method of controlling and directing sound has a promising prospect in application. Rigorous physical parameters are required for perfect cloaking, which is unpractical for experimental realization. Reduction is an alternative when imperfect cloaking performance is acceptable. In this paper, theoretical analysis of reduced acoustic cloaks based on transformation acoustics is presented. Fundamental features of acoustic cloaking are derived and discussed. Layered medium theory is employed to design reduced acoustic cloaks. Reasonable physical parameters and tolerable cloaking performance are expected in the prerequisite of the work. The properties of the designed reduced acoustic cloaks are studied. Numerical simulation shows that the reduced acoustic cloaks have effective cloaking performance.

5aPA10. Research on experiments for high frequency sound propagation with rough surface in shallow water. Xudong An (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, anxudong@mail.ioa.ac.cn)

A experiment was carried out to research high frequency underwater sound propagation with rough surface in shallow water. A wave-maker was used to simulate the sine wave surface with different wavelength, amplitude and cycle in a pool (108m$\times$7m$\times$3.5m). The multiple propagation tracks and the multiple reflections at surface and bottom as well as focusing and defocusing due to reflection from surface waves were measured and analyzed. The accuracy and efficient of experiment are verified by the agreement between the numerical simulation and the analysis results of the measured experimental data.

5aPA11. Acoustical emission of jet-edge systems using a vibrating element in water. C Li, Jingjun Deng, and Lixin Bai (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing 100190, China, lichao@mail.ioa.ac.cn)

The jet-edge system using a vibrating element (liquid whistle) was investigated experimental using high-speed camera. The vibrating element vibrated periodically with a few millimeters displacement and cavitaion bubbles were observed near the tip of the vibrating element. The acoustic emission of the system was measured with hydrophones in some place. The low frequency and shock signals were observed in the pressure signal, and whistle can be heard at the same time. In frequency field, the line and broadband spectrum exist. The foundation frequency f0 of the vibrating element and harmonics compose the line spectrum. The former 10 order harmonics can be seen obviously. As the flow velocity increasing, the frequency f0 and the harmonic do not change, but the intensity of the high order harmonics and broadband spectrum become strong. Base the experimental result, we consider that the f0 is the foundation frequency of the vibrating element, and the harmonics and the continuous spectrum is the bubble cavitation emission. Acknowledgement: This work was supported by the National Natural Sciences Foundation of China (10804118)and he National Science and Technology Major Project of China (No. 2011ZX05032-003)

Session 5aPP

Psychological and Physiological Acoustics: Auditory Function, Mechanisms, and Models (Poster Session)

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Bradley Mcpherson, Cochair
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Contributed Papers

All posters will be on display from 9:20 a.m. to 12:40 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:20 a.m. to 10:40 a.m. and contributors of even-numbered papers will be at their posters from 10:40 a.m. to 12:40 p.m.

5aPP1. Electrophysiologic assessment of (central) auditory processing disorder in children with non-syndromic cleft lip and/or palate. Xiaoran Ma (B092, 5F, Division of Speech and Hearing Sciences, 34 Hospital Road, The Prince Philip Dental Hospital, Sai Ying Pun, Hong Kong, xiaoran@hku.hk), Lian Ma (School of Stomatology, Beijing University), and Bradley McPherson (5F, Division of Speech and Hearing Sciences, 34 Hospital Road, The Prince Philip Dental Hospital, Sai Ying Pun, Hong Kong)

Cleft of the lip and/or palate is a common congenital craniofacial malformation worldwide, particularly non-syndromic cleft lip and/or palate (NSCL/P). Though middle ear deficits in this population have been universally noted in numerous studies, other auditory problems including inner ear deficits or cortical dysfunction are rarely reported. A higher prevalence of educational problems has been noted in children with NSCL/P compared to craniofacially normal children. These high level cognitive difficulties cannot be entirely attributed to peripheral hearing loss. Recently it has been suggested that children with NSCLP may be more prone to abnormalities in the auditory cortex. The aim of the present study was to investigate whether school age children with (NSCL/P) have a higher prevalence of indications of (central) auditory processing disorder [(C)APD] compared to normal age matched controls when assessed using auditory event-related potential (ERP) techniques. School children (6 to 15 years) with NSCL/P and normal controls with matched age and gender were recruited. Auditory ERP recordings included auditory brainstem response and late event-related potentials, including the P1-N1-P2 complex and P300 waveforms. Initial findings from the present study are presented and their implications for further research in this area — and clinical intervention — are outlined.

5aPP2. Change deafness with recognizable and unrecognizable sounds. Vanessa Irsik (UNLV- 6151 Mountain Vista #527 Henderson, Nv 89014, irsikv@unlv.nevada.edu), Melissa Gregg, and Joel Snyder (UNLV 4505 S. Maryland Parkway, P.O. Box 455030, Las Vegas, Nevada 89154-5030)

Change Deafness with Recognizable and Unrecognizable Sounds Change deafness is the remarkably frequent inability of listeners to detect changes occurring in their auditory environment. In this study, we used behavioral measures and event-related potentials (ERPs) to determine if change deafness is a fundamental auditory sensory process, rather than simply a reflection of verbal or semantic memory limitations. Change detection performance was examined for scenes composed of four recognizable or unrecognizable sounds. Listeners completed a change detection task by making a same/different judgment for two consecutively presented scenes that were either the same (Same trials) or had one sound replaced by another sound (Change trials). The behavioral data indicated substantial change deafness for both recognizable and unrecognizable sounds, indicating that change deafness is not the result of verbal or semantic memory limitations. In Change trials, N1 ERPs were less smaller in the post-change scene on trials in which the change was not detected, suggesting that change deafness is associated with less robust sensory encoding. P3 ERPs to the post-change scene were also smaller for non-detected changes, which may reflect lack of memory updating, attention, and/or awareness during change deafness. Overall, the results provide novel information regarding the stages of processing involved in change deafness in natural auditory scenes.

5aPP3. Attentional modulation of EEG signals. Inyong Choi (Center for Computational Neuroscience and Neural Technology, Boston University, 677 Beacon St., Boston, MA 02215, iychoi@bu.edu), Hari Bharadwaj (Department of Biomedical Engineering, Boston University), and Barbara Shinn-Cunningham (Center for Computational Neuroscience and Neural Technology, Boston University, 677 Beacon St., Boston, MA 02215)

Physiological measures revealing effects of selective attention are needed to develop insights into how auditory attention is controlled. Here, we used electroencephalography (EEG) to measure neural activity when listeners attended one of two competing auditory streams. Listeners identified whether a target three-tone melody had a rising, falling, or non-monotonic pitch contour while ignoring a competing, simultaneous melody. To enhance stream segregation, the streams differed in timbres (oboe and cello) and perceived lateral position (to the left, to the right). We "tagged" each of the streams by adding amplitude modulation (AM) at different gamma-band frequencies. We found that the evoked gamma activity at the AM frequencies was statistically above the noise floor, and depended on the spatial configuration and the modulation frequency, as might be expected. Importantly, for identical acoustic inputs, the relative strength of the AM frequencies depended on which stream a listener attended. Moreover, the pattern of responses depended on what cue directed the listener to attend to a particular target. Results suggest that the control of attention engages different neural networks, depending on the feature directing attention. These results will be discussed in light of efforts to develop brain-machine interfaces that monitor auditory attention.

5aPP4. What is so hard about selectively attending? Ross K Maddox, Willy Cheung, and Adrian KC Lee (University of Washington, Portage Bay Building, P.O. Box 357988, 1715 Columbia Road North, Seattle, WA 98195, rkmaddox@uw.edu)

As the number of elements that make up an auditory scene increases, it becomes harder to selectively attend just one of those elements. Previously, the limit of listeners' abilities to attend a target stream of repeating letters in an overcrowded scene was tested. In that experiment, each stream consisted of a repeated monotonized and localized spoken letter (an "item"), with a repetition period of 1 s. Among streams, item onset times were distributed across each repetition. Listeners were asked to detect when the attended target letter changed to an oddball "R" for a single repetition, ignoring such occurrences in the non-target streams. With a constant repetition period, adding streams to the stimulus meant that the number of items per second increased proportionally. The decrease in performance could thus be a result of having more streams in the scene, or because of the increased item rate. Here, a similar experiment was performed, holding the item rate constant, rather than the repetition period. The results allow us to disentangle the effects of the number of distractor streams and the item rate, yielding insight into the specific reasons for the diminished ability to selectively attend. Funded by USA-NIH-T32DC009975 (RKM) and R00DC010196 (AKCL).

5aPP5. Effects of spatial attention on across-frequency grouping in speech. Miguel Cepeda, Virginia Best, and Barbara Shinn-Cunningham (Boston University 677 Beacon Street Boston, MA 02215, mdcepeda@bu.edu)

To understand speech, listeners integrate information across disparate frequency regions. Such integration may depend not only on low-level grouping cues (e.g., correlations in envelope across frequency) but also on volitional attention (e.g., integrating only frequency bands from a desired direction). Here, listeners reported the content of a target sentence conveyed by two spectrally separated, narrow bands of speech presented to the left ear. Two additional speech bands, centered between the other bands, were presented either to the left or the right ear. These additional bands were either from the target speech (matched) or an independent sentence (conflicting). In the key experimental block, listeners were instructed to attend to the speech on the left while either conflicting bands or, infrequently, matched bands were presented on the right. Splitting the target across the ears degraded intelligibility, showing that spatial separation interferes with grouping. However, directed spatial attention had no effect: intelligibility was equivalent when listeners explicitly were told to attend to the target bands split across both ears and equivalent trials in which listeners volitionally attended only the left ear. Results show that for structure-rich speech, directed attention does not overcome automatic grouping processes: across-frequency grouping of related speech bands is obligatory.

5aPP6. Functional subnetwork structure in auditory cortex for stream segregation. Takahiro Noda, Ryohei Kanzaki, and Hirokazu Takahashi (University of Tokyo, 153-8904, noda@brain.imi.i.u-tokyo.ac.jp)

Perceptual integration and segregation of alternating tone sequence differing in frequency (ABA-ABA-...) depend on the frequency differences (Delta-Fs) between A and B tones and the inter-tone intervals (ITIs) between successive tones. In the auditory cortex, tonotopic separation, forward suppression and multisecond habituation have been considered as possible neural correlates of this perceptual phenomenon. This model, however, cannot completely account for the van Noorden's perceptual boundary and the temporally continuous perception of auditory streaming. Here we examined the temporal changes of the functional network properties in auditory cortex to tone sequences with different Delta-Fs and ITIs. Specifically, we recorded local field potentials using microelectrode arrays from anesthetized and awake rats and constructed the functional network based on phase synchrony in gamma-band oscillation. As the results, the networks consisted of sub-networks highly correlated with tonotopy, and the sub-network selective to B tones lasted for a prolonged period at large Delta-F. Such characteristic substructures of functional network are a possible candidate of neural mechanisms of auditory stream segregation.

5aPP7. Probing the cortical dynamics involved in ignoring a low-salience distracting sound. Eric Larson and Adrian KC Lee (University of Washington, Portage Bay Building, Room 204, 1715 Columbia Road North, Seattle, WA 98195, larsoner@uw.edu)

Verbal communication in many everyday settings requires listeners to ignore distracting auditory events. However, how the brain coordinates the suppression of even low-salience auditory distractors remains unclear. In studies of visual attention, it has been observed that the frontal eye fields (FEF) are involved in orienting and maintaining attention to targets of interest. With multiple fMRI studies implicating FEF in orienting auditory attention, we sought to examine the temporal dynamics of auditory attentional control in the presence of a low-salience distractor. We utilized a task that required listeners to attend to and report one of two simultaneous, but spatially-separated, spoken digits. A visual arrow-cue prompted listeners to report the stimulus originated from that hemifield. Shortly following the visual cue, we presented listeners a noise burst that either originated from the same hemifield or the opposite side as the visual cue. We utilized simultaneous magneto- and electro-encephalography to investigate the cortical dynamics involved in suppressing the potentially distracting noise burst. Using source-space analysis, we observed significantly stronger activation in left FEF when the noise burst was presented in the to-be-attended hemifield, providing further support that left FEF is specifically involved in maintaining auditory spatial attention. Funded by USA-NIH T32DC000018(EL) and R00DC010196(AKCL).

5aPP8. Effects of changes in the depth feeling of the visual target on the simultaneity perception of an auditory-visual event. Hiroshi Hasegawa, Tomoharu Ishikawa, Masao Kasuga, and Miyoshi Ayama (Utsunomiya Univ., hasegawa@is.utsunomiya-u.ac.jp)

In this study, we investigated the simultaneity perception between a visual stimulus and its associated sound. We carried out experiments of an auditory-visual stimulus presentation using an audio-video clip of a man beating a drum on a road. The visual stimulus had a feeling of depth with a perspective view of the road. There were four kinds of distance between the visual target of a man beating a drum and the video camera to capture the target of 5, 10, 20, and 40m, and we called these distances as "the presentation distances." We presented the auditory-visual stimuli to experimental subjects at each presentation distance of 5, 10, 20, and40 m under various conditions, where we varied the feeling of depth of the visual stimulus from -40% to 40% and the time delay between the auditory and visual stimulus from -8F to 8F (1F = 1/30s). We analyzed the experimental results and calculated the point of subjective simultaneity (PSS) of the auditory-visual event. As a result, the PSS intended to increase as the feeling of depth increased of the visual stimulus. This result shows that changes in the depth feeling of the visual stimulus could influence the simultaneity perception.

5aPP9. Robustness of audio-visual spatial disparity in peripheral field. Ryota Miyauchi (Japan Advanced Institute of Science and Technology, 1-1 Asahidai, Nomi, Ishikawa, 923-1292, Japan, ryota@jaist.ac.jp), Dea-Gee Kang, Yukio Iwaya, and Yôiti Suzuki (Research Institute of Electrical Communication, Tohoku University, 2-1-1 Katahira, Aoba-ku, Sendai 980-8577, Japan)

In our previous studies, a perceptual phenomenon, audio-visual peripheral spatial disparity (AVPSD), has been found. The spatial coincidence of perceived locations of a sound and flash presented in the peripheral field varies from their coincident physical location. Since the stimulus condition used in the previous experiments was limited, three new experiments were conducted to investigate the robustness of the spatial disparity between the perceptual and physical locations of the sound and flash. In Experiments 1 and 2, the frequency characteristics of the sound and the repetition of stimulus appearance were respectively changed from those used in the previous experiments. In Experiment 3, the presentation timing of the sound or the flash was delayed for one second. The results of Experiments 1 and 2 show that the spatial disparity was robustly appeared. In contrast, the spatial disparity disappeared in Experiment 3. These results suggest that the features of the stimulus pattern do not affect the spatial disparity, but the simultaneity of the sound and flash plays an essential role. Acknowledgment: This research was supported by a Grant-in-Aid for Specially Promoted Research (No. 19001004) and Grant-in-Aid for Young Scientists (B) (No. 21730584).

5aPP10. Visual deprivation improves auditory scene analysis. Marie-Soleil Houde, Marianne Bélanger, Douglas Shiller, and François Champoux (Université de Montréal, C.P. 6128, Succursale Centre-Ville, mariesoleilhoude@alumni.uottawa.ca)

The present study aims to examine the effect of temporary visual deprivation on a process central to auditory segregation, namely harmonicity. A harmonicity discrimination task was administered twice, with an interval of 90 minutes, in two groups of individuals. One group was temporary deprived of visual information during that interval period. Control conditions revealed that auditory capabilities were similar across groups. However, results in the

subsequent testing session revealed that temporarily deprived individuals showed a remarkable improvement in the auditory task in comparison with the non-visually-deprived group (i.e. control group). These results suggest that temporary blindness can improve auditory segregative processing.

5aPP11. Acoustical dummy head for Chinese adults. Na QI and Zihou MENG (Communication University of China, qina@cuc.edu.cn)

An acoustical dummy head is designed based on the Chinese national code of adult head geometry. From a survey of more than four hundreds Chinese adult auricle structure, a typical physiological structure of auricle is selected for the acoustic dummy head. Eyes, nose and other head structure are simplified. The HRTF of the acoustical dummy head is measured. The cues relating to sound source localization, including interaural time difference (ITD),interaural level difference (ILD), and spectral features introduced by pinna, nose, hair and other detail structure of head, are analyzed. And then the analytical data is compared with a full-scale dummy head. The result shows that the acoustical dummy head retains the main acoustic characteristics attributed to auditory localization. A listening test is carried out to verify this property.

5aPP12. Effects of spatial localization on sentence recognition in noise. Tatsuo Nakagawa (Yokohama National University, Faculty of Education and Human Sciences, 79-2 Tokiwadai, Hodogaya-ku, Yokohama, Japan, nakagawa@edhs.ynu.ac.jp)

Speech recognition in noise improves when speech and noise are separated in space. Interaural level and time difference cues are used in spatial localization. Most people with hearing loss have better hearing in the low frequencies. Interaural cues for sound localization are also more important in the low frequencies. An objective of the current experiment was to assess the effects of spatial separation of speech and noise on the recognition of low-pass filtered speech in persons with normal hearing and hearing loss. Speech levels corresponding to 50% correct recognition of sentences were measured in a 65dB SPL multi talker noise. Spatial benefits for listeners with and without hearing loss are discussed.

5aPP13. Effect of similarity between target speech and time-reversed masker on speech intelligibility. Bin Jiang and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, binjiang@mail.ioa.ac.cn)

The similarity between target speech and interfering signal may affect masking. Since time-reversed masker is an unintelligible speech-like signal, the influence of intelligibility can be avoided. This paper describes a sentence intelligibility test with target speech masked by different time-reversed maskers to evaluate the similarity effect. The time-reversed masker is processed by locally time-reversal segments of speech uttered by same-talker, same- and different-sex talker. The results show that the amount of masking is highly dependent on the similarity between target speech and time-reversed masker: time-reversed masker of target speech gives a rise in average speech reception threshold (SRT) of about 3 dB and 9 dB compared to that of another talker with same- and different-sex, respectively. Assuming the energetic masking effect for time-reversed maskers of different talkers to be similar, the amount of similarity with same-sex is larger than 6 dB and the amount of similarity with same-talker is larger than 9 dB.

5aPP14. Comparison of frequency lowering algorithms on Mandarin speech recognition. Yunyi Zhang (Institute of Acoustics, Chinese Academy of Sciences, zyy780@sina.com), Jie Cui, and Ling Xiao

This paper examined several frequency lowering algorithms for improving speech recognition in Chinese listeners with high-frequency sensorineural hearing loss, including both frequency compression and frequency transposition algorithms. The frequency lowering algorithms were evaluated by a subjective listening test. Monosyllabic words in Mandarin spoken by a female and a male talker were used as speech materials. Speech materials processed by frequency compression and frequency transposition were presented to normal-hearing listeners in the listening tests, as well as the control condition of unprocessed speech. More benefits in speech recognition were observed with speech spoken by female talker than by male talker. It was showed that frequency lowering made little difference for some words while words with dental consonant were most infected by frequency lowering. The results indicate that Mandarin recognition for high-frequency hearing loss may benefit from frequency lowering in some cases, while whether choosing frequency lowering or not and which algorithm to choose depends on individuals.

5aPP15. The effect of spectral asynchrony on speech recognition in demanding listening conditions. Magdalena Wojtczak, Sachin Rai, Jordan A. Beim, and Andrew J. Oxenham (University of Minnesota, N218 Elliott Hall, 75 East River Rd., Minneapolis, MN 55455, wojtc001@umn.edu)

Speech recognition is remarkably robust to overall spectral asynchronies as large as 160-180 ms when performance is measured in quiet. One reason for such robustness may be substantial redundancy in the speech stimulus. This study investigated the effect of spectral asynchrony on speech recognition in listening conditions yielding 80% or less correct recognition score for synchronized speech (baseline condition). IEEE sentences mixed with a speech-shaped noise or two-talker babble were divided into 1/3-octave bands. Successive bands were progressively delayed from low to high, or from high to low, center frequency. The overall delay between the lowest and highest-frequency channels was varied between 0 and 160 ms. The difficulty of the task in the synchronous condition was determined by different signal-to-noise ratios and by presenting either the full stimulus or a lowpass-filtered version. The results show that performance is less robust to spectral asynchrony when speech is presented with an interfering background, although it remains relatively unchanged for overall delays up to 40-ms, independent of the direction of the delay as a function of frequency. The results were analyzed using the speech transmission index. [Supported by NIH grant R01DC010374].

5aPP16. Auditory and cognitive factors in speech and environmental sound perception of cochlear implant listeners. Valeriy Shafiro (Rush University Medical Center, valeriy_shafiro@rush.edu), Stanley Sheft, Sejal Kuvadia, Brain Gygi, and Kim Ho

This study examined the role of acoustic pattern discrimination abilities and higher-order cognitive factors on the perception of spectrally degraded speech and environmental sounds. Over the course of four testing sessions and a week of environmental sound training, cochlear implant and normal-hearing listeners (tested with a 4-channel implant vocoder simulation), received a 160-item environmental sound test, and isolated word and sentence tests. Listeners were also tested with a battery of cognitive tests which included measures of inhibition, closure, working memory and executive function, and a battery of psychoacoustic tests of spectral and temporal pattern processing. A great deal of overlap in the processing abilities for speech and environmental sounds was found in both groups of listeners. Centrally, speech and environmental sounds were associated with working memory and executive function measures (i.e., digit-span-backward, letter-number sequence). Psychoacoustically, the two sound classes were associated with measures of spectral resolution and temporal fine-structure processing. These findings indicate that successful perception of acoustically complex meaningful patterns under conditions of spectral degradation involves joint contributions from specific peripheral factors and central processes, and suggest directions for improving listener performance through training [Support provided by NIH/NIDCD].

5aPP17. Word retrieval process uses phonemic cues in memory-impaired epilepsy analyzed by Rey's auditory verbal learning test. Keiko Asano (Juntendo University, School of Medicine, 1-11-2905 Kinkocho, Kanagawa-ku, Yokohama-City, Japan, keiko_asano@sakura.juntendo.ac.jp), Hidehiro Okura (Juntendo University, School of Medicine), Hidenori Sugano, Mitsuko Nakano, Keiko Fusegi, and Hajime Arai (Juntendo University, School of Medicine)

This study investigated what kinds of phonemic strategies memory-impaired patients with diagnoses such as frontal lobe epilepsy compared to the strategies healthy people use in order to memorize and retrieve words in the aspects of Rey's Auditory Verbal Learning Test (AVLT). This Auditory Verbal Learning Test is widely used in the field of clinical and neuropsychological assessments to examine word learning and memory. This test has two stages: A list of 16 unrelated, concrete nouns is presented over five learning trials with immediate recall tested following each presentation. After a delay interval of 30 minutes with no further presentations of the list, delayed recall is assessed. The number of words correctly recalled is commonly adopted as quantitative information in clinical assessment. After the tests, participants are asked what kinds of strategies they used to memorize the words in terms of examining the process of decoding, recall and retrieval. Spontaneous memorization strategies used by memory-impaired patients are visual-oriented and episode-making, whereas the most healthy people use the method of grouping a few words as one chunk. Contrary to self-monitoring strategies, especially among the epilepsy patients, phonemic cues are unconsciously used to retrieve the words. The implication on the function of different brain areas activated between patients and healthy people will also be discussed.

5aPP18. Acoustic cues used by blind travelers. Helen Simon (Smith-Kettlewell Eye Research Inst, San Francisco, CA, helen@ski.org), Deborah Gilden (Smith-Kettlewell Eye Res Inst. San Francisco, CA), John Brabyn (Smith-Kettlewell Eye Res Inst, San Francisco, CA), Al Lotze (Smith-Kettlewell Eye Res Inst., San Francisco, CA), and Harry Levitt (Advanced Hearing Concepts, Bodega Bay, CA)

People with vision loss rely heavily on subtle environmental sound cues for safe and efficient travel ("Wayfinding"). Using our laboratory-developed instruments, the acoustic cues available to blind individuals, with and without hearing loss, during real-life pedestrian travel were recorded. Acoustic signals picked up by electret condenser microphones in the ear canals were fed to a wearable digital audio recorder. Head and body movements were monitored by accelerometers and gyroscopes mounted on the heads and torsos of subjects during typical travel situations such as walking along a corridor with an open doorway. A skilled wayfinder can detect the presence of an open doorway from the acoustic characteristics of the ambient sound field (the "acoustic signature"). The salient characteristics of the acoustic signature when passing an open door were found to be below 1500 Hz. These data confirm previous work regarding ambient room noise near a wall and an opening (Ashmead, 1999). Unlike previous work, the current study also measured the interaural characteristics of the acoustic signature. The results of this investigation will be used to develop the design requirements of a special-purpose hearing aid for people with both vision and hearing loss. (Funded by NIDRR and Smith-Kettlewell Eye Research Institute.)

5aPP19. Contribution of pitch-accent information to lexical decision in Japanese. Ikuyo Masuda-Katsuse (Kinki University, katsuse@fuk.kindai.ac.jp)

Word intelligibility, latency in shadowing, and brain activations when listening to spoken Japanese with incorrect pitch accents compared to normal, were investigated. Japanese is one of pitch-accent languages. In a pitch-accent language, "where" the pitch falls is important, whereas in Mandarin Chinese "how" pitch rises and falls is important. The word intelligibility scores were examined based on the adequacy rating of the accent types. Results reveal that under noisy conditions, the higher the adequacy of the accent type was rated, the higher the word intelligibility was. Under the clean condition, a significant negative correlation was shown between latency in shadowing and adequacy rating of accent types. In the event-related fMRI experiment, brain activities were investigated when listening to words with incorrect pitch accents (INCORRECT condition) and words with normal pitch accents (NORMAL condition). The contrast between INCORRECT and NORMAL revealed the significant activation in the bilateral inferior frontal, the bilateral precentral, and the left supplementary motor area, and right superior temporal gyrus. The result support the view that speech perception is a sensory-motor process, suggesting when a pitch accent of the stimulus was mismatch to the template, silent rehearsal might be repeated to make a lexical decision.

5aPP20. A cross-language study of breathing rhythm related to different linguistic structures in Chinese and Korean. Hanna Oh (Dept. of Chinese Language and Literature, Peking University, Beijing 100871, China, hanna.pku@gmail.com)

This paper investigates correlations between respiration patterns and different linguistic structures in Chinese and Korean.The study includes comparative analysis between the Korean and Chinese scripts of poetry and news with same meaning and different languages, and the subjects included 4 native speakers (2 males, 2 females) for each language. It can be made certain by measuring physiological and acoustic parameters by means of two respiratory sensors for chest and abdomen and electroglottalgraphy. Preliminary results show that due to the poetry has fixed frame, therefore regardless of language and sex, the respiratory rhythm of the poetry is quite similar in two languages. In contrast, for news scripts, the data show that breathing groups of the Chinese quite correspond to intonation units. In contrast,the Korean is more flexable in planning breathing rhythm,generally breathing resets appears not only in intonation phrase but in accentual phrase,and chest respiration is mostly used for the accentual phrase. It suggests that different linguistic structures operate powerfully upon preplanning and respiratory patterns. Furthermore, the results show the correlation of voice quality variation and respiratory rhythm in different languages.

5aPP21. Boundary tones in production and comprehension of in prodrop language: the case of Korean. Hyunah Ahn (University of Hawaii at Manoa, Department of Second Language Studies, 1890 East-West RD, Honolulu, HI 96822, hyunah1112@gmail.com)

This paper investigates if boundary tones have either a gradient or a categorical indication of structural phrasing. The role of prosody in syntactic disambiguation has been well documented and a recent study showed Korean speakers use the presence and absence of boundary tones to disambiguate a null-argument sentence borne out of double nominative construction in Korean (Ahn, 2011). Two experiments show that boundary tones play an important role both in comprehension and in production in spoken Korean. In Experiment 1, speakers read a paragraph including contexts and the critical sentence given in a latin-square design. The critical sentences in this production experiment include both null-argument and non-null-argument sentences to see if the use of boundary tones is for addressees or for the speaker himself. In Experiment 2, participants judge the naturalness of null-argument sentences given in contexts and answer comprehension questions probing if they got the correct reading of the two ambiguous interpretations. The results show a clear interaction effect of prosody and structure on the accuracy to the comprehension questions. A gradient-categorical dichotomy of prosody in sentence processing will be discussed and suggestions for a further study will be made to investigate the relationship of boundary tones and comprehension.

5aPP22. Estimating listeners' internal noise from intensity and spectral-shape discrimination tasks. Huanping Dai (University of Arizona, Tucson, AZ 85721, hdai@email.arizona.edu), Feng-Yi Chuang, and Andrew Lotto

Within the framework of signal-detection theory, listeners' ability to process stimulus information is partly limited by their internal noise. Knowing the internal noise of listeners will allow the researcher to predict their performance in various detection or discrimination tasks. For the purpose of estimating internal noise, in this study we measure the ability of listeners to discriminate the intensities of pure tones (Exp. 1) and two-tone complexes (Exp. 2), and to discriminate the spectral shapes of two-tone complexes (Exp. 3). The results from five listeners were fitted with an internal-noise model. The analysis showed that a major portion of the internal noise is correlated across frequency channels. This outcome is consistent with the conclusions of previous studies that estimated internal noise using different methods. Under the dominance of channel-correlated internal noise over channel-specific internal noise, listeners' ability to integrate information over frequency should be reduced, and their performance in detection and discrimination should be less affected by uncertainty for the signal parameters.

5aPP23. High-frequency complex pitch: a search for temporal cues and for a role of spectral indices. Sébastien Santurette, Torsten Dau (Centre for Applied Hearing Research, Technical University of Denmark, DTU Bygning 352, Oersteds Plads, 2800 Kongens Lyngby, Denmark, ses@elektro.dtu.dk), and Andrew J. Oxenham (Department of Psychology, University of Minnesota, 75 East River Parkway, Minneapolis, MN 55455)

Harmonics in a complex tone are typically considered unresolved when they interact with neighboring harmonics in the cochlea and cannot be heard out separately. Recent studies have suggested that the low pitch evoked by unresolved high-frequency harmonics may be coded via temporal fine-structure cues. However, these conclusions rely on the assumptions that combination tones were properly masked and that the ability of listeners to hear out individual partials provides an adequate measure of resolvability. Those assumptions were tested by measuring the audibility of combination tones

and their effects on pitch matches, the effects of relative component phases and of dichotic presentation, and listeners' ability to hear out individual partials. The results confirmed that combination tones affected pitch, but pitch remained salient when they were masked. The lack of dependence of pitch on relative component phases or dichotic presentation provided no evidence in favor of temporal cues. Moreover, similar trends were observed between pitch salience and the listeners' ability to hear out individual partials. The results are consistent both with the use of place information and with a temporal code based on the combination of information across auditory channels. [Supported by the Danish Research Foundation and by NIH grant R01DC05216.]

5aPP24. Envelope pitch at different stimulation sites of cochlear implant. Qinglin Meng, Meng Yuan, Hongyu Mou, and Haihong Feng (Shanghai Acoustics Laboratory, Chinese Academy of Sciences, No. 456, Xiaomuqiao Road, Shanghai, China, mengqinglin08@gmail.com)

Pitch perception researches showed that temporal modulation plays an important role in pitch perception from unresolved harmonics. Fundamental frequency (F0) enhancement through temporal modulation of envelope was utilized in some cochlear implant (CI) signal processing strategies [e.g., Laneau, et.al., 2006] to improve the pitch perception of CI. In this study, Temporal Modulation Transfer Function (TMTF) and the relationship between Amplitude Modulation Rate (AMR) and pitch perception at different stimulation sites (apical, middle, and basal) were investigated. The stimuli are sinusoidal-amplitude–modulation pulse trains. Two psychophysical experiments were carried out: Two-Alternative-Forced Choice on modulated/un-modulated stimuli and AMR-Pitch Ranking. Experimental results showed that some of the CI subjects exhibited distinct performance at different stimulation sites, which implies that issues of stimulation sites and the subject's individual performance should be considered for F0 enhancement strategies on CI. More experiments are in progress and whole results will be shown on the conference.# This work is supported by National Natural Science Foundation of China (11104316), and Shanghai Natural Science Foundation (11ZR1446000).

5aPP25. Loudness of sounds with a subcritical bandwidth. Jan Hots (Department of Experimental Audiology, Otto-von-Guericke University, Leipziger Strasse 44, D-39120 Magdeburg, Germany, jan.hots@med.ovgu.de), Jan Rennies (Project Group Hearing, Speech and Audio Technology, Fraunhofer IDMT, Marie-Curie-Strasse 2, D-26129 Oldenburg, Germany), and Jesko Lars Verhey (Department of Experimental Audiology, Otto-von-Guericke University, Leipziger Strasse 44, D-39120 Magdeburg, Germany)

The predicted loudness for narrowband signals with bandwidths smaller than the bandwidth of the auditory filter, i.e., with a subcritical bandwidth, depends on the type of loudness model. While models analyzing the long-term spectrum (stationary loudness models) predict loudnesses for these subcritical signals which are independent of the bandwidth of the sound, models sensitive to the temporal structure of the signal (dynamic loudness models) predict higher loudnesses for narrowband noise signals than for tones due to the inherent level fluctuations of the noise. The present study measures the loudness of subcritical sounds for center frequencies of 750, 1500, and 3000 Hz. For all three center frequencies, the level of the narrowband noise had to be up to 5 dB higher than the level of the equally-loud tone at the center frequency. This experimental finding is at odds with both predictions, i.e., the loudness is neither the same for all signals nor is the level of the narrowband noise smaller than that of the equally-loud tone. The data indicate a special character of sounds with a distinct tonal character which is not accounted for in current loudness models.

5aPP26. Amplitude modulation detection by human listeners in reverberant sound fields: carrier bandwidth effects and binaural versus monaural comparison. Pavel Zahorik (Department of Psychological and Brain Sciences, University of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu), Duck O. Kim, Shigeyuki Kuwada (Department of Neuroscience, Univ. of Connecticut Health Center, Farmington, CT 06030-3405), Paul W. Anderson, Eugene Brandewie, Regina Collecchia, and Nirmal Kumar Srinivasan (Department of Psychological and Brain Sciences, University of Louisville, Louisville, KY 40292)

Previous work [Zahorik *et al*., POMA, **12**, 050005 (2011)] has reported that for a broadband noise carrier signal in a simulated reverberant sound field, human sensitivity to amplitude modulation (AM) is higher than would be predicted based on the broadband acoustical modulation transfer function (MTF) of the listening environment. Interpretation of this result was complicated by the fact that acoustical MTFs of rooms are often quite different for different carrier frequency regions, and listeners may have selectively responded to advantageous carrier frequency regions where the effective acoustic modulation loss due to the room was less than indicated by a broadband acoustic MTF analysis. Here, AM sensitivity testing and acoustic MTF analyses were expanded to include narrowband noise carriers (1-octave and 1/3-octave bands centered at 4 kHz), as well as monaural and binaural listening conditions. Narrowband results were found to be consistent with broadband results: In a reverberant sound field, human AM sensitivity is higher than indicated by the acoustical MTFs. The effect was greatest for modulation frequencies above 32 Hz and was present whether the stimulation was monaural or binaural. These results are suggestive of mechanisms that functionally enhance modulation in reverberant listening. [Work supported by the NIH/NIDCD.]

5aPP27. Behavioral and electrophysiological measures of stimulus envelope and fine structure contributions to the binaural masking level difference. Ann Clock Eddins, Makenzie Kline, and David A. Eddins (Dept of Communication Sciences & Disorders, University of South Florida, 4202 E Fowler Ave., PCD 1017, Tampa, FL 33620, aeddins@usf.edu)

The goal of this study is to establish electrophysiological methods for estimating the relative contribution of stimulus envelope and fine-structure cues across a range of clinical populations using a binaural masking level difference (BMLD) paradigm. Stimulus envelope cues were manipulated by the using narrowband noise maskers (50 Hz) with the inherent envelope fluctuations of Gaussian noise (GN) or the reduced envelope fluctuations of low-noise noise (LNN). Fine-structure cues were manipulated by choosing signal frequencies and masker center frequencies of 500 and 4000 Hz. The availability of fine structure cues to the auditory system differs markedly between these two frequencies. Electrophysiological measures consisted of far-field evoked potentials targeting the P1-N1-P2 cortical components. Stimulus maskers were presented continuously at a spectrum level of 60 dB SPL while signal level was varied in descending steps. Threshold was estimated as the lowest signal level that generated a reliable N1-P2 response. Validation of cortical indices was achieved via behavioral measures of the BMLD. Signal level was varied using an adaptive three interval, three-alternative forced choice procedure with a three-down, one-up adaptive tracking rule. Raw thresholds and derived BMLD values using both measurement methods will be reported for a baseline population of 10 young, normal-hearing listeners.

5aPP28. Examining enhancement conditions with an auditory nerve model. Erica Hegland and Elizabeth Strickland (Purdue University, SLHS Dept, 500 Oval Drive, West Lafayette, IN 47907, ehegland@purdue.edu)

Psychophysical studies have shown that the ability to detect a signal in a masker may be improved by presenting a preceding sound (precursor) identical to the masker, an effect called overshoot. When the masker has no spectral notch around the signal frequency, overshoot has been predicted using a physiologically-realistic auditory nerve (AN) model in which gain is reduced by the precursor. This may simulate the medial olivocochlear reflex (MOCR). When there is a spectral notch in the masker at the signal frequency, the resulting improvement in signal detection is sometimes called enhancement. It has been suggested that enhancement may be due to adaptation of suppression, which could be related to gain reduction in the cochlea, or to adaptation of inhibition, which could occur more centrally. Physiological studies of the AN have shown either no adaptation of suppression [Palmer et al., J. Acoust. Soc. Am. 97, 1786-1799] or a reduction in suppression when the MOCR was elicited [Kawase et al., J. Neurophys. 70, 2533-2549]. The purpose of the present experiment is to use the AN model to investigate the relationship between gain reduction and suppression with stimuli used in previous studies of enhancement. [Research supported by NIH(NIDCD) R01 DC008327]

5aPP29. Modeling the effects of sensorineural hearing loss on temporal coding in the auditory nerve. David Axe (Weldon School of Biomedical Eng., Purdue Univ., 206 S. Martin Jischke Dr., West Lafayette, IN 47907, davidraxe@gmail.com), and Michael Heinz (Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907)

Recent psychoacoustical studies have suggested a temporal-fine-structure (TFS) deficit in listeners with sensorineural hearing loss (SNHL). Physiological studies generally have not found a reduction in the strength of TFS coding in auditory-nerve responses; however, a number of other effects on temporal coding have been found that may be related to these perceptual temporal processing deficits [e.g., Kale and Heinz, JARO (2010); Scheidt et al., Hear. Res. (2010)]. These physiological effects of SNHL include enhanced envelope (ENV) coding, changes in relative TFS/ENV coding that depend on stimulus bandwidth, broadened tuning, loss of tonotopicity for complex sounds, reduced latencies and traveling-wave delays, and altered temporal dynamics such as enhanced onset responses, faster adaptation, and slower recovery from stimulation. The current study evaluated which of these physiological effects were accounted for by an existing computational model of auditory-nerve responses [Zilany et al., JASA (2009)], and the extent to which these properties are predicted to depend on outer- vs. inner-hair-cell dysfunction. The degree to which existing models account for these effects of SNHL on temporal coding will help to elucidate the responsible mechanisms, and is important for future diagnostic and rehabilitative applications. [Work supported by NIH Grant No. R01DC009838.]

5aPP30. A modified channel model for the auditory peripheral system. YongHee Oh, Evelyn M. Hoglund, and Lawrence L. Feth (The Ohio State University / OH 43210, oh.172@osu.edu)

The correlated channel model (Zhang, 1995) was proposed as a modification of the (Durlach, et al., 1986) channel model by assuming that the internal noise in each peripheral channel is correlated. However, both models are limited because signals resolved into N channels by the peripheral processing unit are assumed to be statistically independent. Based on the results of several recent studies, the signals themselves may be correlated due to nonlinear properties of the basilar membrane, even before the addition of interchannel noise. In this study, the correlated channel model was extended using correlated and weighted signals, and the sensitivity index d' for a multi-tone detection task was derived. The performance of the modified model was characterized by parameter estimation, and compared with the results of a spectrotemporal integration experiment. [Research supported by a grant from the Office of Naval Research # N000140911017.]

5aPP31. Medial olivocochlear influence on stimulus-frequency otoacoustic emission input-output functions. James Dewey and Sumitrajit Dhar (Roxelyn & Richard Pepper Department of Communication Sciences and Disorders, Northwestern University, Frances Searle Building, 2240 Campus Drive, Evanston, IL 60208, jbdewey@u.northwestern.edu)

Modulation of cochlear mechanics by the medial olivocochlear efferent system is characterized by a reduction in active, outer hair cell-mediated amplification of basilar membrane motion. This increases cochlear thresholds and linearizes basilar membrane input-output functions for low-to-moderate stimulus levels. Significant efferent effects have also been observed for responses to higher stimulus levels, potentially reflecting changes in the mechanical properties of the cochlear partition. In humans, sound activated changes in stimulus-frequency otoacoustic emissions have been used as a tool for investigating the dynamics of the medial olivocochlear reflex. However, the degree to which the amplitude and phase of otoacoustic emissions are related to those of basilar membrane motion is not entirely clear. For the purposes of comparison with invasive physiological measurements in animals, stimulus-frequency otoacoustic emission input-output functions were obtained from human subjects in the presence and absence of contralateral acoustic stimulation. Medial olivocochlear effects were quantified in terms of the absolute change in emission level as well as the vector change, which incorporates changes in emission phase. The extent to which efferent modulation of emissions in humans reflects that observed in previous reports of basilar membrane motion will be discussed.

5aPP32. Comparison of pure tone thresholds obtained via automated audiometry and standard pure tone audiometry. David A. Eddins (Department of Communication Sciences & Disorders, University of South Florida, PCD 1017, 4202 East Fowler Avenue, Tampa, FL 33620, deddins@usf.edu), Joseph P. Walton (Department of Communication Sciences & Disorders, University of South Florida, PCD 1017, 4202 East Fowler Avenue, Tampa, FL 33620), Adam E. Dziorny (University of Rochester Medical Center, 601 Elmwood Ave., Rochester, NY 14642), and Robert D. Frisina (Department of Biomedical Engineering, University of South Florida, PCD 1017, 4202 East Fowler Avenue, Tampa, FL 33620)

It is likely that the role of automated audiometry will expand in both clinical and research settings in the next few years. A novel method for measuring pure tone thresholds using an automated threshold measurement method is reported here. The Automated Audiometry for the NIH Toolbox (AANT) test was developed for use in the NIH Toolbox multi-disciplinary evaluation battery which contains over 20 tests of sensory function and cognition. Development goals included low system cost, high accuracy, test administration time under 10 minutes, and automated calibration and measurement procedures suitable for use by evaluators without specialized training. Here we report the results of a validation study in which pure tone thresholds obtained using the AANT algorithm and hardware are compared to pure tone thresholds measured using the gold standard clinical method, standard audiometric hardware, and experienced examiners. Air-conduction thresholds will be reported for test frequencies of 500, 1000, 2000, 4000, 6000, and 8000 Hz using both methods. Additional measures for each subject include otoscopy, screening tympanometry, and the Hearing Handicap Inventory for Adults or Elderly. Data will be reported for 100 subjects between the ages of 4 and 85 years in age ranges.

5aPP33. Development of an evaluation system for cochlear implant. Zhenya Yang, Sheng Wang, Meng Yuan, and Haihong Feng (Shanghai Acoustics Laboratory, Chinese Academy of Sciences, No. 456, Shanghai Xu Hui Qu Xiao Mu Qiao Rd., Shanghai, China, yzy@mail.ioa.ac.cn)

The cochlear implant (CI) stimulation parameters such as pulse width, phase gap, pulse amplitude, and stimulation rate are the key and fundamental features in the objective evaluation of a CI device. In this study, an electronic evaluation system was developed to measure the above CI parameters for fully understanding the performance of a CI device. This system is multi-functional, powerful and flexible, which contains several modules, e.g. source signal generation, I/O synchronization, pulse sequence acquisition, data processing, and pulse/waveform display. A user-friendly graphic interface was developed including many useful functions and options to make different measurement and evaluation. The measurement results and original data can be easily exported from the evaluation system for further analysis. The accuracy and limitation of the system will also be discussed in this work. This system is suitable for CI parameter verification, speech processing research strategy evaluation, multi-electrode streaming, and CI quality examination. This work is supported by National Natural Science Foundation of China (11104316), and Shanghai Natural Science Foundation (11ZR1446000).

5aPP34. Sound source localization in the horizontal plane through the bilaterally applied bone-conducted ultrasonic hearing aids. Takuya Hotehama and Seiji Nakagawa (National Institute of Advanced Industrial Science and Technology (AIST), MOL105, 1-8-31 Midorigaoka, Ikeda, Osaka 563-8577, Japan, t-hotehama@aist.go.jp)

Bone-conducted ultrasound (BCU) can be perceived not only by normal hearing but also by the profoundly hearing impaired who cannot make use of a conventional hearing aid. BCU hearing aid (BCUHA), in which an ultrasonic carrier is amplitude-modulated (AM) by collected external sound and presented through a bone-conduction vibrator onto the mastoid portion, has been developed for the profoundly hearing impaired. To obtain useful information in order to realize accurate sound source localization through the BCUHAs. In this study, localization performance in the horizontal plane through bilaterally applied BCUHAs was investigated by psychological experiments. Results show that subjects hardly lateralized when the

BCUHAs were simply applied bilaterally with the double-sideband modulation method. On the other hand, the localization performances were improved when the inter-lateral intensity and time differences of presenting signals of BCUHAs were enhanced on the basis of those of the collected external sounds and when the "transposed modulation" was employed as the modulation method. Our results suggest the necessity of the inter-lateral cooperative system to modify the spatial parameters of the output signal for accurate sound localization through the BCUHAs.

FRIDAY AFTERNOON, 18 MAY 2012 HALL A, 2:00 P.M. TO 2:40 P.M.

Session 5pAAa

Architectural Acoustics and Psychological and Physiological Acoustics: Objective and Subjective Parameters of Spatial Impression in Performing Arts Spaces II

Michelle Vigeant, Cochair
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Jin Yong Jeon, Cochair
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Contributed Papers

2:00

5pAAa1. Subjective experiment on preferable reverberation of a musical practice room by professional players. Akihiro Nakajima (Graduate School, the University of Tokyo, Ce402, 4-6-1 Komaba, Meguro-ku, Tokyo, Japan, nakajima@iis.u-tokyo.ac.jp), Sakae Yokoyama, Sohei Tsujimura, Shinichi Sakamoto (I.I.S. the University of Tokyo, Ce401, 4-6-1 Komaba, Meguro-ku, Tokyo, Japan), Ami Tanaka, and Yoshihide Shiba (Nikken Sekkei LTD., 2-18-3 Iidabashi, Chiyoda-ku, Tokyo, Japan)

A musical practice room is very important space for players to improve their skills. However, no architectural design guideline of the musical practice room exists in Japan and the individual room is being designed based on architect's intuition and experiences. For a basic investigation aiming to establish the guideline, we are now investigating appropriate acoustic conditions of musical practice rooms for players by subjective experiments. In this study, we focused on reverberation time of the room and subjective experiments using a three dimensional sound field simulation were conducted for professional players including woodwind instrument players, string players and singers. In the experiment, a room having 6 conditions of reverberation time was virtually simulated in the sound field simulation system established in an anechoic chamber. As a result, a clear tendency of preferable reverberation time dependent on the difference of types of the instruments was found and a range of preferred reverberation time for a practice room was obtained.

2:20

5pAAa2. Measurement of acoustic characteristics in a traditional Korean Buddhist temple complex. Edwin Stephen Skorski III (Keimyung University, College of Architectural Studies, 1095 Dalgubeoldaero, Dalseo-Gu, Daegu, 704-701, Korea, sskorski@yahoo.com)

Traditional Korean Buddhist Temple complexes, typically found in rural, mountainous settings, hold great cultural, historical, and architectural significance in the Republic of Korea. These buildings, constructed primarily of wood, with little or no mechanical fasteners, create unique acoustic environments used for chanting, spoken word, music, and quiet meditation. Within the temple complex there are multiple halls and shrines of varying size and geometry that are utilized for worship. This presentation documents the room acoustic characteristics of a variety of worship spaces within a prototypical traditional Korean Buddhist Temple complex. These spaces include the main Buddha Hall as well as surrounding halls dedicated to additional Buddhas and Bodhisattvas. Using a laptop based measurement system; field measurements were conducted to document reverberation time (RT), clarity (C-50), speech transmission index (STI), and background noise levels. Data gathered is discussed and analyzed in regards to the primary use of each space. This research was conducted utilizing BISA Research Grant funds provided by Keimyung University.

Session 5pAAb

Architectural Acoustics and Psychological and Physiological Acoustics: Psychoacoustics in Rooms III (Lecture/Poster Session)

Philip Robinson, Cochair
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Bernhard Seeber, Cochair
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Invited Papers

2:40

5pAAb1. Effects of interaural level differences on the externalization of sound. Jasmina Catic, Sebastien Santurette, Torsten Dau (Centre for Applied Hearing Research, Technical University of Denmark, Lyngby, Denmark, jac@elektro.dtu.dk), and Jörg Buchholz (National Acoustic Laboratories, Chatswood, Australia)

Distant sound sources in our environment are perceived as externalized and are thus properly localized in both direction and distance. This is due to the acoustic filtering by the head, torso, and external ears, which provides frequency-dependent shaping of binaural cues such as interaural level differences (ILDs) and interaural time differences (ITDs). In rooms, the sound reaching the two ears is further modified by reverberant energy, which leads to increased fluctuations in short-term ILDs and ITDs. In the present study, the effect of ILD fluctuations on the externalization of sound was investigated. A psychoacoustic experiment was performed in a standard IEC 268-13 listening room by normal-hearing listeners. Individual binaural room impulse responses were used to simulate a distant speech source delivered via headphones. The speech signal was then processed such that the naturally occurring fluctuations in the ILDs were compressed, while the ITDs were preserved. This manipulation reduced the perceived degree of externalization mainly for broadband and highpass filtered speech. In the case of lowpass filtered speech, the compression of ILD fluctuations did not affect externalization. Overall, for sounds that contain frequencies above about 1 kHz the ILD fluctuations were found to be an essential cue for externalization.

3:00

5pAAb2. Auditory localization in realistic environments by normal-hearing and hearing-impaired listeners. Jorg M. Buchholz, Virginia Best, and Gitte Keidser (National Acoustic Laboratories, Australian Hearing, Chatswood NSW 2067, Australia, jorg.buchholz@nal.gov.au)

The ability to correctly localize sounds is important for general awareness of the auditory scene and communication in adverse acoustic conditions. However, most localization studies are performed in rather simple and artificial conditions. In particular, very few studies have considered localization in reverberant environments or in the presence of complex interferers, and no studies have systematically investigated the effect of distance. In the present study, localization performance was therefore measured as a function of source-receiver distance using a virtual auditory environment. With increasing source-receiver distance the direct-to-reverberation energy ratio decreases and the auditory system increasingly relies on mechanisms related to the precedence effect. Both aspects may be particularly problematic for hearing-impaired listeners. The acoustics of a cafeteria was simulated with the room acoustic software ODEON for a large number of source-receiver locations. Signals were generated for a 3D array of 41 loudspeakers using the loudspeaker-based room auralization (LoRA) toolbox. Localization performance was measured in normal-hearing and hearing-impaired listeners with a bilateral hearing loss in the simulated cafeteria with and without a multi-talker speech background. The experimental results were correlated with a number of acoustic measures derived from dummy head recordings of the different acoustic conditions.

3:20

5pAAb3. Explorations by the visually impaired of real and virtual rooms. Brian Katz (LIMSI-CNRS, BP 133, Université Paris Sud, 91403 Orsay, France, brian.katz@limsi.fr), and Lorenzo Picinali (Faculty of Technology, De Montfort University, The Gateway Leicester, LE1 9BH, UK)

Virtual acoustic simulations of two interior environments were presented to visually impaired individuals. Interpretations of the acoustic information, through block map reconstructions, were compared to reconstructions following in-situ exploration as well as playback of binaural and Ambisonic walkthrough recordings. Simulations used off-line HOA RIR synthesis and a hybrid rendering combining pre-convolved signals and real-time convolutions for sounds related to user displacement and self-generated noise. Results showed that listening to passive binaural playback or Ambisonic playback, which also included interactive head-movements, provided less usable information than a virtual simulation with respect to the acquisition of spatial information of an interior architectural environment. The presence of both dynamic cues relative to displacement and controlled events such as finger snaps, as included in the virtual condition, were deemed highly valuable by the participants. Virtual acoustic simulations provided acoustic information that allowed for highly

correlated detailed map reconstructions relative to a real exploration condition. Some differences were found between the two experimental corridors, with the more complex environment offering better results than the corridors with more diffuse noise sources. This study was supported in part by a grant from the European Union (STREP Wayfinding, No. 12959).

Contributed Paper

The following abstract will be presented in poster format. The poster will be on display and the author will be at the poster from 3:40 p.m. to 4:00 p.m.

5pAAb4. A modulation-transfer-function-based method for restoring sub-band power envelope from noisy reverberant speech. Shota Morita, Masashi Unoki (School of Information Science, JAIST, 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan, s-morita@jaist.ac.jp), Xugang Lu (National Institute of Information and Communications Technology, 3-5 Hikaridai, Seika-cho, Soraku-gun, Kyoto 619-0289, Japan), Yang Liu, Masato Akagi (School of Information Science, JAIST, 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan), and Ruediger Hoffmann (Laboratory of Acoustics and Speech Communication, Dresden University of Technology, Helmholtzstrasse 10, Dresden 01069, Germany)

The concept of the modulation transfer function (MTF) can be successfully applied to evaluating the quality of speech transmission in room acoustics (noisy reverberant environments) as functions of reverberation (reverberation time) and additive noise (signal to noise ratio) (Houtgast and Steeneken, J. Acoust. Soc. Am., 77, 1069-1077, 1985). This paper proposes a method of restoring the power envelope from noisy reverberant speech based on the MTF concept. The proposed method does not need the impulse response and noise conditions in room acoustics to be measured to enhance speech. The proposed approach suppresses the effects of reverberation and noise on the power envelopes by restoring the smeared MTF. We carried out massive simulations of noise-suppression and dereverberation on noisy reverberant speech to objectively evaluate the proposed method. The results revealed that the proposed method could simultaneously work well with both the suppression of noise and dereverberation. We further tested the proposed method as a front-end processor for ASR systems in noisy reverberant environments, and compared it with other methods (MFCC, CMN, spectral subtraction, and RASTA filtering on a constant-bandwidth filterbank). The results demonstrated that the improvement in recognition with the proposed method was more effective than that in extremely noisy reverberant environments.

FRIDAY AFTERNOON, 18 MAY 2012 S424, 2:00 P.M. TO 4:40 P.M.

Session 5pAB

Animal Bioacoustics and Acoustical Oceanography: Acoustic Animal Tagging

David Mellinger, Cochair
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Invited Papers

2:00

5pAB1. New generation pinger using pseudo noise sequence signal. Toyoki Sasakura (Fusion Inc. 1-1-1-806, Daiba, Minato-ku, Tokyo 1350091, Japan, sasakura@fusion-jp.biz)

Ultrasonic biotelemetry system is one of the useful methods to observe fish migration and behavior. New generation pinger, which has small size, long life, long distance and high recognition ability was developed using pseudo noise sequence signals.. Size, lifetime, distance and collision avoidance are tradeoff relation each other. To solve this, the advanced technology using CDMA mobile phone communication was adopted. Pseudo noise sequence signal was applied for transmitting signal and correlation processing was used for receiving signals. The transmitting signal consisted of 31 bits pseudo noise sequence signal and the correlator of receiving signal was 992 steps using FPGA(Field Programmable Gate Array). As a result, the new pinger has the size of $\phi 10\times 35$mm long including depth sensor, sound transmission lasts 240 days when 30 seconds repetition using the small battery SR626SW 32mAh. The signal could be achieved over 1,000 meters distance in a field validation. A unique set of 32 ID codes of the pseudo noise sequences were found to minimize collision of identification. In addition, depth and temperature information could be transmitted. The new generation pinger contributes not only to the research work of biotelemetry but also to fisheries and aquaculture purposes in the world.

2:20

5pAB2. High resolution data from DTAGS on the response of humpback whales to noise from seismic air guns. Rebecca Dunlop (School of Veterinary Science, University of Queensland, Gatton, QLD 4343, Australia, r.dunlop@uq.edu.au)

The BRAHSS (Behavioural Response of Australian Humpback Whales to Seismic Surveys) series of experiments uses a multi-platform approach to determine the behavioural and acoustic response of humpbacks to noise from seismic air gun arrays. One of the data collection platforms utilises the DTAG, or acoustic digital recording tag. DTAGs are small suction cup tags which contain a hydrophone as well as x, y and z plane accelerometers, magnetometers and depth sensors. Data from the DTAGs allows fine scale movement data of the tagged whale (dive profile, pitch, roll and heading movements, fluking rates) to be viewed as a pseudotrack whilst simultaneously listening to sound field of the whale (air gun shots and vocal sounds from nearby whales). This paper presents some results of responses of the whales to air gun sounds recorded by DTAGs and comparison with visual observations at the time. Since DTAGs provide a continuous record of whale vocalizations and 3D movements, vocal and physical reactions are detectable immediately as the first air gun shot occurs, whereas visual observation are limited to the appearance of the whales at the surface.

2:40

5pAB3. On-board telemetry of biosonar sounds from free-flying bats. Shizuko Hiryu (Faculty of Life and Medical Sci., Doshisha Univ., 610-0321, shiryu@mail.doshisha.ac.jp), Naohiro Matsuta, Shigeki Mantani (Faculty of Engineering, Doshisha Univ., 610-0321), Emyo Fujioka, Hiroshi Riquimaroux, and Yoshiaki Watanabe (Faculty of Life and Medical Sci., Doshisha Univ., 610-0321)

Analysis of the bat's reactions to relevant target echoes enables us to directly assess biosonar performance. Here, we recorded the sonar broadcast and its echoes the bat received during flight by using an on-board telemetry microphone (Telemike) mounted on the bat's back. Telemike recordings confirmed that flying bats adjust the amplitude and frequency of their sonar broadcasts to compensate for increases in echo amplitude and for Doppler-shifts. For insect capturing, the bat exhibited Doppler-shift compensation for echoes from the static target ahead, but not for echoes from the target moth even though the flying bat attended to the moth for capture. Positive and negative Doppler shifts (acoustic glints) caused by insect fluttering were observed in the constant-frequency component of observed echoes, which synchronized with wingbeat cycle of the moth. Combined frequency and amplitude compensation for the static target may be advantageous for detection of acoustic glints of target prey. We also constructed multiple-microphone arrays for tagging wild aerial-feeding insectivorous bats. Not only the location of the bat, but also direction and directivity of the bat's broadcast can be measured. This will allow us to investigate 3-D search algorithm of multiple targets by the bat. [supported by JSPS and ONR]

3:00

5pAB4. A review on bio-sonar behaviour research of Yangtze finless porpoise using animal bone acoustic data loggers. Ding Wang (Key Laboratory of Aquatic Biodiversity and Conservation of the Chinese Academy of Sciences; Institute of Hydrobiology of the Chinese Academy of Sciences, Wuhan 430072, P.R. China, wangd@ihb.ac.cn), Tomonari Akamatsu (National Research Institute of Fisheries Engineering, Fisheries Research Agency, Kamisu, Hasaki, Kashima, Ibaraki 314-0408, Japan), Kexiong Wang, and Songhai Li (Key Laboratory of Aquatic Biodiversity and Conservation of the Chinese Academy of Sciences; Institute of Hydrobiology of the Chinese Academy of Sciences, Wuhan 430072, P.R. China)

In recent 10 years, a miniature stereo acoustic data logger (A-tag, W20-AS, Little Leonardo, Japan) has been used to observe the bio-sonar behaviour of free-ranging Yangtze finless porpoise (Neophocaena phocaenoides asiaeorientalis). The A-tag is small enough to be attached on the one of the smallest odontocetes by using suction cups. The A-tag with two external hydrophones can record the sonar pulse intensity, inter-click-intervals, and sound source direction measured by time arrival difference of sounds at the two hydrophones. Major outcomes by A-tag were the in situ off-axis sonar beam pattern, attention and approach phase of biosonar behaviour, scanning sonar of rolling animals, and social contacts among free-ranging Yangtze finless porpoises. In addition, frequent ultrasonic sound production was confirmed that seemed to be a reason of high detection performance of this species using passive acoustic monitoring of finless porpoises. This biosonar monitoring method using miniature data logger system developed in a freshwater system in China provided new insights of underwater behaviour of other small odontocetes producing high-frequency echolocation clicks, such as harbour porpoises and white-beaked dolphins.

3:20

5pAB5. Sperm whale coda communication studied with multiple acoustic tags. Peter Teglberg Madsen (Aarhus University, Build 1131, Aarhus, Denmark, peter.madsen@biology.au.dk)

Female sperm whales spend their entire lives in matrilineal groups with long term social bonds. However, contrary to other toothed whale species with a complex, long term social structure, sperm whales seemingly only use stereotyped patterns of clicks, named coda, to radiate acoustic clan identity with little or no signature encoding. Due to their deep diving behavior in offshore water, it is difficult to study the acoustic behavior and source properties of sperm whale communication signals. To alleviate that, multiple acoustic Dtags was deployed on sperm whales to study the behavioral context and source parameters of coda production. It is shown that more than 50% of all codas are produced during deep dives where the active space can cover the entire foot print of the social group despite low apparent source levels of around 180 dB re 1uPa. Intra click information in the form of spectral and IPI information is heavily distorted as a function of aspect with little room for individual encoding. The unequivocal assignment of codas to tagged individuals have demonstrated a large potential for information transfer in the inter click intervals of codas suggesting a much more complex acoustic communication in sperm whales than previously proposed.

3:40

5pAB6. Acoustic tagging for counting feeding events of captive Amazonian manatees. Mumi Kikuchi (The Laboratory of Fisheries Biology, The University of Tokyo, 1-1-1, Yayoi, Bunkyo, Tokyo 164-8639, Japan, mumi@cocoa.plala.or.jp), Tomonari Akamatsu (National Research Institute of Fisheries Engineering, Fisheries Research Agency, 7620-7 Hasaki, Kamisu, Ibaraki 314-0408, Japan), Diogo A. de Souza, Fernando C. W. Rosas, and Vera M. F. da Silva (Aquatic Mammals Laboratory (LMA), National Institute of Amazonian Research (INPA), Aleixo, CEP 69060-001, Manaus, Brazil)

The Amazonian manatee is one of four extant species in the mammalian order Sirenia. They are restricted to the freshwater rivers, lakes and floodplains of the Amazonian river basin where they eat floating and emergent aquatic plants. Visual observation of wild manatees is nearly impossible because of the turbid water and tiny exposure of nose at gentle respiration movement, which consequently precludes study of their underwater behaviour. In this study, we applied animal-worn sound recorder (AUSOMS-mini, System Intec Co., Tokyo, Japan) to two captive Amazonian manatees at INPA, Brazil, in order to record their mastication sounds. Five species of aquatic plants were offered to the manatees separately. Mastication-sounds were extracted by custom made software developed on Matlab for off-line analysis. The mastication intervals and temporal sound structure depended on species of plants. In addition, individual difference of mastication intervals observed in the two manatees was probably due to different body sizes. Overnight recording showed that both manatees kept feeding for more than 0.5 hours followed by 1.6 $\pm$ 1.3 hours interruption of feeding. Other sounds such as insect calls, rainfall, and vocalizations of other manatees could be useful to understand habitat selection, weather and presence of conspecifics.
Acknowledgment Research and Development Program for New Bio-industry Initiatives

4:00

5pAB7. Using bioacoustic and satellite tags to study sperm whale depredation behavior, and to test new acoustic tracking methods. Delphine Mathias (Marine Physical Laboratory, Scripps Institution of Oceanography, San Diego, CA 92093-0238, delphine.mathias@gmail.com), Aaron Thode (Marine Physical Laboratory, Scripps Institution of Oceanography, San Diego, CA 92093-0238), Jan Straley (University of Alaska Southeast, Sitka AK 99835), John Calambokidis, Gregory Schorr (Cascadia Research Collective, Olympia, WA, 98501), and Russel Andrews (School of Fisheries and Ocean Sciences, University of Alaska Fairbanks, AK 99775)

Sperm whales have been depredating black cod from demersal longlines in the Gulf of Alaska for decades, but the behavior has now become pervasive enough that it is starting to affect government estimates of the sustainable catch, motivating further studies of this behavior. In 2007 and 2009, 11 bioacoustic "BProbe" tags were attached to adult sperm whales off Southeast Alaska under both natural and depredation foraging conditions. Measurements of the animal's dive profiles, acoustic behavior, and angular velocities allowed two categories of depredation to be identified. The dive depths and durations of "deep depredating" whales are similar to those of natural dives, but acoustic parameters show significantly significant differences. By contrast, "shallow depredating" whales conduct dives that are much shorter, shallower, and are four times more acoustically active than during natural foraging dives. In 2010, both a satellite and bioacoustic tag were deployed on a sperm whale near a two-element vertical tracking array. The location and acoustic data from the tags were used to confirm the long-range tracking ability of the fixed array system to ranges of at least 5 kms. [Work conducted under the SEASWAP program, supported by the 2011 North Pacific Research Board graduate student research award]

4:20

5pAB8. Adaptive prey tracking by echolocating porpoises studied with acoustic tags. Danuta Maria Wisniewska (Aarhus University, Department of Bioscience, Zoophysiology, C. F. Moellers Alle 3, bygn. 1131, DK-8000 Aarhus C, Denmark, danuta.wisniewska@biology.au.dk), Mark Johnson (Sea Mammal Research Unit, Scottish Oceans Institute, East Sands, University of St Andrews, St Andrews, Fife, KY16 8LB, UK), Kristian Beedholm, and Peter Teglberg Madsen (Aarhus University, Department of Bioscience, Zoophysiology, C. F. Moellers Alle 3, bygn. 1131, DK-8000 Aarhus C, Denmark)

Studying the behavior of aquatic echolocators and their prey has proved to be challenging. However, recent studies using Dtags on several toothed whale species in the wild have identified sequences of echoes interpreted as stemming from ensonified prey, along with accelerometer signatures possibly indicative of feeding events. The present study aimed at verifying those findings in a controlled environment, and elucidating what echograms may tell us about echolocation behavior during prey capture. We applied DTAG-3 tags, sampling sound at 500 kHz, to trained harbor porpoises during captures of dead and live fish of different species and sizes. To look at details of the feeding events, we used tag-synchronized high-speed underwater cameras. The prey targets gave rise to echoes that could be traced back to ranges of up to five meters and often tracked in the high repetition rate buzzes initiated at short ranges. These buzzes were typically closely tied to distinct acceleration signatures of attempted prey captures. Results show that porpoises can carefully track their prey with adjusted click intervals both during approach and buzz phases and can dynamically accommodate prey movements to provide high spatial resolution without range ambiguities.

Session 5pBA

Biomedical Acoustics and Physical Acoustics: Acoustic Microscopy Imaging Methods for Biomedical Applications II (Lecture/Poster Session)

Jonathan Mamou, Cochair
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Tadashi Yamaguchi, Cochair
yamaguchi@faculty.chiba-u.jp

Contributed Papers

2:00

5pBA1. Photoacoustic microscopy of tissue lesions induced by high-intensity focused-ultrasound. Amaury Prost, Olivier Simandoux, Jean-Marie Chassot, Emmanuel Fort, and Emmanuel Bossy (Institut Langevin, ESPCI ParisTech, CNRS UMR 7587, Université Paris 7, INSERM ERL U979, 10 rue Vauquelin, 75231 Paris Cedex 05, France, amaury.prost@espci.fr)

Photoacoustic imaging is a recent and rapidly developping technique that can provide images of optical absorption at depth in soft tissue. In this work, photoacoustic imaging is applied to detect tissue lesion induced by high-intensity focused-ultrasound (HIFU).Tissue changes during HIFU therapy include changes in optical tissue properties that may be detected using photoacoustics. However, although it has been demonstrated that HIFU lession could indeed be detected using photoacoustics (Chitnis et al., JBO 15(2), 2010, Cui et al., JBO 15(2), 2010), the exact origin of the observed contrast remains unclear. In this study, lesions induced in different types of biological tissue are imaged in vitro using a photoacoustic microscope. Lesions are induced in the tissue samples with therapeutic ultrasound in the MHz frequency range, and a photoacoustic microscope with imaging frequencies from the MHz to the tens of MHz range, are used to obtain acoustic-resolution photoacoustic images of lesions located at different depths in tissue. Several optical wavelengths, from 532 nm to the therapeutic window in the near-infrared are used to perform a spectroscopic study of the observed contrast.

2:20

5pBA2. Acoustic characteristics measurement of rat liver by multi-frequency ultrasound microscopy. Kenta Inoue (Faculty of Engineering, Chiba University, Chiba, Chiba 263-8522, Japan, z8t0804@students.chiba-u.jp), Yoshifumi Saijo (Graduate School of Biomedical Engineering, Tohoku University, Sendai, Miyagi 980-8579, Japan), Kazuto Kobayashi (Honda Electronics Co.,Ltd, Toyohashi, Aichi 441-3193, Japan), Jonathan Mamou (F. L. Lizzi Center for Biomedical Engineering, Riverside Research, New York, NY 10038), and Tadashi Yamaguchi (Research Center for Frontier Medical Engineering, Chiba University, Chiba, Chiba 263-8522, Japan)

Hepatitis is a growing health concern and early disease detection is critical. Ultrasound is ideally suited for real-time imaging of liver, but typical ultrasound images do not display quantitative tissue information because it is first necessary to understand the complex interaction between ultrasound and tissue and scattering models based on tissue properties must be devised. Towards this aim, speed-of-sound (SOS) and attenuation from three types of rat livers (normal, fatty, and fibrosis) were measured with a scanning acoustic microscope using transducers with center frequencies from 1-MHz to over 100-MHz. Results indicated that SOS and attenuation measured with each transducer showed the following trend. Variability in SOS and attenuation values of normal liver was much smaller than other livers at any frequencies. In the fatty liver, SOS was 20 m/s slower and the attenuation was 1.0 dB/cm/MHz larger than in the normal liver. In fibrosis, SOS and attenuation had values between those of normal and fatty liver. Additionally, the relation between the pathologic state of liver and SOS and attenuation was investigated. Correlation between the ultrasound wavelength and the distribution and size of fat or fiber deposits in the liver was investigated using the corresponding stained histology photomicrograph.

2:40

5pBA3. Magnetoacoustic tomography with magnetic induction: a novel approach for electrical impedance imaging. Qingyu Ma, Feng Zhang, Xiaodong Sun, and Xuanze Chen (Nanjing Normal University, maqingyu@njnu.edu.cn)

As a novel image approach, magnetoacoustic tomography with magnetic induction (MAT-MI) was proved to posses the potential merits of enhanced contrast and high spatial resolution. In this paper, based on the theoretical analysis of pulsed magnetic excitation, eddy current induction, acoustic vibration, acoustic transmission and acoustic waveform collection, the principle of MAT-MI was deduced in formulae for cylindrical measurement configuration. It was proved that the collected acoustic waveforms comprise the information of conductivity distribution of the object in various vibration amplitudes and opposite phases. A tomographic algorithm for diffraction acoustic source was employed to perform image reconstruction for a two-layer cylindrical phantom model and the conductivity configuration in terms of shape and size was reconstructed. The theoretical theory was also testified by the experimental results for a tissue-like sample phantom that the collected waveforms had good agreements with the simulation predictions and reconstructed image provided the conductivity distribution of the model in both the configuration and the dimension. The favorable results of the research suggested a potential feasibility of MAT-MI in electrical impedance imaging.

3:00

5pBA4. Three-dimensional quantification of freshly-excised human lymph node properties using high-frequency ultrasound. Jonathan Mamou (Riverside Research, New York, NY, jmamou@riversideresearch.org), Alain Coron (Laboratoire d'Imagerie Paramétrique, CNRS and UPMC Univ Paris 06, Paris, France), Emi Saegusa-Beecroft (Department of Surgery, Kuakini Medical Center and University of Hawaii, Honolulu, HI), Masaki Hata (Department of Surgery, Juntendo Medical Center, Tokyo, Japan), Michael L. Oelze (Bioacoustics Research Laboratory, University of Illinois, Urbana-Champaign, IL), Eugene Yanagihara (Department of Surgery, Kuakini Medical Center and University of Hawaii, Honolulu, HI), Tadashi Yamaguchi (Research Center for Frontier Medical Engineering, Chiba University, Chiba, Japan), Pascal Laugier (Laboratoire d'Imagerie Paramétrique, CNRS and UPMC Univ Paris 06, Paris, France), Junji Machi (Department of Surgery, Kuakini Medical Center and University of Hawaii, Honolulu, HI), and Ernest J. Feleppa (Riverside Research, New York, NY)

Human lymph nodes excised from cancer patients during lymphadenectomy can contain small clinically-important metastatic regions that can be missed because conventional histopathology methods do not allow nodes to

be examined over their entire volume. In this study, more than 250 lymph nodes were scanned in 3D using a 26-MHz ultrasound transducer before histology processing. Acquired radio-frequency data were processed using 3D regions-of-interest to yield thirteen quantitative ultrasound (QUS) estimates. The QUS estimates are related to tissue microstructure and are hypothesized to be different in normal nodal tissue and metastatic tissue. Four QUS estimates were obtained from backscattered spectra and the remaining nine were derived from envelope statistics. Following ultrasound scanning, serial-section histology was performed at 50-μm intervals to depict cancer foci in 3D. Classification based on QUS estimates was performed using linear-discriminant analyses in a step-wise approach, and areas under ROC curves (AUCs) were computed. The AUC for the linear combination of four QUS estimates was 0.87 for a dataset of 95 breast-cancer nodes. Similarly, using only two QUS estimates, an AUC of 0.95 was obtained for a dataset of 160 gastrointestinal-cancer nodes. These results suggest that QUS may provide an effective tool for detecting metastatic foci in lymph nodes.

3:20

5pBA5. Automatic segmentation of ultrasound contrast images of arteries for determining boundaries between arterial inner walls and blood flows: in vivo animal studies. Ming Qian, Lili Niu, Ruibo Song, Guan Qi, and Hairong Zheng (Shenzhen Institutes of Advanced Technology, Chinese Academy of Sciences, 1068 Xueyuan Avenue, Shenzhen University Town, Shenzhen, P.R. China, ming.qian@siat.ac.cn)

Ultrasound contrast images could be used together with ultrasonic particle image velocimetry technology to obtain multi-component field of arterial blood velocity and shear stress, which is helpful to understand atherogenic process and predict plaque rupture. Near-wall flow can hardly be measured accurately due to artery pulsation and tissue movement. Measuring accuracy for PIV analysis can be improved by detecting the boundaries between arteries and blood flow and determining region of interest in arterial ultrasound contrast images. A fully-automatic algorithm is proposed, which is named as time frame difference snake(TFDS). The ultrasound contrast images are subjected to median filter using a window frame difference method to de-speckle the lumen and highlight the arterial wall. The filtered image is subsequently subjected to an automatic initialization procedure to obtain the initial contour for the snake model. The TFDS model takes advantages of the standard deviation matrix of consecutive images, and is applied to obtain the final delineation of the boundary. Ultrasound contrast images of mouse carotid arteries are used for performance evaluation. The computer-generated segmentation is compared with the hand-outlined delineations. Statistical analysis shows that our algorithm's segmentations agree with that of trained operators for both the near-end and far-end carotid artery wall.

The following abstract will be presented in poster format. The poster will be on display and the author will be at the poster from 3:40 p.m. to 4:00 p.m.

5pBA6. Two-dimensional multi-layer of arterial elasticity measurement considering geometric transformations. Lili Niu, Ming Qian, Ruibo Song, Long Meng, Guan Qi, Xin Liu, and Hairong Zheng (Shenzhen Institutes of Advanced Technology, Chinese Academy of Sciences, 1068 Xueyuan Ave., SZ University Town, Shenzhen 518055, China, ll.niu@siat.ac.cn)

Knowledge of the arterial elasticity can provide an important reference for understanding arterial wall changes that may occur before and during the early stages of atherosclerosis. In practice, the movements of arterial wall during the cardiac cycle are complex, which undergo not only translation but also rotation and deformation subjected to pulse pressure, transmural pressure and shear force. Conventional correlation based methods for arterial wall movement evaluation consider only the translational component and limit their accuracy. This study proposes a texture matching method based on ultrasonic B-mode image for accurately measurement the movement of the arterial wall with geometric transformations, improving the accuracy of elasticity measurement. Feasibility of the method was studied in an in vitro silicone tube phantom and in vivo carotid arteries of mice. The calculated elastic modulus for the in vitro silicone tube agrees well with the results obtained from mechanical testing, deviating only 6.2%. The mean elastic modulus of the carotid arteries of mice is 122.4$\pm$29.2kPa. The results demonstrate that the texture matching method can measure elastic modulus of the arterial wall with geometric transformations accurately. It may be clinically useful for aiding early detection of atherosclerosis.

FRIDAY AFTERNOON, 18 MAY 2012 S221, 2:00 P.M. TO 4:40 P.M.

Session 5pEA

Engineering Acoustics, Underwater Acoustics, and Biomedical Acoustics: Acoustic Sensors and Actuators IV

Hairong Zheng, Chair
hr.zeng@sait.ac.cn

Contributed Papers

2:00

5pEA1. Synchronization intercomparsion method and sea trials of acoustic doppler current profiler (ADCP). Jianbo Wu, Kai Deng, Chao Gao, Jun Li, and Changhong Wang (Institute of Acoustics, Chinese Academy of Sciences, wujianbo@mail.ioa.ac.cn)

The advantage and disadvantage of intercomparsion methods adopted routinely for performance comparison of Acoustic Doppler Current Profiler (ADCP) were analyzed. A new ADCP intercomparsion method was introduced, by which ADCPs were synchronized to work alternately ping by ping in the same steady platform, and the requirements for achieving creditable comparing measurements were also discussed. Several intercomparsion tests of two ADCPs at mooring shelves and subsurface buoy were conducted in northern South China Sea. One ADCP is 150 kHz self-contained ADCP made by Institute of Acoustics, the other is RDI 150 kHz Quartermaster ADCP. A statistical method was used to process the measurement data obtained through the new intercomparsion method, and criterions were adopted to evaluate the performance of ADCPs. Comparing with other methods, the synchronization intercomparsion method is more objective and credible.

2:20

5pEA2. Humidity measurement using a ultrasonic probe based on sound attenuation. Takahiro Motegi, Koichi Mizutani, and Naoto Wakatsuki (1-1-1 Tennodai, Tsukuba, Ibaraki 305-8577, motegi@aclab.esys.tsukuba.ac.jp)

Measurement of spatial average humidity has been required for air conditioning management in facilities, to obtain a change tendency of environment. For example, spatial humidity management is important to prevent the damage of cultural assets. However, conventional sensors such as wet-dry hygrometers measure the local humidity at a measurement point and do not reflect the spatial humidity. In this paper, humidity measurement using a ultrasonic probe is proposed. A ultrasonic probe, which measures the sound attenuation coefficient in acoustical path, reflects the averaged humidity among the path and achieves non-contact measurement of the spatial average humidity between ultrasonic transducers. The performance of the proposed technique is evaluated in experiment, and it is found that if the probe knows the information of temperature, humidity can be successfully measured from sound attenuation coefficient.

2:40

5pEA3. Controllable fuel cell humidification by ultrasonic atomization. Chia-Chi Sung, Chang-Yuen Bai, Shao-Jui Chang, and Jau-Hong Chen (Department of Engineering Science and Ocean Engineering, National Taiwan University, ccsung@ntu.edu.tw)

Compared to the conventional bubble type used for PEMFC (proton exchange membrane fuel cell) humidification, ultrasonic atomization has the advantages of smaller size, it is easy to refill and change temperature, and, more importantly, humidity controllability. To improve the performance of the FC, this study developed a unique ultrasonic atomization system, including a gas heating pipe, ultrasonic driving circuit, ultrasonic atomizer, and humidity sensors. The reactant gases used are hydrogen and air. When the reactant gas obtains sufficient heat from the heating pipe and enters the ultrasonic atomizer, gas humidification is increased due to the ultrasonic micro water droplets blending into the gases. The humidity is controllable by adjusting either the heating temperature or driving voltage. The size of the micro water droplets can also be manipulated by adjusting the driving frequency. A higher driving frequency leads to smaller droplet mean diameter, which in turn increases the humidification efficiency. At least 90% relative humidity can be maintained under the condition that the flow rate of anode and cathode's reactant gas is restricted to between 1LPM and 25LPM. Under the same condition, the conventional bubble type can only achieve around 80%.

3:00

5pEA4. A free-flooded flextensional transducer with tube-beam coupling structure. Yaozong Pan, Xiping Mo, Yongping Liu, Yong Chai, and Yunqiang Zhang (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-Huan-Xi Road, Beijing 100190, China, lapan@mail.ioa.ac.cn)

The transducer consists of two annulus steel end plates, a tube stacked by piezoelectric rings as the driving element and a dual slotted concave aluminum shell coupled one end plate to the other. The shell has two concave parts and an annulus ring between them; and each part is equally slotted into several beams with gaps between each other. The tube and the beams form the tube-beam coupling structure. A sealing boot is covered on both inner surface of the stack and outer surface of the shell so that the water could freely flood through the ring stack. Some vibrating modes of the transducer including a cavity resonance are used to broaden the band. A prototype is fabricated and measured. The cavity resonant frequency is about 1250Hz with the transmitting voltage response of 130dB and the TVR fluctuates less than 6dB from 1800Hz to 5700Hz. This work was supported by the National Natural Science Foundation of China (No. 11074276).

3:20

5pEA5. A new type of wide band and wide beam longitudinal transducer. Yong Chai, Xiping Mo, Yongping Liu, Yaozong Pan, and Yunqiang Zhang (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-Huan-Xi Road, Beijing 100190, China, chaiyong@mail.ioa.ac.cn)

A new type of wide band and wide beam longitudinal transducer is introduced in this paper. The beam width could be expanded significantly by union of the special design about the shape of the head mass and a bucket tail mass which could both enhance radiating ability and reduce the longitudinal size. And the working bandwidth could also be expanded efficiently by using mode coupling between the flex mode of the head mass and the longitudinal mode. Using the finite element analysis, some major electro-acoustic parameters of a prototype transducer with a size of 74mm in maximum outside diameter and 76mm in total length are computed. It is shown that the projector has good performance. The resonance frequency is about 14kHz with a transmitting voltage response of 141dB. The Q factor for -3dB bandwidth is 1.3 and the beam width for -6dB bandwidth would be over 200 degrees.

3:40

5pEA6. A binaural hearing assistance system with front-back discrimination capability. Tsuyoshi Usagawa, Atsuya Saho, and Yoshifumi Chisaki (Kumamoto University, 2-39-1 Kurokami, Kumamoto 860-8555, Japan, tuie@cs.kumamoto-u.ac.jp)

Binaural hearing assistance systems have a well known ambiguity in front-back discrimination which is called as "front-back confusion" or "cone of confusion" in psychoacoustics. It is known that spectral cue of sound provides keys to solve this confusion in binaural listening condition and the peaks and notches of spectral components play main role to estimate the vertical angle in sagittal coordinate. In this paper, a binaural hearing assistance system with new front-back discrimination method is proposed. The discrimination method is implemented on an artificial neural network using interaural level and phase differences as input in selected frequency bins. The performance of the proposed system is examined for simulated conditions as well as experiments.

4:00

5pEA7. On direct optimization in mode space for robust supergain beamforming of circular array mounted on a finite cylinder. Yong Wang, Yixin Yang, and Yuanliang Ma (Institute of Acoustic Engineering, Northwestern Polytechnical University, yxyang@nwpu.edu.cn)

A direct optimization method in mode space for robust supergain beamforming of circular array mounted on a finite rigid cylinder is presented. According to the concept of eigen-decomposition, the beam pattern is decomposed into a series of eigen-beams weighted by modal coefficients. The modal cross spectral matrix in isotropic noise field is calculated from sound scattering theory based on boundary element method(BEM). This beamforming method gives the most suitable modal coefficient vector directly under the related constraint conditions via second-order cone programming, so that there is no need to transform indirectly from the weighting vector in sensor space which is essential in the modal robust supergain beamforming method proposed before. The results of simulation show that the direct modal beamforming method in this paper can not only improve robustness using the white noise gain constraint, but also change the mode orders to provide trade-off between array gain and robustness in low frequencies. Beam performance measures such as sidelobe level can be optimized in addition to array gain, and in this way more effective schemes for designing practical robust supergain beamformers can be developed.

4:20

5pEA8. Electromagnetic tomographic ultrasonic sensor. Pol Grasland-Mongrain, Jean-Martial Mari, Bruno Gilles, Jean-Yves Chapelon, and Cyril Lafon (LabTAU INSERM u1032, 151 Cours Albers Thomas, 69424 Lyon Cedex, pol.grasland-mongrain@inserm.fr)

A tomographic method based on the Lorentz force for the measurement of the pressure of an ultrasound transducer is presented. When a metal wire is vibrating under the influence of a pressure field created by an ultrasound

transducer while submitted to a magnetic field, the Lorentz force induces an electrical current. This current is considered proportional to the integral of pressure along the wire. By moving the wire perpendicular to the ultrasound axis, and rotating it around the same axis, a sinogram of the pressure field can be elaborated. Then an inverse Radon transform of the signal gives the pressure field spatial distribution. An experiment was conducted where a 1 MHz transducer generated an ultrasound wave with a focal point at 4 cm. A 100 μm in diameter shielded copper wire was placed perpendicular to the ultrasound propagation axis, and inside a 300 mT magnetic field created by a permanent magnet. The main advantages of the hydrophone created by the wire-magnet system are the large frequency bandwidth and the resistance to high pressure, parameters still under investigation. Possible disadvantages are the sensibility to electromagnetic noise and the possible distortion of the pressure field when using a too thick wire.

FRIDAY AFTERNOON, 18 MAY 2012 S421, 2:00 P.M. TO 4:00 P.M.

Session 5pNSa

Noise and ASA Committee on Standards: Environmental Noise and Regulations II

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Paul Schomer, Cochair
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Maurice Yeung, Cochair
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Contributed Papers

2:00

5pNSa1. Noise control for 24-hour rock drilling in urban area. Wilson HO, Isaac CHU (Wilson Acoustics Limited, Unit 616, Technology Park, 18 On Lai Street, Shatin, Hong Kong, China, who@wal.hk), Etienne Baranger (Dragages – Maeda – BSG Joint Venture, Mui Fong Street, Sai Ying Pun, Hong Kong), and Richard Kwan (MTR Corporation, MTR Headquarters Building, Telford Plaza, Kowloon Bay, Hong Kong)

Dill and blast is often utilized for construction of vertical shaft in hard rock geology for various tunneling projects in Hong Kong. Such excavation work is usually restricted during 1900-0700 hours due to noise concern, especially in urban area. For MTR West Island Line Contract No. 703 SHW to SYP tunnels, extensive noise mitigation measures have been employed at the Tunnel Boring Machine (TBM) launching shaft where located less than 10m from the nearest noise sensitive receiver (NSR). A well deigned shaft cover is the most prominent feature of noise mitigation measures and the insertion loss of the shaft cover was found to be minimum 49dB(A) on site. Groundborne noise impact from rock drilling was predicted within the statutory criteria and verified on site. Construction noise permit for 24-hour rock drilling by 2-boom jumbo drill was obtained.

2:20

5pNSa2. Controlling environment noise—a hong kong experience. Hom Au, Kin Wui Cheng, and Flora Kit Mei Lin (Environmental Protection Department, Hong Kong SAR, 26th Floor Southorn Centre, Hong Kong, frhau@epd.gov.hk)

Controlling environmental noise in a metropolitan city, like Hong Kong, has always been a challenge to both the Authority and the industries. This is partly attributable to the past neglect in environmental planning during the early 80s', and partly to the compacted urban settings in Hong Kong. Strategically, the Environmental Protection Department has attempted to pre-empt noise problems through early inventions in projects' planning stage, while in parallel effort was made to address existing noise problems through abatement programs and enforcement of the noise control legislation. This paper briefly outlines the overall framework being implemented in Hong Kong, both legislatively and administratively, to control different kinds of environmental noise sources. Some of the crucial thoughts in the framework and the experience gained through enforcing the relevant legislative provisions will be discussed.

2:40

5pNSa3. "Background" noise levels in mining towns—a case study. Chris McNeillie (SLR Consulting Australia Pty Ltd, Level 1, 514 Sturt Street, Townsville, 4810 QLD, Australia, cmcneillie@slrconsulting.com)

Legislation and planning policy often stipulate "background plus" criteria for assessing the noise impact of industry and validating noise related complaints. However, these criteria generally do not take into account the fact that industrial noise may be an established and accepted part of the noise environment of an area. This paper presents a case study of an assessment of a large-scale mining operation in central Queensland, Australia where mine noise contributed significantly to the background noise level in an adjacent Township. The paper explains why a true background plus approach was not appropriate and describes how, instead, 3D noise modelling was used to "benchmark" existing mine noise emissions for the purposes of controlling future increases.

3:00

5pNSa4. A deterrent to repeated noise offences. Louis P.L. Chan, C. L. Wong, K.W. Cheng, and T.W. So (Environmental Protection Department, Government of the Hong Kong Special Administrative Region, lplc@epd.gov.hk)

Since the Noise Control Ordinance (NCO) was implemented in 1989, various environmental noise sources and activities have been regulated through different control mechanisms. Nevertheless, the penalty in terms of fines imposed for breaches of the NCO seemed not sufficient to deter the recurrence of offences, in particular during the 90s, when Hong Kong was actively enhancing its infrastructure to accomodate the growth of its thriving economy. Although the maximum fines under the NCO had been doubled in 1994, repeated violations were still serious at that time, with one reason being the

continued disregard of those legal requirements by the corporate management. Despite the court had in some offence cases actually imposed the maximum fines, the corporate management simply treated the fines as part of project expenses as they were not held personally liable for the actions of their companies. That malpractice was also disturbing a level playing field in the industry, creating unfairness to other law-abiding persons. A legislative amendment was successfully introduced in 2002 to hold the top management of bodies corporate personally liable for repeated noise offences. This paper will describe the framework of this provision with reference to similar legislation in other countries. The corresponding deterrent effect over the past years would be analysed. Moreover, a code of practice providing practical guidance to prevent violation would also be discussed in the paper.

3:20

5pNSa5. Environmental noise from wind farms, prediction and measurements. Alice Elizabeth González (IMFIA - Facultad de Ingeniería, elizabet@fing.edu.uy)

Although mechanical noises of wind turbines have been eliminated with the development in their design, aerodynamic noise generation during operation is inherent to them in nature. Their major acoustic emission occurs in low frequencies. To predict noise levels in the environment associated with the operation of stationary sources-wind turbines included-, the methodology of ISO 9613-2 is usually recommended. It is not only a standard method of calculation but it is also the recommended methodology in the European Union. However, under some conditions (such as a stable atmosphere) the predictions of this method may significantly underestimate the expected environmental noise levels. Lately, other methodologies of prediction that take in account atmospheric stability have been developed. Nevertheless, predictions from different calculation methods do not always result on the same sound pressure levels. Different prediction methods are compared in this article. Their accuracy is then evaluated by checking environmental sound pressure levels with the predicted ones. Some issues that should not be neglected at the moment of working with noise emission sources of great height are remarked.

3:40

5pNSa6. Systematic approach for wind noise analysis of buildings for engineering applications in Hong Kong. C. W. Ng, K.W. Lo, and Sam P.S. Tsoi (Ove Arup & Partners Hong Kong Limited Level 5 Festival Walk, 80 Tat Chee Avenue, Kowloon Tong, Hong Kong, william.ng@arup.com)

Wind noise for tall buildings can be a concern for densely populated area. It is needed to develop a systematic approach of wind noise analysis for building design in engineering applications. This paper presents an innovative approach based on a three-tier analysis framework for wind noise assessment of external features in urban environment. The framework is divided into screening review of external locations susceptible of impact; broad assessment of areas susceptible to impact; and focused model assessment. In Tier 1, review of facade design is conducted to identify susceptible impacted areas. Screening criteria related to wind speed, shapes, dimension of facade elements are derived for evaluation of facade/architectural forms. Fundamental principles are applied to different turbulent flows to estimate relative significance based on Lighthill's Theory of aerodynamic noise. Statistical analysis of meteorological data is conducted from weather, wind tunnel/ AVA study data to form representative wind conditions. In Tier 2, broad assessment of categorised areas is conducted by simplified calculations using scaling laws/ regeneration prediction technique to estimate wind noise significance. In Tier 3, focused assessment of potential areas is conducted for specific evaluations. CFD is applied to flow fields for computation of aero-acoustic characteristics of vortex sources using Lighthill-Curle equation.

FRIDAY AFTERNOON, 18 MAY 2012 HALL C, 2:00 P.M. TO 4:00 P.M.

Session 5pNSb

Noise: Ship Noise and Vibration II (Lecture/Poster Session)

Lin He, Cochair
helin202@public.wh.hb.cn

Changgeng Shuai, Cochair
chgshuai@163.com

Contributed Papers

2:00

5pNSb1. Analysis on transmission of sound and vibration of underwater double cylindrical shells connected with noncontinuous slab ribs. Zhenguo Bai (Dept 5, Mailbox 116#, WuXi City, JiangSu Province 214082, China, xiaobzg@126.com)

To analyze the sound and vibration transmission characteristics of double finite cylindrical shells, modal superposition method is adopted to establish the analytical mode, in which interaction induced by water between shells and slab ribs connecting the shells is taken into account. The in-face and out-face motion of slab ribs bring forth the interaction of shells, and the former factor features in the vibration transmission of the double shells below the ring frequency of the shell. The circumferential discrete slab ribs is mathematically described as algebraic sum of some step functions, which is then expanded in sine function series to fit in with the control equation of double shells. The results show that: whether the slab ribs are discrete or continuous as an annular rib, the level of sound radiation of double shell doesn't alter a lot, instead of which, only the peak points move a bit. Keywords: Sound radiation, Step function, Double cylindrical shells

2:20

5pNSb2. An active vibration control method based on nonlinear reverse model and Fx-LMS algorithm. Li Yan, He Lin, and Shuai Changgeng (Institute of Noise & Vibration, Naval University of Engineering, Wuhan 430033, P.R. China; Science and Technology on Ship Vibration and Noise Laboratory, 430033, P.R. China, liyan312a@yeah.net)

The Filtered-X algorithm based on linear adaptive filters had been widely used in the active vibration control, but its effectiveness was limited by the nonlinearity of the active actuators. In this paper, the nonlinear reverse model of actuator was built using polynomial method and neural network algorithm, and an improved Fx-LMS algorithm based on reverse model linearization was put forward to address the nonlinearity of the actuators in active vibration isolation system. Simulation and experiment showed

that the improved Fx-LMS algorithm had better effect on periodic vibration control with nonlinearity compared with the traditional Fx-LMS feedforward control method.

2:40

5pNSb3. Low creep isolator design and research for Ship propulsion engine. Shu Lihong, He Lin, and Yang Xue (Institute of Noise & Vibration, Naval University of Engineering, Wuhan 430033, P.R. China; Science and Technology on Ship Vibration and Noise Laboratory, 430033, P.R. China, shulihong313@yeah.net)

A mechanical model based on Zener component was presented, which can perfectly describe and predict the creep characteristics of materials as parameters can be extracted with genetic algorithm not only from the step loading process, but also from the no-ideal loading process. This model can be applied to numerical calculation and prediction of vibration isolator's creep characteristics. The creep characteristic of the isolator has relationship with the creep characteristics of materials and the structure of the isolator. We can get the best creep characteristic and appropriate static and dynamic properties of the isolator by right design. A type of low creep vibration isolator made in polyurethane was designed for ship propulsion engine. Experimental results accorded with the numerical calculation result well.

3:00

5pNSb4. Study on the dynamic model of magnetorheological elastomer vibration isolator. Yang Xue, He Lin, and Shuai Changgeng (Institute of Noise & Vibration, Naval University of Engineering, Wuhan 430033, P.R. China; Science and Technology on Ship Vibration and Noise Laboratory, 430033, P.R. China, yangxue312@yeah.net)

Magnetorheological elastomers vibration isolator is a new controllable device, which shows a widely application in active or semi-active control. In fact, it is very important to constitute a precise dynamic model of the MR elastomer isolator, which gives help for design the control strategy and control method, and achieve good control effectiveness. At present the models is convenient to analyze and calculate the change of MR elastomer's properties, but it can not put up the saturation of the stiffness and the parameter of model is magnetic field strength which adding the complexity of engineering application. In this paper, a phenomenon model was constituted to describe the dynamic performance of the MR elastomer vibration isolator. Using the test results of the isolator, parameters of the model were achieved. By comparing the simulation results with the test results, we can find that they tally well with each other, which validates the dynamic model.

3:20

5pNSb5. A study of designing and computational method of single convoluted air spring. Zhao Yinglong, Lv Zhiqiang, and He Lin (Institute of Noise & Vibration, Naval University of Engineering, Wuhan 430033, P.R. China; Science and Technology on Ship Vibration and Noise Laboratory, 430033, P.R. China, zhaoyinglong409@yeah.net)

The method of design and calculation of single convoluted air spring is studied in this paper. And the method is used to design an air spring which is meet the demand. The load, static stiffness, dynamic stiffness, and normal frequency of the air spring are calculated and tested. The aramid cord filament framework layer of the air bag and the air inside are modeled by the Rebar Element and Cavity Element of the MSC.Marc software respectively. And the nonlinear Finite Element Analysis (FEA) model is obtained. The characteristics of the air spring are calculated by FEA. Lastly, the characteristics of the air spring are tested by the MTS test machine. The design and simulation results are accord with the experiment results. The theoretical analysis and experimental results indicate that the design method is simple and exact, the nonlinear FEA model simulate exactly the air spring, and the design and simulation method can be used to design the single convoluted air spring.

The following abstract will be presented in poster format. The poster will be on display and the author will be at the poster from 3:40 p.m. to 4:00 p.m.

3:40

5pNSb6. A study on the transfer matrix and dynamical characteristics of flexible pipe bend. Shuai Changgeng and Lv Zhiqiang (Institute of Noise & Vibration, Naval University of Engineering, Wuhan 430033, P.R. China; Science and Technology on Ship Vibration and Noise Laboratory, 430033, P.R. China, shuaichanggeng411@yeah.net)

Much of the noise originates from pressure and flow fluctuations generated by pump. These pressure fluctuations are transmitted along the hydraulic lines and cause unbalanced forces at pipe bends and changes of section. The flexible pipe bend, a recently developed flexible component of pipe, can usually reduce thermal expanding reaction and endure large and complex stress and strain arising from different movements and also due to complex geometry. The rubber hose wall can conduct oscillating longitudinal forces that may interact with fluid waves inside. Up to now, little attention has been paid to study its performances of vibration damping and noise attenuation in pipeline. The transfer matrix, an effective numerical method, is applied to analyze and evaluate the transfer performance of flexible pipe bend. However, because of its geometrical complexity, it is difficult to directly calculate the transfer matrix of the flexible pipe bend. In this paper, the flexible pipe bend is divided into several parts, each of which can be considered as a short straight pipe. Through the conversion between the global coordinate system and local coordinate system, a transfer matrix of the flexible pipe bend is derived and applied, which provide a practical method to analyze the frequency, modal and response of the flexible pipe bend.

Session 5pNSc

Noise and Architectural Acoustics: Noise Effects On Occupant Comfort and Performance in Buildings II

Lily Wang, Cochair
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C. M. Mak, Cochair
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Contributed Papers

1:40

5pNSc1. The acoustics approach to new towns and buildings planning.
James Wing Ho Wong (Allied Environmental Consultants Ltd. 19/F, Kwan Chart Tower, 6 Tonnochy Road, Wanchai, Hong Kong, gk@aechk.com)

Sound have been known to affect the physical, emotional, mental and spiritual states. Sound affects pulse rate, skin temperature, blood pressure, muscle tension, and brain wave activity. It helps release biochemicals, such as endorphins. It relaxes, excites, releases emotions, and helps to travel to altered states of consciousness. Sound has been applied in architectural design in ancient times and recent years, it helps in healing all those disease in which stress plays a role. This study develop advanced acoustic design concepts in use of sound healing in new towns and buildings planning. Harmonic building design and town planning is an exciting and emerging field that links the ancient sciences and arts with sound and vibration to heal the spirit, mind and body. The goal is to assist the individual and community at large in establishing healthier patterns for living in wholeness. It draws fundamentally on a unified view of the cosmos in which all things are inextricably bound and in relationship to one another whether it be an organ system, a family or an ecosystem. When our bodies experience an illness or disharmony, this will initially develop within the organizational energy matrices. The energy matrix of the body's cellular makeup is made up of a highly complex multitude of frequency interactions. When the body is subject to sound stimulation, it sets up a series of harmonic oscillations that influence a change in phase and frequency in the energy field or matrix on the cellular level. The result will be to reorganize the energetic matrix back to a more balanced state. This paper investigate how does sound and harmonic healing work. What is a typical resonance based building look like? What is the underlying concept about resonance healthy communities and how can Harmonic Town Planning design techniques be applied to create healthy communities.

2:00

5pNSc2. A study on acoustical protection effects of plenum windows.
Y.G. Tong (Faculty of Civil and Environmental Engineering, Universiti Tun Hussein Onn Malaysia, Johore, Malaysia, ygtong@uthm.edu.my), and S. K. Tang (Department of Building Services Engineering, The Hong Kong Polytechnic University, Hong Kong, China)

Urban noise is likely to continue as a major issue in densely populated cities. The use of natural ventilation for energy saving and sustainable reasons become difficult to implement for the buildings in noisy urban environments. A staggered inlet and outlet openings design of plenum window was investigated in the present study. Two-dimensional model simulation was used to examine the effect of acoustic benefits of this window system relative to the different sound incidence of angle by solving numerical Helmholtz equation. Fully opened window is used as reference case. The results show that the insertion loss of plenum window is affecting by its staggered design and orientation of the window. There is peak of insertion loss observed at lower frequency which becomes sharper when the opening size of the window becomes smaller. [Y.G. Tong sponsored by Ministry of Higher Education, Malaysia.]

2:20

5pNSc3. Innovative approach of acoustic design of high performance facade with auralization at Arup SoundLab.
C. W. Ng (Ove Arup & Partners Hong Kong Limited Level 5 Festival Walk, 80 Tat Chee Avenue, Kowloon Tong, Hong Kong, william.ng@arup.com), Henry C.K. Chan, and Sam P.S. Tsoi

This paper presents an innovative approach of acoustic design of high performance facade for buildings in the vicinity of an airport. The sound insulation performance of the facade system is established by an aircraft noise auralization of the aural environment in the built environment such as a Star hotel at Arup SoundLab for determining an acoustically suitable glazing system. Arup SoundLab uses a 3D loudspeaker reproduction system called Ambisonics to reproduce the sound phenomena in a dedicated room that allows people to listen to simulated sound in virtual spaces before actual construction. The audio samples can be generated to mimic the dynamic indoor acoustic environment for the takeoff noise profile of B747 aircraft. After listening to the audio samples for different glazing configurations and ambient conditions, subjective responses from a group of representative individuals to the audio sound track that simulates a possible future situation are collected for evaluation analysis. The engineering development of high performance acoustic facade is presented including laboratory testing to reach a confidence level for achieving the end results. An opportunity for identifying potential cost saving is also discussed as a result of this aircraft noise auralisation. Examples will be presented to illustrate the approach.

2:40

5pNSc4. Innovative arc screen development for adaptive traffic noise mitigation of housing development using full scale prototype testing.
Sam P. S. Tsoi, C. C. Lai, C. K. Lau (Ove Arup & Partners Hong Kong Limited Level 5 Festival Walk, 80 Tat Chee Avenue, Kowloon Tong, Hong Kong, sam.tsoi@arup.com), and Kenneth H. K. Wong (Housing Department, Hong Kong)

The Housing Authority is responsible for public housing development in Hong Kong. With increasing demand for sites of challenging environmental conditions, the authority has been continuously diverting efforts on innovative design to tackle site environmental issues. One of the frontline development is an innovative arc-screen structure, which was devised to become one of the new generations of adaptive traffic noise mitigation. Its inception was initiated from the need to reduce traffic noise impact of a major highway parallel to the public housing development in Sai Chuen Road. As scale model evaluation of traffic noise reduction is somewhat limited, the project commissioned a full scale testing of an arc-screen prototype built into a 3-storey full size stacking up housing flat model for evaluation. This is believed to be the first time of a full scale experimental evaluation of traffic noise mitigation for housing development. The arc-screen structure has achieved 5 dB(A) nominal noise attenuation performance. Results are encouraging for public housing development applications and will form new generations of adaptive architectural forms of arc-screen uses to suit different needs. The first application is already planned at the Sai Chuen Road development and in-situ performance evaluation will be assessed in second phase.

Session 5pNSd

Noise, ASA Committee on Standards, Psychological and Physiological Acoustics, and Animal Bioacoustics: Impulsive Noise Exposure Metrics: Development and Validation

Richard Mckinley, Cochair
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Invited Papers

3:00

5pNSd1. Comparative study of military impulse noise criteria currently in use and identification of research needs. Steve Goley (Mechanical Engineering, University of Cincinnati, Cincinnati, OH 45221-0072, steve.goley@gmail.com), and Jay Kim (Mechanical Engineering, University of Cincinnati, OH 45221-0072)

Impulse noise criteria currently being in use, such as MIL-STD 1474D, Pfander, Smoorenburg and LAeq8hr, employ respectively a different noise metric to estimate the exposure risk to military noises. In these criteria, the risk is assessed based on the characteristics of the noise such as the peak level or equivalent energy. Recently a new approach that utilizes a simulation program of the auditory system called AHAAH is advocated as an alternative, which considers characteristics of not only the noise but the auditory system response in assessing the risk. In this work, underlying assumptions and analytical structures of widely used impulse noise criteria and AHAAH are examined to understand their relative strengths and shortcomings. Then, performances of the criteria are compared by utilizing an existing animal data obtained by exposing chinchillas to various impulse noises. Linear correlations of the noise metric with the PTS at 0.5, 1,2,4 K-Hz inflicted in chinchillas are used as the basis of the comparison which indicates that LAeq8hr and AHAAH show better performance. Research needs to enable improvement of existing criteria and development of a new improved criterion are discussed. Acknowledgment: Travel for this invited presentation was supported by the Air Force Research Laboratory

3:20

5pNSd2. Changes in distortion product oto-acoustic emissions after exposure to continuous and impulsive noise. Miguel Angel Aranda de Toro (GN ReSound; Lautrupbjerg 7, DK-2750 Ballerup, Denmark, maadtoro@gnresound.dk), Rodrigo Ordoñez, and Dorte Hammershøi (Acoustics, Department of Electronic Systems, Aalborg University; Fredrik Bajers Vej 7-B5, DK-9220 Aalborg Ø, Denmark)

Temporary changes in the hearing of human subjects were monitored with distortion product otoacoustic emissions (DPOAEs) after control sound exposures in a laboratory. The objectives of the experiment were to investigate whether the +5~dB penalty for impulsiveness used in international standards and legislation correlates to a higher risk of hearing damage. Subjects were exposed to two types of binaural recordings consisting of a continuous broad-band noise-exposure normalized to $L_{EX,8h}$ = 80~dB and the interaction of the previous stimulus with a noise of impulsive character normalized to $L_{EX,8h}$ = 75 + 5~dB penalty = 80~dB. The results show that the effects on DPOAE levels from the two stimuli could be compared in terms of their total acoustic energy.

3:40

5pNSd3. Scientific basis and shortcomings of EU impulse noise standards. Karl Buck, Véronique Zimpfer, and Pascal Hamery (French-German Research Institute, 5 rue du Général Cassagnou, BP 70034, 68301 Saint-Louis, France, karl.buck@isl.eu)

In 2003 the Directive 2003/10/EC has been published by the European Union. It defines the maximum noise exposure levels for workers and the related necessities for hearing protection and for conservation programs. The limits for continuous noise are based on A weighted exposure levels as defined by ISO 1999:1990. For impulse noise the Directive limits the maximum peak pressure level to 140 dB including hearing protection. This single value limit which does not take into account any frequency or duration information is not adequate to evaluate the hazard of an impulse noise and lacks any scientific validation. Although it may not present many problems in industry, for the armed forces it has become a major problem, as it limits training and bans the use of some weapon. Therefore Germany has not implemented this Directive for its forces but uses the Damage Risc Critera (DRC) designed by Pfander. In other countries which have implemented the Directive "As-is", other criteria based on A-weighted energy (LAex,8h) proposed by Dancer or the AHAAH proposed by Price are taken into consideration for being used. The presentation will discuss scientific basis of the different DRCs for impulse noise in use or being candidate in Europe.

4:00

5pNSd4. A case for using A-weighted equivalent energy as a damage risk criterion for impulse noise exposure. William J. Murphy (National Institute for Occupational Safety and Health, Hearing Loss Prevention Team, 4676 Columbia Parkway, Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov), and Richard L. McKinley (Air Force Research Laboratory, 2610 Seventh Street, Wright-Patterson Air Force Base, OH 45433-7901)

Damage risk criteria (DRCs) for continuous noise rely upon epidemiologic analyses of populations of persons exposed over several years to noise in occupational environments. In 2006, the U.S. Army proposed to update the MIL-STD 1474D to use the Auditory Hazard Assessment Algorithm for Humans (AHAAH) and discontinue using the peak sound pressure level, envelope duration and number of impulses. The National Institute for Occupational Safety and Health has conducted two separate evaluations of the data used to justify the AHAAH methodology and found that the use of the A-weighted equivalent energy L_{Aeq8} was more suitable for the purposes of predicting the effects of temporary threshold shifts (TTS) both in humans and in chinchillas. The L_{Aeq8} method provided best fit for the TTS outcomes and demonstrated the greatest discrimination (ability to predict TTS) when compared to AHAAH, MIL-STD 1474D and two other proposed DRCs. Similarly, L_{Aeq8} was found to give the best-fit and greatest discrimination for the chinchilla impulse noise exposures. The L_{Aeq8} affords the best sensitivity and specificity for discrimination of potential hazards and has the greatest level of integration with present occupational exposure standards and prospective hearing protection labeling regulations.

4:20

5pNSd5. Using impulsive peak insertion loss of hearing protectors with impulsive damage risk criteria. Richard McKinley (Air Force Research Laboratory, 2610 Seventh St., AFRL/11HPW/RHCB, WPAFB, OH, 45433-7901, richard.mckinley@wpafb.af.mil), Hilary Gallagher (Air Force Research Laboratory, 2610 Seventh St., AFRL/11HPW/RHCB, WPAFB, OH 45433-7901), and William Murphy (National Institute for Occupational Safety and Health, 4676 Columbia Parkway, MS C-27, Cincinnati, OH 45226-1998)

Impulsive noise presents special challenges for hearing conservation. The scientific community continues to search for an impulsive noise exposure criterion which accurately assesses the hearing damage risk for both short and long duration impulses. Other factors, such as the use of hearing protectors, earmuffs and earplugs, also affect the hearing damage risk but the current criteria do not address methods for using the attenuation of hearing protectors in impulsive noise. The relatively new ANSIS12.42-2010 “Methods for the Measurement of Insertion Loss of Hearing Protection Devices in Continuous or Impulsive Noise using Microphone-in-Real-Ear or Acoustic Test Fixture Procedures” describes methods for measuring the peak insertion loss of a hearing protector in impulsive noise. This paper will describe a method to apply the peak insertion loss data to impulsive noise damage risk criteria for an estimate of allowable impulsive noise exposure when using hearing protection.

Contributed Paper

4:40

5pNSd6. A hybrid model for predicting the sound level of shooting noise from recoilless rifles. Byunghak Kong (School of Mechanical and Aerospace Engineering, Seoul National University 301-1214, 1 Gwanak-ro, Gwanak-gu, Seoul 151-742, Korea, bhgong03@snu.ac.kr), Kyuho Lee (School of Mechanical and Aerospace Engineering, Seoul National University 311-105, 1 Gwanak-ro, Gwanak-gu, Seoul 151-742, Korea), and Soogab Lee (Institute of Advanced Aerospace Technology and School of Mechanical and Aerospace Engineering, Seoul National University 311-105, 1 Gwanak-ro, Gwanak-gu, Seoul 151-742, Korea)

In the present study, acoustic signals from M40A1 recoilless rifle were measured according to recommendations of International Organization of Standardization (ISO) using B&K 2250 sound analyzers. Contrary to expectations from previous studies about shooting noise, the signal recorded at each point shows different waveform in comparison with the others. This means that an additional acoustic source was present, and there was no doubt that it was located at a tail nozzle of the weapon through temporal and geometric relations between the microphone arrays. Spectral analysis and frequency band filter were used to find out what kind of acoustic source there was, and it was concluded that acoustic properties of the additional source was much close to those of jet noise rather than blast noise. Based on these results, consequently, a hybrid model has been developed to predict the sound pressure level of shooting noise from recoilless rifles using predicting algorithms applicable to acoustic sources, blast noise and jet noise, relatively.

Session 5pPA

Physical Acoustics: Negative Radiation Forces Exerted by Acoustical and Optical Beams

Phil Marston, Cochair
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Jack Ng, Cochair
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Invited Papers

1:40

5pPA1. Single beam acoustic tweezer. K. Kirk Shung (Department of Biomedical Engineering, University of Souhtern California, Los Angeles, CA 90089, kkshung@usc.edu)

Single beam acoustic tweezer, a distant cousin of optical tweezer, has been recently experimentally validated. A prerequisite of acoustic tweezer as in optical tweezer is a sharply focused beam with a steep intensity variation within the dimension of a particle. As the frequency of an acoustic beam reaches 100 MHz or higher, the beam diameter may approach cellular level allowing acoustic tweezing or trapping of a cell. Recent experimental results have shown that it is possible to trap lipid spheres of 100 μm diameter at 30 MHz and 15 μm diameter leukemia cells at 200 MHz. These results along with the experimental arrangement and potential biomedical applications of acoustic tweezer will be discussed in detail in this paper.

2:00

5pPA2. Pulling particles backward using a forward propagating beam. Jack Ng (The Hong Kong University of Science and Technology, jack@ust.hk), Jun Chen, Zhifang Lin (Fudan University), and C. T. Chan (The Hong Kong University of Science and Technology)

Can the scattering force of a forward propagating beam pull a particle backward? A photon carries a momentum of $\hbar{*}k$, so one may expect light will push against any object standing in its path. However, light can indeed “attract” in some cases. For example, a focused light beam can attract particles, due to the gradient force. But it is probably more appropriate to say that the gradient force “grabs” rather than “pulls”, as the particle will remain stable in the trap after being drawn to the focus. Here, we discuss another possibility — a backward scattering force which is always opposite to the propagation direction of the beam so that the beam keeps on pulling an object towards the source without an equilibrium point. In the absence of intensity gradient, using a light beam to pull a particle backwards is counter intuitive. The underlining physics is the maximization of forward scattering via interference of the radiation multipoles. We show explicitly that the necessary condition to realize a pulling force is the simultaneous excitation of multipoles in the particle and if the projection of the total photon momentum along the propagation direction is small, attractive optical force is possible.

2:20

5pPA3. Negative acoustic radiation forces produced by Bessel beams: acoustic tractor beams and scattering. Philip L. Marston (Physics and Astronomy Dept. Washington State University, Pullman, WA 99164-2814, marston@wsu.edu), Likun Zhang, and David B. Thiessen (Physics and Astronomy Dept. Washington State University, Pullman, WA 99164-2814)

The theory for negative acoustic radiation forces on objects in various Bessel beams is reviewed [P. L. Marston, J. Acoust. Soc. Am. 120, 3518–3524 (2006); L. K. Zhang and P. L. Marston, Phys. Rev. E 84, 035601 (2011)] together with geometric insight provided by researchers in optics. When examining conditions favorable for negative radiation forces it can be helpful to consider the far field scattering for axisymmetric objects of interest because the axial radiation force can be directly related to the asymmetry in the scattering and the conic angle of the beam. The analysis also serves to clarify the way in which absorption by the object contributes to positive forces and to clarify the influence of helicity. Considering the symmetry properties of the scattering provides additional insight as does the evaluation of radiation forces and wave fields using the computational method of finite elements. In the non-absorbing case it can also be helpful to express the partial-wave amplitudes in terms of phase shifts since in acoustics there is only one phase shift for each partial-wave. [Marston and Thiessen were supported by ONR and Zhang was supported by NASA.]

2:40

5pPA4. Use optical beams to push and pull: an old paradigm with new features. Cheng-Wei Qiu (National University of Singapore, 4 Engineering Drive 3, Singapore 117576, chengwei.qiu@nus.edu.sg)

We present a fundamentally distinguished schematic for a novel micromanipulation which realizes stable trapping and continuous optical traction/pulling of molecules at one go, opening up its widely appealing potentials in physics and biomedical engineering. We theoretically investigate the origin of pulling force by modeling tractor beams and its explicit correlations on the laser types, particle's

parameters, and beam polarization modulation. The novel technology of tractor beam is developed and proposed via a systematic analysis from electromagnetic scattering theory, condition of pulling force, and experimental investigation by using binary-lens optics. Instead of treating the object as a "blackbox" and interfering multiple beams, our technique exploits a single beam with phase modulation, and more importantly we study the fundamentals of the energy exchange, force direction switching, and the far-field radiation flipping. The major criterion is to manipulate the beam-particle interference to maximize the transfer of momentum along the forward direction, so that the reaction force will be dragging the particle all the way towards light source continuously, i.e., the fantasy of tractor beam.

3:00

5pPA5. Attraction by sound—examples on the negative acoustic radiation forces of Bessel "tractor" beams on a sphere. Glauber Silva (Universidade Federal de Alagoas, Maceio AL, Brasil, glauber@pq.cnpq.br), and Farid Mitri (Los Alamos National Laboratory, Los Alamos, NM)

Expressions for the axial (i.e. acting along the axis of wave propagation) and transverse (lateral) acoustic radiation forces on a suspended object in an ideal fluid are derived for arbitrary shaped beams. These expressions are obtained from the integration of the radiation stress tensor in the far-field. Numerical examples on non-vortex and vortex Bessel beams on a spherical target are presented. It is demonstrated the existence of both "pushing" and "tractor" behaviors for which the axial force can be oriented, respectively, along the direction of wave propagation or opposite to it. This can be adjusted by choosing specific values of the beam cone angle and the sphere size factor ka (where k is the wavenumber and a is the sphere's radius). Furthermore, by changing the topological charge m of the beam and the position of the sphere, the transverse force may be oriented radially (outward or inward) for m=0 or may exhibit a vortex pattern for $m \geq 1$.

3:20

5pPA6. Acoustic trapping of particle by a periodically structured stiff plate. Feiyan Cai, Fei Li, Long Meng, and Hairong Zheng (1068 Xueyuan Avenue, Shenzhen University Town, fy.cai@siat.ac.cn)

Acoustic tweezers have found potential applications in both the physical and life sciences in the past two decades due to their abilities of contactless, controllable and noninvasive trapping and manipulating micro-scale objects regardless of material transparency. Approaches to acoustic trapping of micro-objects required large field intensities. They are usually generated directly by acoustic transducer, cannot be redesigned easily, nor can the corresponding acoustic radiation forces be modulated efficiently. Recently, the artificial structures, such as phononic crystal or metamaterials, can provide a facile way of controlling the field distribution of acoustic wave. These artificial fields may have potential application in acoustic manipulation. In this work, we demonstrate that a geometrical modulated trapping force can exert on an object as it is near the surface of structured artificial brass plate at resonance frequency. The mechanism and condition of this trapping effect are discussed via the field and force analyzing, respectively.

3:40

5pPA7. Parameter space for scattering and radiation forces for symmetric objects based on unimodular partial-wave s-functions. Philip L. Marston (Physics and Astronomy Dept., Washington State University, Pullman, WA 99164-2814, marston@wsu.edu), and Likun Zhang (Physics and Astronomy Dept., Washington State University, Pullman, WA 99164-2814)

For situations in which dissipation is negligible it can be helpful to parameterize the scattering and radiation forces on axisymmetric objects using partial-wave s-functions [L. K. Zhang and P. L. Marston, Phys. Rev. E 84, 035601 (2011)]. The reason is that without dissipation, for axisymmetric objects illuminated by axisymmetric beams along the axis of the object, for each partial wave the scattering becomes characterized by a single parameter: the partial-wave phase shift. As a consequence of the finite diameter of the object, only a limited number of partial waves need to be included in the analysis of the radiation force and the scattering. In the case of a sphere illuminated by an electromagnetic beam, there are separate electric and magnetic partial waves. In the simpler acoustic case, the phase shifts and the cone angle of a Bessel beam may be used to characterize regions of negative radiation force and large scattering asymmetry. Fabrication of the associate object becomes a separate issue. It is not difficult to select the partial-wave phase shifts such that reduced backscattering with negative radiation forces occur with a cone angle of the beam as small as 22 degrees. [Supported by ONR and NASA.]

Contributed Papers

4:00

5pPA8. An experimental study of acoustically induced rocking motion of simple asymmetric geometries. Gwendolyn V. Rodgers, Peter H. Rogers, and James S. Martin (George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332, grodgers3@gatech.edu)

A series of experiments were done to test theoretical predictions that an acoustically small, rigid (aluminum) hemisphere and semicircular cylinder excited by a plane acoustic wave in water would each produce a rocking motion perpendicular to the incident wave direction about the axis parallel to the incident wave. A plane standing wave was generated in a short open-ended thick-walled cylindrical waveguide with the waveguide's axis perpendicular to the symmetry axis of the hemisphere. Measurements were taken along the hemisphere from top to bottom to determine if any rocking motion occurred. The expected vibrational motion in the incident direction and symmetry-forbidden perpendicular vibrational motion were also measured. The transverse displacement of the hemisphere at each point was determined by using an ultrasonic vibrometer. The incident wave direction motion was measured using a hydrophone, an accelerometer, and a Laser Doppler Vibrometer (LDV). The motion in the incident wave direction was found to be more than 10 times greater than the maximum rocking displacement for each case.

4:20

5pPA9. Negative optical torque. Jack Ng (HKUST, jack@ust.hk), Jun Chen, Zhifang Lin, and C. T. Chan

While it is now clear that optical scattering force can be negative (opposite to the propagating direction of the beam), less attention is being paid to the optical torque. Here, we show that the optical torque exerted on a structure by an electromagnetic wave can be in an opposite direction to the

angular momentum of the incident beam. This negative torque is closely related to the symmetry of the structure, and by playing around with the rotational symmetry and density of the structure, one can control the magnitude of the negative as well as the positive torque.

4:40

5pPA10. Optical micro-manipulation of absorbing particles with tractor and bottle beams. W. Krolikowski, V. Shvedov, C. Hnatovsky, N. Eckerskorn, and A. Rode

This paper presents an innovative approach of acoustic design of high performance facade for buildings in the vicinity of an airport. The sound insulation performance of the facade system is established by an aircraft noise auralization of the aural environment in the built environment such as a Star hotel at Arup SoundLab for determining an acoustically suitable glazing system. Arup SoundLab uses a 3D loudspeaker reproduction system called Ambisonics to reproduce the sound phenomena in a dedicated room that allows people to listen to simulated sound in virtual spaces before actual construction. The audio samples can be generated to mimic the dynamic indoor acoustic environment for the takeoff noise profile of B747 aircraft. After listening to the audio samples for different glazing configurations and ambient conditions, subjective responses from a group of representative individuals to the audio sound track that simulates a possible future situation are collected for evaluation analysis. The engineering development of high performance acoustic facade is presented including laboratory testing to reach a confidence level for achieving the end results. An opportunity for identifying potential cost saving is also discussed as a result of this aircraft noise auralisation. Examples will be presented to illustrate the approach.

FRIDAY AFTERNOON, 18 MAY 2012 S428, 2:00 P.M. TO 4:40 P.M.

Session 5pPP

Psychological and Physiological Acoustics: Subjective Evaluation and Auditory Perception

Dongxing Mao, Cochair
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Junji Yoshida, Cochair
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Invited Paper

2:00

5pPP1. Overall loudness perception properties under dichotic listening condition. Zhiyue Shao (Institute of Acoustics, Tongji University, No. 1239, Siping Road, Shanghai 200092, China, zyshao921@gmail.com), and Dongxing Mao (Institute of Acoustics, Tongji University, No. 1239, Siping Road, Shanghai 200092, China)

Dichotic listening conditions under which sound levels differ at the two ears are typical listening conditions, and there are yet no widely accepted results of binaural loudness summation. Researches show, the relationship between global loudness and interaural level difference (ILD) differs greatly for broadband noise and pure tone signals. Overall perceived loudness with interaural level difference (ILD) under different signal bandwidths was investigated through subjective listening experiment. In the subjective experiment, with the arithmetic mean of sound levels in both ears kept constant, the ILDs of dichotic signals ranged from 2dB to 12dB with 2dB step. Subjects were asked to match the overall loudness of dichotic test signals with that of diotic ones at the same center frequency and bandwidth. Signal of four bandwidths, which are 1/12, 1/6,1/3 and 1/1 octave were taken for the experiments. Results showed that the overall loudness of dichotic signals increases nonlinearly with ILD, and the narrower the signal bandwidth was, the faster the speed of the increase was. Comparison of subjective results with Moore's loudness model showed, deviation from Moore's model became larger as signal bandwidth narrowed.

Contributed Papers

2:20

5pPP2. Influence of SPL distribution on loudness impression for environmental noise. Yuta Chaki and Junji Yoshida (Osaka Institute of Technology, m1m11419@st.oit.ac.jp)

To realize comfortable living environment, it is necessary to reduce the environmental noise according to the suitable environmental noise evaluation index. For considering the environmental noise evaluation index, it is important to know characteristics of loudness evaluation toward long-term noise because the environmental noise was evaluated for long-term. In this study, to investigate the evaluation characteristics, we focused on the distribution of sound pressure level in duration of environmental noise and performed loudness evaluation test using long-term noise by changing the distribution. In the test, four traffic noises in duration of 10 min which loudness was adjusted equality by pre-processing were employed as base sounds. In addition, modified four noises in which the deviation of sound pressure level were increased (the mean value was same as the base sounds) were used as target sounds. Fourteen subjects evaluated the loudness of eight traffic noises immediately after hearing each sound. As a result, it was found that loudness impression toward the target sound was evaluated to as softer than the base sound. Consequently the long-term sound having large deviation of sound pressure level was clarified to be evaluated softer than the sound having small deviation.

2:40

5pPP3. Interaction between vehicle exterior design and perceived loudness of vehicle acceleration sound. Takumi Igata and Junji Yoshida (Osaka Institute of Technology, m1m11403@st.oit.ac.jp)

To increase cabin comfort, not only decreasing sound pressure level but also improving the sound quality is essential. In addition, when the sound quality of the vehicle matches with the exterior design, the sound is considered to be perceived more attractive. In this study, to investigate the relationship between the perceptions of the sound and the exterior design, the loudness of acceleration sounds were evaluated with vehicle exterior images. In the loudness evaluation test, 10 acceleration sounds and 10 exterior images were prepared and half of the images were luxury vehicle images and the others were sporty vehicle images. The experimental subject evaluated the loudness of presented sound with the luxury or sporty vehicle exterior image. Twenty one male subjects participated in the test. Eleven of them drove vehicles frequently and the others were infrequently. As a result, frequently driving subjects felt the acceleration sound louder by having luxury vehicle impression. In contrast, infrequently driving subjects felt the sound softer by having the same vehicle impression. Consequently, it was clarified that perceived loudness was influenced by vehicle exterior design, and the tendency was changed depending on the driving frequency.

3:00

5pPP4. Annoyance perception of environmental noises by hearing impaired listeners. Srikanth Vishnubhotla, Jinjun Xiao, Buye Xu, Martin McKinney, and Tao Zhang (Starkey Labs, Inc., 6600 Washington Ave. S., Eden Prairie, MN 55344, Srikanth_Vishnubhotla@starkey.com)

Annoyance perception of environmental noises is an important topic in many fields, including transportation, environmental studies and hearing aid design. While annoyance perception of normal hearing (NH) listeners has been studied extensively (e.g. Fastl & Zwicker 2006, Versfeld & Vos 1997, Alayrac et al 2010, Palmer et al 2006), data on annoyance perception of hearing impaired (HI) listeners is scant. In this study, we investigate the annoyance perception of typical environmental noises by both NH and HI listeners, using listeners with unilateral hearing loss. We use the magnitude estimation procedure to obtain the annoyance ratings and a paired-comparison method to obtain the annoyance preference for different environmental noises. The experimental data are analyzed to reveal the underlying dimensions of annoyance perception and differences between NH and HI listeners. A functional model for annoyance perception is developed for both HI and NH listeners. Finally, potential applications of our results are discussed in the context of hearing aid design.

3:20

5pPP5. Relative contributions of on- and off-frequency spectral cues in increment detection. Harisadhan Patra (Bloomsburg University of PA, 226 CEH, 400 E 2nd Street, Bloomsburg, PA 17815, hpatra@bloomu.edu), Petula Vaz (Bloomsburg University of PA, 223 CEH, 400 E 2nd Street, Bloomsburg, PA 17815), Abhijeet Patra, Joseph Motzko, and Adam Burkland (Bloomsburg University of PA, 338 CEH, 400 E 2nd Street, Bloomsburg, PA 17815)

Increment detection is often used as a measure of intensity resolution, where a listener detects a brief increment in intensity in a longer duration stimulus. Recent findings suggest that listeners rely on multiple cues rather than on-frequency energy cues. The relative contributions of on- and off-frequency cues are evaluated in five normal-hearing young adults. Detection thresholds for a 50-ms increment added in the middle of a 450-ms, 1000-Hz pedestal were obtained in quiet, telegraph noise (TN), and notched-noise (NN) conditions. The pedestal level was 60 dB SPL. NN stimuli were generated by filtering TN while keeping the notch geometrically centered at 1000 Hz. The NN-notches were also varied by increasing the notch-width unevenly on either side of the center frequency. The low-frequency cutoffs of the NN-notches were 125, 250, 500, and 750 Hz while the high-frequency cutoffs were 1500, 2000, and 4000 Hz. Although the NN-notches were centered at 1000 Hz, increment detection thresholds were poorer in NN conditions than in quiet. Detection thresholds were similar in TN and NN with the narrowest notch conditions. Results suggest that listeners may rely on a decision process based on multichannel on- and off-frequency cues. [Supported by BU research and scholarship grant]

3:40

5pPP6. Temporal order judgment characteristics at the offset of auditory-visual stimuli. Junji Yoshida, Kouhei Ueda (Osaka Institute of Technology, yoshida@med.oit.ac.jp), and Hiroshi Hasegawa (Utsunomiya University)

In this study, we investigated the influence of auditory-visual stimuli presentation timing at the onset on the perception of the offset asynchrony. To investigate the characteristic, we performed temporal order judgment test using auditory-visual stimuli. Pure tone and white rectangular were used as auditory and visual stimuli. In the test sequence, we prepared five presentation timings (0, $\pm$128, $\pm$256 ms) and nine disappearance timing (0, $\pm$32, $\pm$64, $\pm$96, $\pm$128 ms), and asked the experimental subject which stimulus disappeared first. From the obtained experimental result, we calculated the point of subjective simultaneity (PSS) at each presentation timing in each subject to obtain the PSS dependency on the onset presentation timing. As a result, the averaged offset PSS among the subjects was almost zero when the auditory-visual stimuli were presented simultaneously but the PSS varied widely among subjects. On the other side, with respect to the offset PSS dependency on the onset presentation timing, all subjects' result indicated the same tendency. The later the auditory stimulus was presented at the onset, the auditory stimulus was perceived to disappear earlier. Consequently, the temporal order judgment at the offset was found to depend on the presentation timing at the onset.

4:00

5pPP7. Ménière's disease impaired hearing character and improving strategy. Yan-Ping Zhang (Huazhong Univerity of Science and Technology, Wuhan 430074, pzzyy0102@163.com), Xian-Ming Long (Hanchuan City Hospital), and Chang-Cai Long (Huazhong Univerity of Science and Technology)

Hearing ability of Ménière's disease victim is usually impaired. They have a higher hearing threshold and a lower discriminating ability of speech, especially in noise enviroment. We demonstrate in psychoacoustics experiment that accompanying the usually observed hearing loss phenomena, the tuning shape of the disease victim is also changed into shallower. This phenomena can be explained by our physics theory about nonlinear cochlea signal procesing and a improving strategy is alao provided.

4:20

5pPP8. How do driving sounds influence the subjective evaluation of longitudinal acceleration? Sabrina Skoda, Jochen Steffens, and Joerg Becker-Schweitzer (Institute of Sound and Vibration Engineering, Duesseldorf University of Applied Sciences, Josef-Gockeln-Str. 9, 40474 Duesseldorf, sabrina.skoda@fh-duesseldorf.de)

The evaluation of vehicle interior sounds in a listening study is always based on multiple sensory interactions. In a driving simulator several visual and somatosensory stimuli are presented to the test subjects in addition to the pure driving sounds. These multisensory stimuli are processed consciously and unconsciously by the human brain. Thus attention processes and also the evaluation behavior change in comparison to monosensory acoustical experiments. In the automotive industry electric powered vehicles currently are a hot topic. These vehicles have a very quiet noise behavior compared to vehicles with combustion engines. So the question arises how driving sounds comprise to the subjective evaluation of a vehicle's acceleration. Do we associate loud driving sounds with strong acceleration? Which part takes the spectral structure of a sound? In a comparative listening study different vehicle interior sounds and different accelerations were evaluated in two different driving simulators. Based on the results from the study the influence of driving sounds on the subjective evaluation of longitudinal acceleration is discussed.