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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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 ($\triangle\alpha 0$) averaged in the region of interest, and to construct $\triangle\beta$, $\triangle\alpha 0$, and B-mode images simultaneously. During HIFU treatment, bubble activities were visible as strong hyperechoic regions in the B-mode images, causing fluctuations in $\triangle\beta$ and $\triangle\alpha 0$ during treatment. $\triangle\beta$ and $\triangle\alpha 0$ increased with the appearance of bubble clouds in the B-mode images to values in the range of 1.5-2.5 [dB/(MHz.cm)] and 4-5 [dB/cm], respectively. After the treatment, $\triangle\beta$ and $\triangle\alpha 0$ gradually decreased, accompanied by fadeout of hyperechoic spot in the B-mode images, until they were stable at 0.75-1 [dB/(MHz.cm)] and 1-1.5 [dB/cm], respectively. After treatment, $\triangle\beta$ and $\triangle\alpha 0$ images outperformed B-mode images by having significantly higher contrast to speckle ratios at all investigated TAP values.

Session 1aEA

Engineering Acoustics and Noise: Mufflers and Silencers

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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.

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.

Contributed Papers

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

Mounya Elhilali, Cochair
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Bosun Xie, Cochair
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Invited Papers

9:20

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

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

9:40

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

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

10:00

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

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

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

Contributed Papers

10:00

1aSCa1. Three- to five-year-old children in Taiwan show little development in their production of monosyllabic Mandarin lexical tones. 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
jjberry@email.arizona.edu

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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Wen Xu, Cochair
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James Preisig, Cochair
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Invited Papers

9:20

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

In 2000 Kilfoyle and Baggeroer authored a review of acoustic telemetry. Since then there has been rapid progress on all aspects of acoustic telemetry. There have been many experimental investigations which have highlighted the doubly spread features of the 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

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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.