Session 1pAA

Architectural Acoustics and Noise: Acoustics in Concert Halls I

Ning Xiang, Cochair
xiangn@rpi.edu

Jiqing Wang, Cochair
wongtsu@126.com

Zihou Meng, Cochair
mzh@cuc.edu.cn

Chair's Introduction—1:55

Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

6:00

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

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

6:20

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

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

6:40

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

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

MONDAY AFTERNOON, 14 MAY 2012 S224 + S225, 2:00 P.M. TO 7:00 P.M.

Session 1pBA

Biomedical Acoustics: Ultrasound Enhanced Drug Delivery

Constantin C. Coussios, Cochair
constantin.coussios@eng.ox.ac.uk

Juan Tu, Cochair
juantutu@gmail.comjuantu@nju.edu.cn

Invited Papers

2:00

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

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

2:20

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

Recent biomedical ultrasound development taking advantage of unique properties of ultrasound contrast microbubbles on ultrasonic molecular imaging, drug delivery and therapy has opened up powerful emerging applications. The intense radial oscillation of microbubbles makes them several thousand times more reflective than normal body tissues and emits significantly stronger acoustic signal. Besides, the oscillation resonance that microbubbles produce has several special properties that can be exploited to improve diagnoses and even for drug delivery and therapy. Moreover, selective ultrasound excitation can spatially-manipulate the microbubbles through the mechanism of acoustic radiation force and also destroy microbubbles to release the drug. In this talk, we present our current progresses on ultrasonic microbubbles as acoustic probes for molecular imaging, precise transportation micro-particles (drug), microbubbles and single cell to any specified loacation with MEMS technique by acoustic radiation force, as well as microbubble contrast agents as vehicles for ultrasound-mediated drug delivery in vitro and in vivo. The mechanism of the precise particle transportation using acoustic radiation force will also be discussed.

2:40

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

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

3:00

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

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

3:20

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

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

3:40

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

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

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

4:00–4:20 Break

Contributed Papers

4:20

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

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

4:40

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

Local drug delivery is studied to cross biological barriers and reduce the systemic side effects. Multimodal agents are being developed to combine step-response activation with the monitoring of its triggered release. The release can be triggered by acoustical or thermal means, e.g. using thermosensitive liposomes. Here we design an optically triggered microcarrier with well-controlled release precision and, in addition, a strong acoustic response in the far field, making the carrier a highly specific photoacoustic agent. The novel biocompatible microcapsules with a shell of fluorinated poly-L-lactic acid mixed with a fluorescent dye were produced with hexadecane oil core as drug-carrier reservoir. Single capsules were excited by a pulsed laser and their responses were monitored through combined ultra-high-speed imaging and sensitive acoustic detection. The experiments support a model where the polymer heats up through dye absorption thereby inducing the shell destruction and the vaporization of the surrounding water, resulting in the core release. Beyond the classical pulsed laser photoacoustics, capsules also respond to CW laser excitation by emitting a continuous signal, which offers promising opportunities for real-time photoacoustics. The subsequent study shows that the prolonged response results from repeated vaporization cycles and a complex interaction of the laser with the dyed polymer.

5:00

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

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

5:20

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

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

5:40

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

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

6:00

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

Encapsulating chemotherapeutic agents in liposomes improves targeting and efficacy of treatments against some tumors. The present work aims at evaluating if sonosensitive liposomes combined with cavitation for drug delivery enhance efficacy and reduce toxicity. Two focused beams were combined for stabilizing the cavitation cloud and an imaging probe used for guidance. Each 1MHz focused transducer had a 5cm diameter and focal length. Exposure conditions were 10.8kW/cm^2 Isppa, 250Hz PRF and 1% duty cycle. Phosphatidylcholine-based nongaseous liposomes were loaded with Doxorubicin. To control for mechanical tissue damage, AT2 Dunning tumors on rats were first exposed to ultrasound only. Treatment induced temperature rose below 0.5° C. The tumor growth was not significantly slowed down by ultrasound, but histological examination of tumors evidenced large areas of necrosis which resorbed one week after ultrasound. The new liposomes were compared with conventional HSPC-based liposomes in terms of efficacy and toxicity on the same tumor model. Ultrasound led to equivalent efficacy when applied on HSPC-based liposomes, while the new liposomes were efficient only with concomitant cavitation. We present a confocal ultrasound set-up able to provide sufficient inertial cavitation for drug release from a nongaseous liposome with reduced systemic toxicity. Eureka-labelled project (E!4056) funded by NRC, FFN and OSEO.

6:20

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

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

6:40

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

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

Session 1pEAa

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

Xiuming Wang, Cochair
wangxm@mail.ioa.ac.cn

Hailan Zhang, Cochair
zhanghl@mail.ioa.ac.cn

Contributed Papers

All posters will be on display from 2:00 p.m. to 3:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:00 p.m. to 2:30 p.m. and contributors of even-numbered papers will be at their posters from 2:30 p.m. to 3:00 p.m.

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

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

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

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

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

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

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

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

Session 1pEAb

Engineering Acoustics: Biomedical Transducers

Ling Xiao, Cochair
xling@mail.ioa.ac.cn

Contributed Papers

2:00

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

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

2:20

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

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

2:40

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

Synthetic transmit aperture (STA) imaging requires huge data amount due to its demand for high image quality, thus this increases the need for high performance hardware and limits the flexibility of the post-processing stages. Compressive sensing (CS) theory shows that the signals can be reconstructed from an extremely smaller set of measurements than what is generally considered necessary by Nyquist/Shannon theorem. In this paper, the CS theory is applied to STA imaging by sparse representation of the image in k-space. This method is tested with Field II simulation and the result shows that the quality of image is maintained with reduced data size by the application of CS theory with reduced sampling rates.

3:00

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

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

3:20

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

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

3:40

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

For therapeutic array transducers, it is required to reduce the electrical impedance of their elements so that the transducer can produce high ultrasonic power at a relatively low drive voltage. For this purpose, a new concept of concave hemispherical piezoceramic transducer element using its breathing mode has been investigated. The vibrational behavior of such a transducer is theoretically, numerically, and experimentally analyzed. Both resonance frequencies of the breathing-mode oscillation of a piezoceramic spherical shell and the volume oscillation of a water sphere are not only inversely proportional to their diameters, but also very close to each other at the same diameter. Numerical simulation of the transducer element showed high acoustical coupling achieved by the coresonance between the piezoceramic and the water sphere half enclosed by the shell. To confirm the effect by the coresonance, simulation replacing water by virtual materials, having the same acoustic impedance as water but different longitudinal velocities, was performed. The electrical impedance curves of the concave shell were very sensitive to the longitudinal velocities of the virtual materials, whereas those of the convex shell remained unchanged, which strongly support the hypothesis. Experimental results with a prototype transducer element will also be discussed.

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

6:00

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

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

Session 1pED

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

Siu Kit Lau, Cochair
slau3@unl.edu

Preston S. Wilson, Cochair
pswilson@mail.utexas.edu

Xiaojun Qiu, Cochair
xjqiu@nju.edu.cn

Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

Contributed Paper

3:40

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

The acoustics in the Department of Acoustic Science and Engineering, Nanjing University, has been evaluated as a state key subject for cultivating acoustic talents from undergraduate students to post-doctoral fellows. Our department prepares undergraduates for entry-level positions in the acoustic field and further education at the master's level. Our educational programs are organized around two overlapping areas: Acoustics and Acoustical signal processing. Core courses include fundamentals of acoustics, acoustic measurement, and acoustic transducer. A series of advanced undergraduate courses have also been developed to provide students with formal training in acoustics, including: electronic acoustics, ultrasonics, architectural acoustics, audio signal processing, active noise control, and etc. The goal of this program is to prepare students with both solid foundation in acoustic science and capabilities in acoustic engineering.

4:00–4:20 Break

Invited Paper

4:20

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

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

Contributed Paper

4:40

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

School of Marine Engineering (SME), Northwestern Polytechnical University (NPU) offers two undergraduate degree programs bearing on the field of acoustics, including information countermeasure and environmental engineering, of which the former is focused on underwater acoustic signal and information processing and the latter on environmental acoustics and noise control. The SME also provides graduate degree programs involving Acoustics, Underwater Acoustic Engineering, Environmental Engineering and Environmental Science leading to a master's degree, among which the former two also possess competency of doctoral education. As a key subject, the education respect to acoustics has developed its own characteristics of teaching reform. The SME is actively involved in fostering the talent of engineering and internationalization among the students. On the one hand, the school is devoted to improve practical and experimental skills of the students. On the other hand, recent years, Acoustics program at the school is gradually promoting its international teaching program.

Invited Papers

5:00

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

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

5:20

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

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

Contributed Papers

5:40

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

Acoustical issues have been important subjects in Hong Kong since its progressive development into a cosmopolitan city. Its densely populated characteristics and vibrant nature are challenges often encountered by professionals of various fields involved in the design and operation of a variety of infrastructures and facilities, from the early stage of landuse planning to daily maintenance of plants and equipment. Acoustical issues have also become livelihood issues as people are demanding better quality of living in terms of their acoustical environment. Hong Kong has been putting efforts to cope with the need to address acoustical issues by the most fundamental means, i.e. education for almost all walks of life, aiming to promote knowledge as well as good practices so as to build a "sound" environment. This paper will describe the development of education for acoustics to meet the challenges in an extremely active city like Hong Kong. The partnering efforts among professional associations, academic institutions, and the industries in fostering professional knowledge and enhancing continual training will also be covered.

6:00

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

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

education for not only designers, but also users of acoustically designed environments is important. For example, knowledge of time alignment of loudspeakers for 5.1 channel surround sound is important for producing a better sound field. Scientific and acoustic knowledge can enrich and improve quality of life by utilizing acoustical technologies, products, services, and environments. An acoustic design class for general students enrolled in liberal arts education has been introduced at the Tokyo University of Information Sciences. The main topics of the class are physical and psychological aspects of sound, noise and noise control, introduction to building acoustics, hearing impairment and hearing aids, soundscapes, music therapy, how to use audio products, sound pictograms, and sound design for video. A questionnaire survey administered to students showed that the class was useful for learning new schemes, technologies, and products concerning acoustic design in their daily life.

MONDAY AFTERNOON, 14 MAY 2012 — S227, 2:00 P.M. TO 5:20 P.M.

Session 1pHT

Hot Topics: 3-D Sound II

Yang Hann Kim, Cochair
yanghannkim@kaist.edu

Jeewoong Choi, Cochair
choijw@hanyang.ac.kr

Contributed Papers

2:00

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

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

2:20

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

For a special teaching application of the telepresence research project BEAMING, a scenario of a remote teacher (the Visitor) teaching a local student to play a xylophone through an embodiment is defined. In order to achieve this, a system is required to record, transmit and render the sound of the xylophone to the teacher in a dynamic scene. In an implementation of such a system, the xylophone is recorded using a mono recording technique. The signal is then processed to spread out the sound of the distributed sound source as multiple point sources in the virtual scene experienced by the Visitor. Finally head tracking allows for a dynamic binaural rendering of the xylophone sound. The goal of this paper is to evaluate the realism of this virtual (auditory) representation of a real xylophone. A listening test is designed to compare characteristics of a real physical xylophone in front of the listener with a rendering using the described system. The evaluation is done with a basis in methods previously used for evaluating the subjective sensation of presence in virtual reality systems, mainly based on questionnaires.

2:40

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

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

3:00

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

Dynamic cues induced by a listener's movement markedly improve sound localization (e.g. Kawaura, 1991; Iwaya, 2003). Using a simplified version of an avatar robot called TeleHead (Toshima, 2004), this study investigated effects of horizontal head rotation on median plane localization. The head of our robot can rotate horizontally, synchronously following a listener's head rotation. Sound signals at the robot's two ears in an anechoic room are captured and reproduced for a listener in a remote soundproof room. The robot rotation was controlled to have in-phase or anti-phase rotation with the listener's head rotation with a ratio between the rotation magnitudes of a listener and the robot of 0.05, 0.1, or 1.0. Results show that the anti-phase dynamic cue increases front-back confusion when the ratio is 1.0, but the localization was little affected when it was 0.05 or 0.1. In contrast, the in-phase dynamic cue suppresses front–back confusion significantly, irrespective of the rotation magnitude. Consequently, the sound localization accuracy can be improved considerably if the robot head's direction of rotation in a remote site and that of a listener are identical, even if the robot head rotation magnitude is as little as 5 % of the listener's. (Work supported by MEXT, Japan)

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

Session 1pNSa

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

David Woolworth, Cochair
dave@oxfordacoustics.com

Jun Yang, Cochair
jyang@mail.ioa.ac.cn

S.K. Tang, Cochair
besktang@polyu.edu.hk

Contributed Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

There are some difficulties for conventional Near-field Acoustic Holography (NAH) to analyze acoustic radiation from a large scale structure. To solve the problem, a method for NAH of large scale structures was presented. In the method, the normal velocities or sound pressure at a few points on the surface of the structure are measured by transducers, and that at other position on the surface of the structure are calculated by means of the radiation mode theory, then the radiated acoustic field may be analyzed by NAH. Since complex coupling terms no longer appear in the radiation modes, and only a few orders of modes are required to describe the acoustic field at low-to-mid frequencies, the accuracy of NAH analysis may be ensured. Moreover, according to the nesting property of radiation modes, the radiation modes at other frequencies can be replaced by that at maximum frequency, consequently the calculating procedure may be simplified and the calculating speed quicken. Finally, NAH analysis of acoustic radiation from large scale structures was illustrated using a $1\text{m}\times1\text{m}$ simply supported steel plate. The results show that the radiated acoustic field can be reconstructed accurately under the circumstances of a few measurement points.

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

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

The radiated sound field of rotating machinery or reciprocating machinery has a significant periodically time-variant nature. This is a kind of non-stationary sound field and called cyclostationary sound field. In the conventional planar near-field acoustic holography(PNAH), this kind of sound field is treated as stationary field, so the information relating to the change of frequency with time will be loss inevitably. In this article, the cyclic spectral density(CSD) instead of the complex sound pressure was adopted as reconstructing physical quantity in the PNAH, and the cyclostationary PNAH(CP-NAH) technique was proposed. Meanwhile, focusing on the calculation complex of CSD and the accuracy of the cyclic nature extracted, the gathering slice method of CSD was proposed by referring time aliasing methods on time series. The experiment results illustrate that the cyclic nature of cyclostationary sound field may be extracted directly and the location of the source determined exactly as well.

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

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

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

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

MONDAY AFTERNOON, 14 MAY 2012 HALL C, 2:00 P.M. TO 3:40 P.M.

Session 1pNSb

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

Klaus Genuit, Cochair
klaus.genuit@head-acoustics.de

K.C. Lam, Cochair
kinchelam@cuhk.edu.hk

A. Lex Brown, Cochair
lex.brown@griffith.edu.au

Contributed Papers

2:00

1pNSb1. Measuring the exposure to sound samples in subjective experiments. Liang Yan, Kean Chen (College of Marine Engineering, Northwestern Polytechnical University, Xi'an 710072, China, lyan1832@hotmail.com), Florian Gomez, and Ruedi Stoop (Institute of Neuroinformatics, University and ETH Zurich, Winterthurerstrasse 190, 8057 Zurich, Switzerland)

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

2:20

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

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

2:40

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

Distortion Product Oto-acoustic Emissions (DPOAE) and Transient Evoked Oto-acoustic Emissions (TEOAE) were measured in subjects before and after attendance to live music. The changes measured were compared to the exposure levels measured at the position of the subject. The main objectives of this experiment were two fold: 1) to assess the validity of the proposed measurement protocol to measure changes in DPOAE and TEOAE after a concert; 2) to test the reliability of the oto-acoustic emission measurement system under field conditions; Initial results shows that it is possible to measure changes in hearing after exposures of relative short duration (<1.5 hours). There are large individual differences both in sound exposure levels as well as in the changes on oto-acoustic emissions produced by similar exposures. Current results will be presented.

3:00

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

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

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

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

It is well known that exposure to excessive noise for long durations can cause a significant noise-induced hearing loss (NIHL). Although most research has focused on occupational sources of NIHL, there is growing concern about the potential damage caused by non-occupational noise exposure such as that from portable stereo system headphones. The purpose of current study was to measure the sound level generated by headphones of portable stereo system, and provide hearing conservation guidelines. Using a B&K Torso and a personal computer, output sound levels across volume control set by thirty participants were measured from headphone driven by music samples of five different genres. Three different styles of headphones (in-ear, circum-aural, supra-aural) were used to determine if styles of headphone influence sound level inside of ears. The study results suggested that the supra-aural headphone used by listener in noisy environment shall set a higher volume, in a result of a larger sound level in eardrum. The authors would like to thank the Institute of Occupational Safety and Health of Taiwan for the support that made the completion of this work possible.

MONDAY AFTERNOON, 14 MAY 2012 — HALL C, 4:20 P.M. TO 6:20 P.M.

Session 1pNSc

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

James E. Phillips, Cochair
jphillips@wiai.com

C.M. Mak, Cochair
becmmak@polyu.edu.hk

Invited Papers

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

Contributed Paper

6:00

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

A new high-rise hotel and residential condominium building had elevator equipment rooms between residential spaces. During construction, elevator equipment noise was audible in adjacent unfinished residential spaces. The developer and architect requested evaluation by an acoustical engineer. Investigatory observations with acoustical and vibration measurements were conducted to determine sources and paths of vibration and noise transmission. Construction noise and the unfinished condition prevented measurements reflective of actual future conditions in occupied spaces, but 1/3 octave vibration measurements in the elevator equipment rooms and on wall surfaces of residential space indicated structure borne vibration transmission that could result in re-radiated audible sound. Sound spectrum measurements in the elevator equipment rooms compared with anticipated airborne noise transmission loss through the demising partition indicated potential levels of residual elevator equipment noise in residential spaces. Primary acoustic sources were determined by observations and validated by vibration and measurements in the equipment room. Anticipated noise levels due to airborne sound and structure borne vibration were compared to full-octave background noise Room Criteria (RC) to determine attenuation requirements. Recommendations for noise and vibration mitigation were developed. Mitigation measures implemented by the construction contractor will be enumerated with subjectively determine results.

MONDAY AFTERNOON, 14 MAY 2012 THEATRE 2, 2:00 P.M. TO 7:00 P.M.

Session 1pNSd

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

David Woolworth, Cochair
dave@oxfordacoustics.com

Wing Tat Hung, Cochair
cewthung@polyu.edu.hk

Ulf Sanberg, Cochair
ulf.sandberg@vti.se

Invited Papers

2:00

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

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

2:20

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

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

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

Contributed Papers

2:40

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

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

3:00

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

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

3:20

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

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

3:40

1pNSd6. A review of the interior noise and vibration characteristics of modern Chinese high speed train. Fusheng Sui, Anne Shen, Jiumei Cheng, and Minmin Yuan (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, sui@mail.ioa.ac.cn)

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

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

5:00

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

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

5:20

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

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

5:40

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

This study is primarily aimed at developing a measure to overcome the limited shielding performance of the noise barriers for the high-speed train. Up to the train speed of 300km/h and more the noise incidence angle and the source height change due to the pronounced aerodynamic noise source parts located at the higher positions compared to the height of the conventional rolling noise source. By means of the experimental analysis on the sound radiation characteristics and the sound pressure distribution around the noise barrier, a prototype of an acoustical attachment is porduced based on the analytical model calculation and numerical analysis. The principle of the Helmholtz resonator is used to optimize the acoustical impedance on the surface of the upper dege of the noise barrier. Using the model it was possible to find an appropriate acoustical property for the impedance according to the target frequency,sound incidence angle relative to the barrier and the receiver position. In this paper the results of the experiment in an anechic room and from the out door experiment are shown and discussed.

6:00

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

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

6:20

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

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

6:40

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

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

Session 1pPA

Physical Acoustics: Thermoacoustics

Anthony Atchley, Cochair
aaa9@psu.edu

Contributed Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

1pPA9. Factors influencing on matrix H of acoustic field modulation in the thermoacoustic system. Lihua Zhou (Key Laboratory of Cryogenics, Technical Institute of Physics and Chemistry, Chinese Academy of Sciences, Beijing, 100190, China; Graduate School of the Chinese Academy of Sciences, Beijing, 100049, China, zlht13@163.com), Xiujuan Xie, and Qing Li (Key Laboratory of Cryogenics, Technical Institute of Physics and Chemistry, Chinese Academy of Sciences, Beijing, 100190, China)

In the system of acoustic field modulation in the regenerator by double loudspeakers (J. Acoust. Soc. Am. 130(5), 2709-2720 (2011)), the relationship among the pressure, velocity and driving voltages is described by a matrix H. H is a function of configuration and location of regenerator, operating frequency and temperature gradient between heat exchangers. The modulation range of acoustic field will vary with the variation of H. In this paper, the influences of factors (y0/l0, L, f, dTm/dx) on the matrix H are researched. The order of these factors effecting on matrix H is given and discussed. The results show that location L and frequency f have important influence on matrix H than y0/l0 and dTm/dx, which are applicable in acoustic field modulation.

5:20

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

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

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

5:40

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

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

6:00

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

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

MONDAY AFTERNOON, 14 MAY 2012 — S423, 2:00 P.M. TO 6:00 P.M.

Session 1pPP

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

Mounya Elhilali, Cochair
mounya@jhu.edu

Daniel Pressnitzer, Cochair
daniel.pressnitzer@ens.fr

Bosun Xie, Cochair
phbsxie@scut.edu.cn

Invited Paper

2:00

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

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

Contributed Paper

2:20

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

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

Invited Papers

2:40

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

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

3:00

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

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

Contributed Paper

3:20

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

Cochlea, known as 'fourier analyzer' in auditory system of mammals, codes the mechanical wave to electrical signals tonotopically. Meanwhile, the auditory coder also works in temporal mode, where envelope (E, i.e. amplitude modulation (AM)) and temporal fine structure (TFS, i.e. frequency modulation (FM)) are both significant but not exactly defined. In several recent researches (e.g. [Smith, et. al, 2002]), hilbert transform (HT) was utilized to extract the E and TFS cues. Though HT is mathematically rigorous, it is lack of clear physical meaning on AM-FM. This research introduces the empirical mode decomposition (EMD) to auditory research field. EMD was designed for analysis of non-linear and non-stationary data including natural sound. It is observed that the outputs of a auditory filter (e.g. Gammachirp), which was born as narrow-band filters, are approximately intrinsic mode functions (IMF) that admit well-behaved hilbert transforms (implying having physical significance). However, non-narrow-band filter (NNBF) output can be decomposed into two obvious IMFs or more, implying that HT is not suited to NNBF. EMD with HT provides auditory researchers one more perspective on AM-FM decomposition of auditory signals.# This work is supported by National Natural Science Foundation of China (11104316) and Shanghai Natural Science Foundation (11ZR1446000).

Invited Paper

3:40

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

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

4:00–4:20 Break

Contributed Papers

4:20

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

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

4:40

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

By dynamically synthesizing binaural signals in free-field and reflective environment, a virtual auditory environment (VAE) real time rendering system recreates realistic auditory events or perceptions for listeners. VAE systems have been applied in various fields, such as the research of binaural hearing, multimedia and virtual reality, among others. In present work, a PC and C++ language-based VAE system is designed and implemented. Schemes for improving the performances of the system, including multiple virtual source synthesis, auditory distance perception, dynamic information simulation for multiple degrees of freedom of listener, as well as dynamic characters of the system, are proposed. The results from measurement indicate that the system is capable of synthesizing 280 virtual sound sources (including free-field sources and image sources for reflections) simultaneously in conventional working mode, or 4500 virtual sources in proposed PCA (principal components analysis) working mode. The update rate is 120 Hz, and the system latency is 25.4ms. A set of psychoacoustic experiments also validate the performance of the system. The function extension of VAE can serve as a flexible and powerful platform for binaural and virtual reality research.

5:00

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

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

5:20

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

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

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

5:40

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

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

MONDAY AFTERNOON, 14 MAY 2012 — S222, 2:00 P.M. TO 4:40 P.M.

Session 1pSA

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

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

Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

Acoustic radiation modes have received increasing attention in the areas of structural acoustic radiation and active structural acoustic control during the past few years. In this paper, a comparative study on the acoustic radiation modes of several typical structures and their associated radiation efficiencies is presented. The present work is also undertaken to extend the acoustic radiation modes into more complex structures, i.e. a thin cylindrical shell, since it was rare described in the literatures. The two analytic approaches for deriving acoustic radiation modes are reviewed briefly first. Numerical examples and comparative analysis are performed from one-dimensional problem to two-dimensional and three-dimensional problems. A grouping characteristic of acoustic radiation modes and their corresponding radiation efficiencies is observed. The shapes of the acoustic radiation modes follow the order of uniform variation, linear, quadratic and high-order variation. It is interesting that the acoustic radiation modes of a cylindrical shell exhibit symmetric and anti-symmetric shapes in the circumferential direction, similarly to its structural modes.

3:40

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

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

4:00–4:20 Break

4:20

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

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

Session 1pSC

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

Estella Ma, Cochair
estella.ma@hku.hk

Benjamin Munson, Cochair
munso005@umn.edu

Contributed Papers

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

1p MON. PM

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

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

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

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

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

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

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

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

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

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

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

Session 1pUWa

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

Daniel Rouseff, Cochair
rouseff@apl.washington.edu

Wen Xu, Cochair
wxu@zju.edu.cn

James Preisig, Cochair
jpreisig@whoi.edu

Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

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

Contributed Papers

3:00

1pUWa4. Sequential analysis for underwater communications. Andrey Morozov (Teledyne Benthos TWR, 82 Technology Park Drive, East Falmouth, MA 02536, amorozov@teledyne.com), and Dale Green (Teledyne Benthos, 49 Edgerton Drive, North Falmouth, MA 02556)

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

3:20

1pUWa5. Adaptive equalization, tracking, and decoding for high-rate underwater acoustic communications. Andrew Singer (University of Illinois, Urbana-Champaign, 110 CSL, 1308 W. Main Street, Urbana, IL 61801, acsinger@illinois.edu)

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

3:40

1pUWa6. Experimental studies of support vector machine based blind equalization for shallow water channels. Wu Fei Yun, Zhou Yue Hai, and Tong Feng (Xiamen University, wfyfly@126.com)

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

4:00–4:20 Break

Invited Paper

4:20

1pUWa7. Focusing, doppler shifts and bubble screening: Parameterizing surface reverberation in different wind regimes. Grant Deane (Scripps Institution of Oceanography, UCSD, La Jolla, CA 92093-0238, gdeane@ucsd.edu), and James Preisig (Woods Hole Oceanographic Institution, Dept. of Applied Ocean Physics and Eng., Woods Hole, MA 02543)

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

Contributed Paper

4:40

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

The underwater communication channels limitations are mainly due to multipath effect and ambient noise. Multipath effect is a serious problem when underwater communication in the shallow water is considered, because it induced severe inter-symbol interference (ISI). Time reversal mirror (TRM) has been widely used in the underwater communication to overcome the multipath effect by evaluating impulse response function (IRF); and further, it can reduce bit error rate (BER). In this paper, underwater communication channel is explored in 175 m x 8 m x 4 m Towing Tank. Both numerical simulation and experimental are conducted to verify the effect of time varying multipath. Image method and statistic analysis are used to simulate the time varying multipath effect due to wave propagation in the shallow water. The BER caused by time varying multipath effect is evaluated by adding arrival time lag variation and arrival signal amplitude variation in impulse response function. The results show that time reversal process is effective to reduce BER and overcome the time varying multipath effect.

Invited Papers

5:00

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

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

5:20

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

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

Contributed Papers

5:40

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

Time reversal processing (TRP) has been proved to achieve temporal compressing when the waveguide environment is invariant. TRP has been exploited in underwater acoustic communications as it can, without any knowledge of the channel, compensate severe inter-symbol interferences (ISI) that are caused by complex multi-path propagation. However, environmental variations occur almost all the time when conducting acoustic communications in the real ocean, and indeed become one of the main factors determining the communication performance. Since the channel is time variant, the ISI can not be removed completely through time reversal. In the present paper, a channel tracking method is developed by using this leftover ISI after time reversal processing in a slowly time-variant environment. Change of the channel response structure is estimated in terms of time-delay shift and attenuation differences. Simulations of the method applied to synthetic data and field experimental results are both provided to demonstrate the method's feasibility. [Work supported by Chinese 863 high-tech program under Grant 2009AA093601]

6:00

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

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

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

6:20

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

A continuous data link under shallow and towing conditions is essential for UUV operation. In shallow water, inter-symbol interference (ISI) due to multipath fading strongly distorts the carrier signal. In addition, the phase of the signals shifts considerably due to the Doppler shift at a high relative speed (~5-kt) between the mother ship and the UUV. The communication performance degrades significantly compared to static communication. Actually, a 1-kt relative speed between the transmitter and receiver causes a phase shift of more than 2 Mπ within 1000 symbols of transported data. In this study, we investigated QPSK acoustic telecommunication systems that incorporate a Turbo corder and digital phase lock loop (DPLL) for the Doppler shift compensation. A towing experiment was conducted in shallow water, and the performance of the system was evaluated. When the receiver was 3 m deep, at speeds of 1.49 and 2.02 kt, we see a considerable number of errors. When the receiver depth was 15 m from the surface, all of the transmitted data were completely reproduced after applying our acoustic communication algorithm.

6:40

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

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

MONDAY AFTERNOON, 14 MAY 2012 S426 + S427, 2:00 P.M. TO 5:40 P.M.

Session 1pUWb

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

Nicholas Chotiros, Cochair
chotiros@arlut.utexas.edu

Ji-xun Zhou, Cochair
jixun.zhou@me.gatech.edu

Shengchun Piao, Cochair
piaoshengchun@hrbeu.edu.cn

Zhenglin Li, Cochair
lzhl@mail.ioa.ac.cn

Contributed Papers

2:00

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

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

2:20

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

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

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

2:40

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

Bottom characterization can be described by Geoacoustic model or bottom reflection model. In this work, simplified bottom reflection model in small grazing angle area was preferred. There are only two parameters P and Q in the model. P contains the phase shift information of bottom reflection. Q shows attenuation characteristics of bottom reflection. Spectrograms of reverberation data show evident coherent striations. The waveguide invariant, Beta, was extracted from these striations. For a Pekeris waveguide, a very simple analytic relation has been given: Beta = 1 + P/(k0Heff), here Heff is the 'effective depth', and Heff = H + P/2k0. P can be deduced using this simple relation firstly. The new developed energy-flux model (with parameters P and Q) of waveguide reverberation based on Perturbation theory was used to inverse Q from the reverberation average intensity decaying line secondly. 2007 Qingdao reverberation experiment data was analysed to illustrate this rapid bottom characterization technique. The result shows that P and Q can be extracted from the reverberation data rapidly. [This work was supported by NSFC under Grant No. 10874201 and No. 11074271]

4:40

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

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

5:00

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

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

5:20

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

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

MONDAY AFTERNOON, 14 MAY 2012

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

ELECTROACOUSTIC MUSIC PERFORMANCE

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

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

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