Session 2pAAa

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

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Chair's Introduction—1:55

Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

Session 2pAAb

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

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Jeffrey Mahn, Cochair
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Chair's Introduction—1:55

Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

Contributed Papers

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

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

5:20

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

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

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

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

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

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

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

Session 2pBA

Biomedical Acoustics and Physical Acoustics: Subharmonic Contrast Imaging

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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

Subharmonic scattering of phospholipid-shell microbubbles excited at very low acoustic pressure amplitudes has been associated with echo responses from compression-only bubbles. These bubbles are near a tension-free buckling state at rest and have initial surface tension values close to zero. In this work, subharmonic scattering of phospholipid microbubbles was investigated as a function of the initial surface tension, which was controlled by changing the hydrostatic pressure through the application of an ambient overpressure. Echo responses from a dilution of an experimental contrast agent were measured as a function of ambient overpressure ranging between 0 to 140 mmHg. The microbubbles were excited using a 64-cycle Tukey-windowed transmit burst with a center frequency of 4 MHz and peak-negative pressure of 50 kPa. Echo-power spectra were calculated, and the subharmonic response was determined for each overpressure value; the subharmonic amplitude increased by 20 dB after applying 140 mmHg overpressure (for which the initial surface tension was assumed to be near zero). In this study, an increase in subharmonic amplitude, instead of a decrease as reported by others [Shi et al., UMB 1999], was measured as a function of ambient overpressure. This observation may be exploited in a new method for noninvasive pressure measurement.

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

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

Contributed Papers

5:00

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

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

5:20

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

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

Session 2pEA

Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials II

Michael Haberman, Chair
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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

Contributed Papers

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

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

5:40

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

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

6:00

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

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

6:20

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

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

6:40

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

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

Session 2pHT

Hot Topics: Community Noise Policy Development II

Marion Burgess, Cochair
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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20–6:00

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

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

Session 2pMU

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

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2p TUE. PM

Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

Contributed Papers

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

Session 2pNSa

Noise: Numerical Methods in Noise II

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Contributed Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

Recent research was carried out to calculate the acoustics generation induced by flow numerically. In this paper, a modified hybrid acoustics/viscous splitting technique was employed in a computational simulation to calculate the acoustic generation when flow passes through a baffle in duct. The acoustics/viscous splitting technique for Computational Aeroacoustics (CAA) was first developed by Hardin and Pope. Traditionally, CAA problems were computed by Direct Numerical Simulation (DNS) to solve the compressible Navier-Stokes equation. However DNS is very expensive in terms of computation power and time consuming. The Hybrid approach is regarded as a alternative of solving the aeroacoustics problem. Perturbation technique is employed to calculate the acoustics perturbation which superimposes with the incompressible solution to obtain the compressible flow solution. In this paper, the hybrid CAA technique was developed and applied in a 2D duct with single baffle.

5:40

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

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

6:00

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

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

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

Session 2pNSb

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

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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

3:40

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

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

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

4:00–4:20 Break

Contributed Papers

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

6:00

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

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

Session 2pPA

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

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

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

Contributed Papers

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

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

5:00

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

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

5:20

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

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

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

Session 2pPP

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

Ruth Litovsky, Cochair
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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

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

Session 2pSA

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

Zhuang Li, Cochair
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Hongwei Liu, Cochair
lhw@mail.ioa.ac.cn

Wilson Ho, Cochair
who@wal.hk

Invited Paper

2:00

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

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

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

Contributed Papers

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

most contributing components. One of possible countermeasures of noise in zoom operating is investigated by an experimental approach

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

6:00

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

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

6:20

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

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

6:40

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

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

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

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

Session 2pSC

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

Charles B. Chang, Chair
cbchang@umd.edu

Contributed Papers

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Session 2pSP

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

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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

Contributed Papers

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

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

5:20

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

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

5:40

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

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

6:00

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

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

6:20

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

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

6:40

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

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

Session 2pUWa

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

John Colosi, Cochair
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Peter Worcester, Cochair
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Invited Paper

2:00

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

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

Contributed Papers

2:20

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

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

2:40

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

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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

5:00

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

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

5:20

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

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

5:40

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

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

6:00

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

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

6:20

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

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

6:40

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

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

Session 2pUWb

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

Marcia Isakson, Cochair
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Jin-Yuan Liu, Cochair
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Invited Papers

2:00

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

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

2:20

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

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

2:40

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

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

3:00

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

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

3:20

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

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

3:40

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

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

4:00–4:20 Break

4:20

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

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

4:40

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

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