Keynote Lecture

Invited Paper

8:20

Acoustics of traditional Chinese theatres. Ji-qing Wang (Institute of Acoustics, Tongji University, Shanghai, Shanghai 200092, China)

The traditional Chinese theatre is a unique architectural form. Chinese opera is a form of imaginary performing art; therefore, it does not require large stage and realistic stage settings. A pavilion stage open on three sides and thrusting into the audience area is its commonly applied characteristic feature. A comparatively low ceiling with elegant dome-like caisson acts as a sound shell, providing beneficial reflections to the audience, and to actors on the stage as well. The older generation Chinese opera goers used the term ?]going to listen opera??which well explains how they placed great demands on vocal performance. In Chinese theatrical history, there were different types of theatre from open-air theatre to hall theatre, built both in cities and rural areas all over the country. Nevertheless, the courtyard theatre was the most popular. Up to the present day, thousands of ancient traditional theatres still exist in China, and many of them are well preserved. Interesting results are reported in this paper after acoustical surveys of these theatres. Acoustical issues are raised from these studies, such as, does the classical parameter of reverberation time still adequate for qualifying the acoustics for a roofless courtyard theatre, or for an amphitheatre as well? A primary subjective survey conducted in our laboratory recently presents the negative conclusion. Another presentation is involved in this paper: the puzzle of vase resonators beneath the traditional stage which was long believed to be effective for sound enhancement as recorded in the Chinese historical accounts. The author also gives other acoustic analysis of the kind with pictorial presentations.

THURSDAY MORNING, 17 MAY 2012 HALL A, 9:15 A.M. TO 12:40 P.M.

Session 4aAAa

Architectural Acoustics and Noise: Healthcare Acoustics

Kenneth Roy, Cochair
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Erica Ryherd, Cochair
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Jerry Li, Cochair
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Chair's Introduction—9:15

Invited Papers

9:20

4aAAa1. Soundscape study for the improvement of neonatal intensive care units. Jennifer Nelson (UF School of Architecture, P.O. Box 115702, Gainesville, FL 32611-5702, jennifer.nelson@ufl.edu), and Gary Siebein (Siebein Associates, Inc., 625 NW 60th Street, Gainesville, FL 32607)

Guidelines for healthcare spaces address day and nighttime Leq and peak levels. However, there are many complex and transient sounds that make up the overall sound levels in healthcare environments. Many of these sounds contribute to the background level, while others are transient noises and alerts to professionals who must hear them to care for their patients. Unfortunately, these noisy environments are also where the patient is placed to heal. Three different Neonatal Intensive Care Units (NICUs) built in different years in Florida were observed and categories of sounds in each were documented. Overall level vs. time measurements made over a one week time period in each NICU were compared with WHO guidelines. Spectral level measurements of individual and combined sounds are also documented in each NICU. The individual sounds were classified into necessary and unnecessary criteria that orchestrate at all times of the day by observing and documenting them. The results of this study show how changes being made in the design and operation of contemporary NICU's are reflected in the measured sound levels, and what future changes can be made to further decrease unwanted noise.

4a THU. AM

9:40

4aAAa2. Evidence based design for hospital corridor noise control—Center for Health Design. Kenneth Roy (Armstrong World Industries, 2500 Columbia Ave, Lancaster, Pennsylvania 17604, kproy@armstrong.com), and Sean Browne (Armstrong World Industries, 2500 Columbia Ave, Lancaster, Pennsylvania 17604)

Armstrong participated as a member of a joint research group including the Center for Health Design and Palomar Pomerado Health in San Diego, California. The goal of this research was to evaluate the effects of flooring and ceiling choices on Corridor Activity Noise, and its perception by both patients and healthcare professionals for 2 material choices: 1. corridors with carpet and standard acoustical ceilings, and 2. corridors with hard flooring and high performance acoustical ceilings. This work was managed jointly by the PPH and CHD, the acoustic measurements were taken by CMSalter Associates, and materials and some data analyses were provided by Armstrong. Test results showed that substitution of a hard surface flooring material for carpeting resulted in a net increase in corridor noise levels with the expected patient and medical professional perceptions of increased annoyance and distraction. However, if the hard surface flooring is combined with a high performance acoustical ceiling, then the rise in noise due to the floor surface is negated with the added absorption of the improved ceiling, such that the result is equivalent in level and perception by both patients and staff.

10:00

4aAAa3. Emerging findings from the Healthcare Acoustics Research Team (HART). Erica E Ryherd (Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405, erica.ryherd@me.gatech.edu), Kerstin Persson Waye (Occupational & Environmental Medicine, Gothenburg Univ., 405 30 Gothenburg, Sweden), James E West (Electrical Engineering, Johns Hopkins Univ., Baltimore, MD 21218), Craig Zimring (College of Architecture, Georgia Institute of Technology, Atlanta, GA 30332), and Jeremy Ackerman (Dept. of Emergency Medicine, Emory Univ. School of Medicine, Atlanta, GA 30322)

Hospital patients, staff, and visitors need healthy soundscapes: patients need to sleep and heal without stress; staff, patients and family need to communicate accurately but privately; staff need to hear alarms and calls for help. Unfortunately, many hospitals are noisy and stressful places. Although there is growing and strong evidence that the hospital soundscape is problematic, there are many remaining questions and obstacles. This presentation will discuss recent case studies and findings from the Healthcare Acoustics Research Team (HART), an international, interdisciplinary collaboration of specialists in architecture, engineering, medicine, nursing, and psychology. HART is actively engaged in research in the United States and Sweden, having worked in a dozen hospitals and a broad range of unit types including intensive care, emergency, operating, long-term patient care, mother-baby, and others. HART seeks to advance the understanding of how various aspects of the hospital soundscape impact occupants, how to best measure and quantify these aspects, and how to translate results into evidence-based-design. Taken as a whole, these studies provide new insight into how to create healthier hospital acoustic climates.

10:20

4aAAa4. Experimentally investigated sleep disturbance of intensive care unit sounds. Kerstin Persson Waye (Occupational and Environmental Medicine, The Sahlgrenska Academy, Gothenburg University, Sweden, kerstin.persson.waye@amm.gu.se), Richard Wallenius (Gothenburg University, Sweden), Eva Maria Elmenhorst (German Aerospace Centre, DLR, Institute of Aerospace Medicine, Cologne, Germany), and Eja Pedersen (Department of Environmental Psychology, Lund University, Sweden)

Patients at intensive care often report fragmented sleep from noise due to care activities from personal, from other patients and alarms. The aim of this study was to explore the effects of original and modified intensive care noises on sleep in 15 healthy subjects. Their sleep was registered with polysomnography during four nights, one adaptation night, one control night and two exposed nights with similar equivalent sound levels of 47 dB LpAeq, but with either a maximum sound pressure level of 64dB LpAFmax or 56 dB LpAFmax. The subjects also answered questionnaires and saliva cortisol was sampled in the morning. The results showed that during exposure nights, subjects had less slow wave sleep and spend more time awake. No relation was found between arousals and maximum sound levels. Apart from an unexpected reduction of time in the REM-stage for the exposure with lower maximum level, there was no impact of the reduction of maximal levels for the sleep parameters recorded. The subjective data supported the polysomnographical findings while cortisol levels were not affected by the conditions. For healthy subjects the reduction of maximal levels from 64dBA to 56 dBA was not enough to improve sleep quality.

10:40–11:00 Break

11:00

4aAAa5. Speech privacy at community pharmacies. Yumi Koyama (School of Pharmacy, Nihon University, 7-24-1 Narashinodai, Funabashi, 7-24-1 Narashinodai, Funabashi, Chiba, Japan, koyama.yumee@nihon-u.ac.jp), Toshiki Hanyu (Department of Construction, Junior College, Nihon University, 7-24-1 Narashinodai, Funabashi, Chiba, Japan), and Kazuma Hoshi (Department of Construction, Junior College, Nihon University, 7-24-1 Narashinodai, Funabashi, Chiba, Japan)

To identify practical ways to assess privacy protection at counseling area in community pharmacies, we conducted site-visit investigations at 84 community pharmacies including on 10 telephone interviews, and internet questionnaire survey on 160 patients. In the typical Japanese community pharmacies, space is small, the counseling area is open, and there are many patients who are waiting until their name is called. In that situation, busy and noisy and loose concentration, patients must try to accurately grasp medical information, and pharmacist are also working in the same situation. At the site-visits, we asked about aural or visual privacy issues and performed a psychological experiment to determine whether the patient-pharmacist conversation could be heard from the waiting seats for patients. At the internet questionnaire, we asked about counseling environment at community pharmacies. Responses to the site-visit investigation and internet questionnaire survey revealed that privacy-related problems were classified into 4 factors: physical environment (speech privacy), information sharing, pharmacist's social role, and a complex mechanism of the medical system. These factors appear to be interrelated, making it difficult to improve patient-centered care.

11:20

4aAAa6. Speech intelligibility in hospitals. Timothy Hsu, Mike Moeller, Jr., Arun Mahapatra, and Erica Ryherd (Georgia Institute of Technology, 771 Ferst Drive, Atlanta, GA 30332-0445, tissue@gatech.edu)

In hospitals, background noise has been shown to be problematic, not only for the patients but also for the staff. With respect to staff members, perceived stress and psychosocial factors can be affected negatively by noise. One particular factor that noise can inhibit is effective speech communication. Speech communication is essential for functions such as evaluation, admittance and treatment of patients. This paper will discuss results from measurements made in several different hospital units where traditional speech intelligibility metrics were analyzed. Additionally, newer analysis techniques such as Noise Occurrence Rates were investigated for their potential usefulness in speech intelligibility applications. Preliminary results show that in general, the hospital units show "poor" to "marginal" Speech Intelligibility (SII) qualitative scores. The results of this study help to better explain how speech is understood in various locations within the hospitals and can aid in hospital designs that support speech communication.

11:40

4aAAa7. Investigation into the acoustical performance of single stud steel wall assemblies. John LoVerde (Veneklasen Associates, 1711 16th St, Santa Monica, CA 90404, jloverde@veneklasen.com), Wayland Dong (Veneklasen Associates, 1711 16th St, Santa Monica, CA 90404), and Aaron Betit (Acentech, 33 Moulton St., Cambridge, MA 02138)

The draft "Interim sound and vibration design guidelines for hospital and healthcare facilities" by the Joint Subcommittee on Speech Privacy of the ASA includes recommended STC ratings for partitions between exam rooms. In a typical hospital, these partitions are about 15 feet high and constructed with 16 gauge studs at 16 inches on center. However, virtually all laboratory testing is performed on 8 foot high walls with 25 gauge studs at 24 in. on center. These tests are used for design and evaluation even though there is little published data on the effects of stud gauge and spacing and wall height on transmission loss. A previous study investigated the acoustical effects of stud gauge and spacing, and documented substantial changes in transmission loss and STC rating [A. Betit, "Performance Dtails of Metal Stud Partition," J. Sound and Vibration, 44(3), 14–16 (2010)]. A second testing program was established to extend the investigation to the effect of changes in wall height. Transmission loss (STC) tests were performed on drywall partitions of various heights and construction. The results of the testing program are presented.

Contributed Papers

12:00

4aAAa8. Aco(s)ustainability (Acoustics and Sustainability)—is a space truly functional for its intended use? Daniel Butko (University of Oklahoma, College of Architecture, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu), and Jack Randorff (Randorff and Associates Inc, 11 W Canyon View Dr, Ransom Canyon, TX 79366)

Numerous scientific studies defining acoustical values of common building materials and assemblies have been performed throughout history and then correlated to occupant health, productivity, and speech intelligibility within the defined space. Since occupant well-being should be a driving factor during design and construction phases, the effects of sound and noise should be considered an inherent component of sustainable design. The functionality of the inhabitable spaces for the intended purpose suddenly increases the scope of sustainability. Without knowledge of previous experiments or publications, various University of Oklahoma College of Architecture students were tasked with answering the following question: Is a space truly functional for the intended purpose? Various built environments were evaluated, basic SPLs and frequencies were documented, and the results were compared with published data for construction materials and methods including OSHA and ANSI regulations. Students were anonymously surveyed concerning acoustical conditions they experienced to define what they considered helpful and distracting conditions. This paper focuses on introducing architecture students to the intrinsic link between acoustics and sustainability, allowing an appreciation for both the art and science contributing to inhabitable space(s).

12:20

4aAAa9. Public address system reinstallation. Wilson Ho and Eddy Ng (Wilson Acoustics Limited, Unit 616, Technology Park, 18 On Lai Street, Shatin, Hong Kong, China, who@wal.hk)

In the last decade, new functions for the Public Address (PA) system have been continuously developed and integrated. On the other hand, the existing PA systems are being less repairable and maintainable due to lack of supply of old version components although the system and loudspeakers are still functioning well within the specifications after many years of use. It appears to be sensible way to reinstall the control and amplification system without replacing existing loudspeakers in order to accommodate the new function and zoning requirements. This approach is being tested in Lantau and Airport Railway (LAR) line in Hong Kong operated since 1998. This paper presents the pros and cons using this reinstallation approach in practice and investigates the feasibility and effectiveness through acoustic simulation and on-site verification test of sound coverage and speech intelligibility.

Session 4aAAb

Architectural Acoustics, Physical Acoustics, Noise, and Signal Processing in Acoustics: Development and Applications of Micro-Perforated Sound Absorbers (Lecture/Poster Session)

Christian Nocke, Cochair
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Jonathan Botts, Cochair
bottsj@rpi.edu

Chair's Introduction—9:15

Invited Papers

9:20

4aAAb1. Brief review on micro-perforated sound absorbers. Christian Nocke, Catja Hilge (Akustikbüro Oldenburg, Katharinenstr. 10, D-26121 Oldenburg, nocke@akustikbuero-oldenburg.de), and Jean-Marc Scherrer

The theory of microperforated panel sound-absorbing constructions has been introduced by D.-Y. Maa in 1975. Since then many variations of micro-perforated sound absorbing devices and materials have been introduced. Materials that have been used to be micro-perforated have been metal, wood, plastics and many others. Stretched sheets used as ceilings, wall coverings and other set-ups have been applied for more than 40 years. In 2001 a nearly invisible micro-perforation has been introduced to the stretched material making it highly sound absorptive. The classical set-up of a micro-perforated sound absorber consists of a micro-perforated panel in front of an air cavity. The sound absorption coefficient of these set-ups can easily calculated with a high accuracy according to the well-known approximation of D.-Y. Maa if all defining geometrical parameters (diameter of microperforation, distance between orifices, panel thickness and air cavity depth) are known. For other assemblies no closed calculation model exists so far. In this contribution measured sound absorption coefficients of various set-ups with micro-perforated materials as well as combinations with different porous materials will be presented.

9:40

4aAAb2. Coupled mode analysis of thin micro-perforated panel absorbers. Cedric Maury (Ecole Centrale Marseille, Laboratoire de Mécanique et d'Acoustique (LMA), CNRS UPR 7051, 31 chemin Joseph-Aiguier, 13402 Marseille cedex 20, France, cedric.maury@centrale-marseille.fr), Teresa Bravo (Centro de Acustica Aplicada y Evaluacion No Destructiva (CAEND), CSIC-UPM, Serrano 144, 28006 Madrid Spain), and Cedric Pinhede (Laboratoire de Mécanique et d'Acoustique (LMA), CNRS UPR 7051, 31 chemin Joseph-Aiguier, 13402 Marseille cedex 20, France)

The prediction of the isolating properties of lightweight Micro Perforated Panels (MPP) is a subject that has been intensively studied due to their important applications in a wide range of areas such as building acoustics and the aeronautic, astronautic and automotive industries. MPPs have been mostly considered as rigid structures, accounting only for inertia and neglecting any vibrating effects. However, simulation and experimental studies on thin MPPs have found that the absorbing performance can experience variations in the low frequency range from the results expected assuming a rigid structure. The work presented here is a theoretical and experimental study on the influence of panel vibrations on the sound absorption properties of thin MPP absorbers. Measurements show that the absorption performance generates extra absorption peaks or dips that cannot be understood assuming a rigid MPP. A theoretical model is established that exactly accounts for structural-acoustic interaction between the micro-perforated panel and the backing cavity without restriction on the absorber cross-sectional shape or on the panel boundary conditions. This model is verified experimentally against impedance tube measurements and laser vibrometric scans of the cavity-backed panel response. The effect of micro-perforations on panel-cavity or hole-cavity resonances is revealed through coupled mode analysis.

10:00

4aAAb3. (Micro-)Perforated wooden panels as sound absorbers. Adrian Eichhorn (Akustik Plus GmbH & Co. KG, a.eichhorn@eichhorn-holzwerkstaette.com), Michael Beckmann (EGGER Holzwerkstoffe Brilon GmbH & Co. KG), and Christian Nocke (Akustikbüro Oldenburg)

Different materials have been used as micro-perforated or perforated panels for applications as sound absorbers. The theory of microperforated panel sound absorbers introduced by D.-Y. Maa in 1975 is independent on the material of the panel. So also micro-perforations in wooden panels will give sound absorption. In combination with porous absorbants the efficiency of the absorbers set-ups can be improved. Modern manufacturing tools for wood and wooden veneers allow for perforations of submillimeter diameters of the single pores. Optically these perforations hardly change the impression of the wooden panel. Acoustically the combination of perforated wooden panels and a backing cavity give highly effective sound absorbers. Measured sound absorption coefficients of various set-ups with (micro-)perforated wooden panels will be presented. The new possibilities in design and applications in architectural acoustics will be discussed.

10:20

4aAAb4. Application of microperf and microslit absorbers. Peter D'Antonio (RPG Diffusor Systems, Inc., 651C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

The theory of microperf and microslit absorbers is well established. Theoretical predictions match measurements with a high degree of accuracy. A normal incidence impedance tube measurement of a microperforated metal panel will illustrate this agreement. In this presentation, I would like to discuss how these transparent and translucent foils and panels have been used as both acoustical and decorative elements in projects. The microperf design is available in polycarbonate and ETFE foils as well as panels. The foils can also be printed for light shading. Since polycarbonate transmits in the infrared, the translucent foils can be used with back lighting on radiant ceilings. Several projects will be presented. The microslit design is typically available in panels, which can be transparent, translucent and digitally printed in graphic designs and signage. A graphically printed microslit atrium project will be profiled illustrating the architectural acoustic potential of this approach. These relatively new absorbers add an important tool to the acoustical palette and this presentation will illustrate how they have been used in a wide range of projects.

10:40–11:00 Break

Contributed Papers

11:00

4aAAb5. Acoustical perforated facings: a synthesis study of theoretical and experimental developments. Luc Jaouen, Fabien Chevillotte, and François-Xavier Bécot (Matelys, 1 rue Baumer, 69120 Vaulx-en-Velin, France, luc.jaouen@matelys.com)

Perforated facings (including Micro Perforated Panels) or perforated ceiling tiles have been widely studied since the fundamental work by Uno Ingard. This work specially focus on flow modifications in the vicinity of the perforations leading to modifications of the reactance of the panel. This communication is a synthesis study of pioneer and recent works on this topic for linear flow regimes, clarifying and correcting some results. One aspect of this communication revisits the length corrections for circular, rectangular and slits perforated panels. A second aspect aims at accounting for perforated panel vibrations based on Biot's theory. Coupling this last approach to the one by Atalla & Sgard [J. Sound Vib. 303, 195–208 (2007)], the current work allows to model perforated facings as elastic-frame porous media. Simulations results are validated using a recently proposed method for the characterization of perforated facings. Based on these results, general trends for dimensioning the acoustical performances of perforated panels are drawn.

11:20

4aAAb6. Hybrid sound absorbers combining micro-perforated panels with conventional absorption mechanisms. Marc Buret (Vipac Engineers and Scientists, 279 Normanby Road, Port Melbourne, VIC 3207, Australia, marcb@vipac.com.au), and King Kwong Iu (NAP Acoustics Far East Ltd., Room 1811, 18/F., Hong Kong Plaza, 188 Connaught Road West, Hong Kong)

Combination of microperforated panels with optimised efficiency in the low frequency range and other sound absorption systems, that provide performance in the mid and high frequencies, is presented for two sound absorber designs. In the first instance, proprietary fabric cover and fibrous absorption have been used to extend the performance range of discrete microperforated absorber units by optimising sound absorption by the edge of the units. The second development consists of conventional Helmholtz resonator perforated panels that have been customised using a micro-perforated panel in view to tune and enhance the low frequency performance. Results of testing conducted in a reverberation chamber are presented.

11:40

4aAAb7. Noise attenuation by sonic crystal barriers made of microperforated units. Victor M. García-Chocano, Suitberto Cabrera, and José Sánchez-Dehesa (Wave Phenomena Group, Universitat Politècnica de València, Camino de vera s.n. (Edificio 7F), E-46022 Valencia, Spain, vicgarch@upvnet.upv.es)

This work studies the absorptive properties of periodic arrays of microperforated cylindrical shells. Structures made of cylinders 3 meters height have been constructed and their reflectance and transmittance spectra are measured in open air at normal incidence. A broadband strong attenuation is found in the low frequency region. Experimental data are supported by model simulations performed in the framework of multiple scattering theory. It is concluded that these structures in combination with high frequency absorbing units are suitable to produce general purpose broadband noise barriers. Work supported by ONR (USA) and MICNN (Spain).

12:00

4aAAb8. Modeling vibro-acoustics behaviour of micro-perforated structures using patch transfer function approach. J.-L. Guyader, L. Maxit (Laboratoire Vibrations Acoustique, Institut National des Sciences Applique'es (INSA) de Lyon, 69621 Villeurbanne, France, jean-louis.guyader@insa-lyon.fr), and L. Cheng (Department of Mechanical Engineering, The Hong Kong Polytechnic University, Kowloon, Hong Kong Special Administrative Region)

Micro-perforated structures with a backing cavity is a device for providing efficient noise absorption. In a practical and industrial setting, the efficiency of Micro-Perforated Structure (MPS) may be influenced by the vibro-acoustic behavior of the surrounding systems, the shape of the micro-perforated structure as well as different kinds of excitation. In this paper, the Patch Transfer Functions (PTF) approach is proposed to model the MPS behaviour in such Complex Vibro-Acoustic Environment The PTF method is a substructuring approach which allows assembling different vibro-acoustic subsystems coupled through surfaces. The proposed PTF formulation of the MPS is capable of taking the micro-perforations and the flexibility of the structures into account and allows easy prediction of the efficiency of a MPS in a practical vibro-acoustic environment. In order to validate the present approach, PTF results are compared with experimental measurements.

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

4aAAb9. A study on the sound insulation performance of obliqe micro-perforated absorbers in shading louver. Chu Wen-Sung, Wang Yu-Che, and Lai Rong-Ping (Department of Architecture, National Cheng Kung University, No. 1, University Road, Tainan 701, Taiwan, n78981222@mail.ncku.edu.tw)

Generally shading louver are used for shading and natural ventilation, but in noisy urban area, sound insulation is necessary, in the trend of green building design, shading natural ventilation and insulation are taken the same weighting, but it is contradictory, meanwhile, the performance of water proof also required in louvre. In study, we try to use oblique micro-perforated for shading louvre, and reach it sound insulation. Acoustics louvres normally use in silencers for Air conditioning duct, it is composited by Perforated plate any glass fiber, for the water proof season, we use oblique micro-perforated instead of glass fiber, and design and some influence factors those are specified of form, blade length, blade width, blade angle and air gap and blade is composed of an infill of sound absorption material enclosed by perforated sheet material, sound insulation of an acoustic louvres generally is not high, particularly at low frequencies. We designed several type of the louver and test it acoustical performance, and discussed it is low, median. high frequencies. Finally, we calculate it's R_w by ISO 140, by the way, we also asses it ventilation, lighting, shading, affect, it is very important in green building.

4aAAb10. Non-flammable woven acoustic flow resistive textile. Marek Kierzkowski (Marek Kierzkowski Acoustic Consultancy, P.O. Box 1217, Mountain Gate, 3156 Victoria, Australia, psowy@bigpond.net.au)

Flow resistive textiles are becoming more popular almost in all aspect of acoustic applications where sound absorption is a primary noise reduction countermeasure: industrial acoustics (reduced sound propagation in work places), architectural acoustics (shaping interior acoustic properties) and the automotive acoustics (bonnet liners, firewall insulators). While the automotive applications are not very demanding in terms of flammability, the industrial and architectural applications must comply with severe flammability restrictions. As it is today only non-flammable porous materials like mineral fibre, ceramic fibre or melamine foam would comply with stringent fire specifications. We will show that the wise choice of the flow resistive textile enable to widen the range of materials complying with actual standards. The traditionally good sound absorbers like polyester fibre or foam could then become available to architects again.

THURSDAY MORNING, 17 MAY 2012 | S424, 9:20 A.M. TO 12:40 P.M.

Session 4aAB

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

Wang Ding, Cochair
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Cynthia Moss, Cochair
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Invited Papers

9:20

4aAB1. Findings on bat sonar through Telemike system. Hiroshi Riquimaroux and Shizuko Hiryu (Doshisha University, 1-3 Miyakotani, Tatara, Kyotanabe, Kyoto 610-0321, Japan, hrikimar@mail.doshisha.ac.jp)

In order to understand how bats conduct echolocation recording what they listen to is essential. Then, we developed an onboard wireless telemetry microphone system (Telemike) for flying bats in our flight chamber. We also developed a sensitive microphone array system for field recordings and for the flight chamber. Some data through microphone array with Telemike will also be introduced. Both CF-FM (Hipposideros turpis, Hipposideros terasensis and Rhinolophus ferrumequinum nippon) and FM bats (Pipistrellus abramus, Miniopterus fuliginosus, and Eptesicus fuscus) were used in the flight chamber experiments, while only FM bats (Pipistrellus abramus and Eptesicus fuscus) were used for the field experiments. In the flight chamber experiment, through Telemike Doppler-shift compensation was confirmed from flying CF-FM bats. Echo amplitude compensation was found in both FM and CF-FM bats. The bats can fly without any collision against walls or their conspecifics in a narrow space. Telemike has recorded overlapped echoes originated from adjacent bats. How they extract weak echo signals in the situation where own echoes are masked by echoes of other bats may be revealed by Telemike experiments. Further, how bats get information from extremely weak echoes coming from a flying small insect will also be clarified in our experiments with Telemike. [Research supported by ONR grant]

9:40

4aAB2. A broad band dolphin mimetic sonar—inspiration and modification from the nature. Tomonari Akamatsu, Tomohito Imaizumi, Koki Abe (National Research Institute of Fisheries Engineering, Fisheries Research Agency, akamatsu@affrc.go.jp), Yasushi Nishimori, Young Wang (Furuno Electric Co., Ltd.), Ikuo Matsuo, and Masanori Ito (Department of Information Science, Tohoku Gakuin University)

Broadband techniques are getting popular for underwater sensing methods because of its high spatial resolution and target discrimination abilities. We have been developing a broadband split beam system to locate and identify each species in the ocean. Our system initially learned from dolphins and then modified architecture appropriately. For example, biological sonar sound was effective for the short range sensing to locate individual target, chirp sound provided clear target image for the long range up to 200 m. Multi angle scanning of a target was proved to be essential for the species discrimination in our system. It was just like a finless porpoises rolled their body possibly to enlarge sensing volume and change beam incident angles to a target. Unlike dolphins, the split beam system was not able to change transmitting beam directions. The sound incident angle to a fish was calculated using the body movement vector and the position of a target fish in the beam. Reconstruction of target strength spectrum according to the incident angle provided significant difference of species between jack mackerel and chub mackerel that has not been possible by conventional active sonar systems.

10:00

4aAB3. Immediate changes in whale hearing sensitivity. Paul E. Nachtigall (University of Hawaii, Marine Mammal Research Program, P.O. Box 1106, Kailua, Hawaii 96734, nachtiga@hawaii.edu), and Alexander Ya Supin (Russian Academy of Sciences, 33 Leninsky Prospekt, Moscow, Russia)

We have been examining the hearing of both the outgoing clicks and the returning echoes of actively echolocating odontocetes using evoked auditory potential techniques. In order to protect themselves from the loud outgoing sound while still maximizing the hearing of the acoustic echo return, odontocete echolocators appear to have developed both passive and active control of hearing. Passive control has been demonstrated by comparing hearing of their own outgoing signals to similar signals presented to them from the outside. Clicks produced by the animal itself are heard about 40 dB down. Active control has been demonstrated by a comparison of hearing outgoing clicks during target present and target absent trials. During target absent trials, when searching for targets, hearing is 20 dB more sensitive than during target present trials. The current critical question is: If the animal is warned that a loud sound is about to arrive, does it possess a mechanism of self-mitigation that will allow it to control its own hearing and reduce the level of the incoming sound? Initial results indicate that a false killer whale will reduce hearing sensitivity by at least 15 dB when warned that a 170 dB signal is about to arrive.

10:20

4aAB4. Localization of the moving object by echolocation. Ikuo Matsuo (Department of Information Science, Tohoku Gakuin University, 2-1-1 Tenjinzawa, Sendai 981-3193, Japan, and Neurosensing and Bio-Navigation Center, Doshisha University, 1-3 Miyakodani, Tatara, Kyotanabe, Kyoto 610-0321, Japan, matsuo@cs.tohoku-gakuin.ac.jp)

Using the echolocation, bats can capture moving objects in real 3D space. Bats emit the frequency modulation sound and can accurately localize these objects from echoes. The object's range could be estimated from delay times between the emitted sound and echoes from objects. These positions in 2D space could be estimated from the difference between delay times at two ears, and the accuracy of localization was dependent on the range accuracy, which was dependent on the frequency width of the emitted sound, the signal-noise-ratio (SNR), and the Doppler shift. It has been shown that the previous proposed model could accurately estimate each range of static objects by using the frequency modulation sound at the low SNR. However, it is unknown whether this model could estimate the moving object in 2D space. In this study, the echoes were measured from the rotating pole by emitting intermittently the LFM sounds. These echoes were analyzed by using the Gaussian Chirplet filters with a carrier frequency compatible with emission sweep rates. It was clarified that this proposed model could track the moving object by estimating object's position in 2D space at each time.

10:40–11:00 Break

11:00

4aAB5. Cochlear structural variants in echolocators. Darlene R. Ketten (Woods Hole Oceanographic Institution, Biology Dept., Woods Hole, MA 02543 and Harvard Medical School, Boston, MA 02114, dketten@whoi.edu), James Simmons (Brown University, Neurosciences, Providence, RI), Hiroshi Riquimaroux (Doshisha University, Graduate School of Life and Medical Sciences, Kyoto, Japan), Scott Cramer, and Julie Arruda (Woods Hole Oceanographic Institution, Woods Hole, MA)

Although microchiropteran bats and odontocete cetaceans operate in radically different media, both have sophisticated sonar capabilities and evident similarities in their ability to detect and analyze ultrasonic signals. This paper compares the similarities and differences of cochlear cytoarchitecture and its implications for ultrasonic encoding and acuity amongst these groups through the use of three-dimensional models obtained via micro-CT imaging of intact heads and temporal bones. Inner ear anatomy was fundamentally similar with notable parallels in fenestral placement and ratios, membrane dimensions, and neural density and distribution across bats and dolphins with common cochlear types. Specialist ears are present in both groups, suggesting that like some CF-CM bats, one or more odontocete species have cochleae with specialized basilar membrane "foveal" regions. Cochlear specializations in both groups are primarily linked to peak spectra of signal, expanded frequency representation, and may enhance tuning in adjacent ear segments by generating standing wave phenomena. [Supported by N45- US Navy Environmental Division and the Office of Naval Research]

Contributed Papers

11:20

4aAB6. An approach for moving target detection with airborne CTFM sonar. Yang Wang, Yong Xu, Benxi Cao, Jingyao Wang, and Jun Yang (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, wy8008@126.com)

Continuously Transmitted Frequency Modulated (CTFM) sonar transmits wideband acoustic signal, demodulates the echo signal, and calculates the target distance by acquiring the frequency of the demodulation output signal. Still objects can be accurately detected using CTFM sonar. However, moving target brings Doppler shift to the echo signal frequency, which makes the output signal frequency of CTFM sonar deviate from the accurate distance of the target. In this paper, a novel airborne sonar sensing approach is proposed. Using a modified transmitted signal, single moving target that appears in a static environment can be detected. Compared to traditional CTFM sonar, a single tone is added to the transmitted signal. The existence and radial velocity of a moving target can be calculated based on the single tone Doppler shift of the echo. Furthermore, the simultaneous kinematic information of the target can be extracted by the system algorithm. An experimental system is developed, and the result of experiments verified the feasibility of the approach.

11:40

4aAB7. Research on blind source separation of marine mammals signal processing under watercraft emitted noise. Zhang Liang and Guo Long-Xiang (Harbin Engineering University, 150001, qq-zhangliang@163.com)

As statistical independence exists between marine mammals sound and watercraft emitted noise, and among different organism signals, the paper introduces blind source separation (BSS) into marine mammal signals processing. BSS with single hydrophone is a special underdetermined blind separation problem, and BSS based on matrix calculating is no longer suitable.

The problem can be solved by expanding channels, and further study finds that second iteration of blind separation can improve the performance. Algorithm simulation and experimental data analysis show that not only marine mammals signal but also different organism signals can be separated by this method with single hydrophone. It is proved that the correlation coefficient of the separated signal is obviously improved, which lays the foundation for the feature extraction and recognition of marine mammals signal. Keywords-marine mammals signal processing; BSS; maximum signal noise ratio criterion; second iteration

12:00

4aAB8. Recovery cycles of inferior collicular neurons in the leaf-nosed bat, Hipposideros armiger. Jia Tang, Zi-Ying Fu (School of Life Sciences, Central China Normal University, Wuhan 430079, China, bobaytang@yahoo.com.cn), Philip Hung-Sun Jen (Division of Biological Sciences, University of Missouri-Columbia, MO 65211), and Qi-Cai Chen (School of Life Sciences, Central China Normal University, Wuhan 430079, China)

When stimulated with biologically relevant constant frequency–frequency modulation (CF–FM) sounds, the inferior collicular neurons of the CF–FM bat, Hipposideros armiger, either only discharged impulses to the onset (76%, single-on neurons) of the CF-FM sounds or to the onset of both CF and FM components of CF–FM sounds (24%, double-on neuron). Some neurons were single-on responders at low sound amplitude but become double-on responders at high sound amplitude. Single-on responders had longer latency and recovery cycle than double-on responders. While most neurons did not respond to the second sound when the paired CF-FM sounds overlapped, 3 single-on and 7 double-on neurons did such that they had "cyclic" recovery cycles with inter-pulse intervals. The different response latency and dynamic variation in the recovery cycle of these two types of neurons suggest they may serve as the neural basis underlying a bat's ability to perform echo ranging throughout different phases of hunting.

12:20

4aAB9. Echolocation beam shape and focusing in the false killer whale (*Pseudorca crassidens*). Laura Kloepper, Paul Nachtigall, and Marlee Breese (Hawaii Institute of Marine Biology, P.O. Box 1106, Kailua, HI 96734, kloepper@hawaii.edu)

Odontocete echolocation signals are thought to be focused by the melon and air sacs, although active focusing has yet to be demonstrated empirically. Because odontocete echolocation signals are variable and the emitted click frequency greatly affects the echolocation beam shape, investigations of beam focusing must account for frequency-related beam changes. Using a fine scale hydrophone array, we measured the shape of the echolocation beam and tested whether the echolocation beam of a false killer whale changed depending on target difficulty and distance while also accounting for frequency-related changes in the echolocation beam. The false killer whale produced a single-lobed echolocation beam that changed in size depending on target distance and difficulty which may be a strategy of actively controlling the emitted beam to maximize energy of the target echo.

THURSDAY MORNING, 17 MAY 2012 S224 + S225, 9:20 A.M. TO 12:40 P.M.

Session 4aBA

Biomedical Acoustics: Bone Quantitative Ultrasound I

Pascal Laugier, Cochair
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Invited Papers

9:20

4aBA1. Correlations between propagation characteristics of guided ultrasonic waves and long bone fatigue. Dean Ta, Zhenggang Zhang, and Weiqi Wang (Fudan University, 220 Handan Rd, Shanghai, China, tda@fudan.edu.cn)

Ultrasonic assessment of long bone has become a topic of interest in recent years. The objective of this paper is to analyze the propagation characteristics of guided ultrasonic waves in fatigue long bones, further to study the influence of varying elastic modulus (EM) of long bones on these characteristics. A hollow cylinder was used to mimic long bones and then to calculate the velocities and wave structures with various EMs. Besides, finite difference time-domain (FDTD) method was used to simulate the propagation of guided waves in long bones at different EMs. The results show that phase/group velocities and central frequencies of guided waves decrease with the decrease in EMs. However, the attenuation of wave modes decrease with increasing EMs. The simulated results of FDTD for all wave modes and parameters are in good agreement with theoretical values. Those results demonstrated that different modes have diverse sensitivity to the variation of EMs, and mode velocities, central frequencies and attenuations can reflect the change of EMs of long bone. Therefore, the propagation characteristics of guided ultrasonic waves may provide a feasible approach to evaluate the early stages of fatigue damage in long bones.

9:40

4aBA2. Progress and challenges in axial transmission measurements of guided modes in cortical bone. Jean-Gabriel Minonzio, Josquin Foiret, Ludovic Moreau, Maryline Talmant, and Pascal Laugier (CNRS - UPMC LIP, 15 rue ecole de medecine, 75006 Paris, France, jean-gabriel.minonzio@upmc.fr)

Cortical bone porosity has been evidenced as being a major if not the major 'footprint' of bone loss and fragility. Several studies report that cortical bone behaves like a waveguide. Measurements of guided mode wavenumbers together with appropriate waveguide modeling have therefore the potential for providing estimations of effective stiffness coefficients (which are largely determined by

cortical porosity) and also cortical thickness. However, data interpretation is challenging due to the heterogeneous, dissipative and irregular nature of the wave guide. Moreover surrounding and internal tissues modify the guided modes. This paper presents current progress by our group in the measurement of the wavenumbers of guided wave modes, using a multi-transmitter multi-receiver axial transmission probe. The guided mode wavenumbers are obtained after projection of test vectors onto the basis of the singular vectors of the transfer response matrix, at each frequency. The method has been validated, first on isotropic elastic and visco-elastic plates, then on bone-mimicking plate and tube phantoms made of transverse isotropic absorbing material. The effect of soft-tissue mimicking layers on top of a bone mimicking phantom has also been studied. Finally, preliminary in vivo testing of the approach on human radius will be presented.

Contributed Papers

10:00

4aBA3. A simulation and experimental study of long cortical bones fracture evaluation using lamb waves. Kailiang Xu, Runxin He, Dean Ta (Department of Electronic Engineering, Fudan University, Shanghai 200433, China, xukl@fudan.edu.cn), Yixian Qin (Department of Biomedical Engineering, Stony Brook University, Stony Brook, NY), Peng Sun, and Weiqi Wang (Department of Electronic Engineering, Fudan University, Shanghai 200433, China)

Ultrasonic guided waves are promising to evaluate fractured long cortical bones. Guided modes conversion, occurred always on the sites of fractures, contains rich information of bone geometric and material properties. The study is to analyze the correlation between Lamb modes conversion and crack depth of fractured long bones. Axial experiments were designed to investigate the influences of different deep fractures in steel-made bone phantoms and sheep diaphyseal tibias over Lamb modes propagation, especially modes conversion. For comparison, the phantoms were modeled by two dimension finite-difference time-domain method. Group velocities and energies of S0, A0 and converted modes were extracted and analyzed under varying crack depth conditions. Being consistent with the simulation, experimental data showed obvious modes conversions occur between S0 and A0 modes. The fracture positions can be predicted from converted modes velocities. It illustrated the modes conversions become gradually observable with the fracture depth increasing, which can be indicated by the parameter, converted energy proportion (EP). The sensitivity and practicality of EP to assess bone fractures were validated in quantitative analysis of phantoms, simulations and sheep tibias experiments in vitro. In conclusion, ultrasonic guided wave is a reasonable method for long cortical bone fracture detection and evaluation.

10:20

4aBA4. Determination of bone thickness: effect of group velocity filtering in multi-component waveguides. Petro Moilanen (Department of Physics, P.O. Box 35, 40014 University of Jyvaskyla, Finland, petro.moilanen@jyu.fi), Maryline Talmant (Laboratoire d'Imagerie Parametrique, UPMC Univ Paris 06, CNRS, UMR 7623, 75005 Paris, France), and Jussi Timonen (Department of Physics, P.O. Box 35, 40014 University of Jyvaskyla, Finland)

Previously, the Jyväskylä group has used the F11 guided wave of an empty tube as the reference dispersion curve for identification of in vitro bone thickness. The role of the marrow has remained an open issue. The objective of this study was thus to test several reference waveguides, empty, fluid-filled and fluid-coated tubes, in modelling of guided modes. For instance, fluid filling in a tube strongly modifies the spectrum of guided waves found for an empty tube, because of coupling of guided waves in the tube wall and in the fluid-filled cavity. However, it is shown that, using group velocity filtering on the signal component of highest amplitude, the effect of fluid filling can essentially be eliminated. The dispersion curve of an apparent F11 mode is obtained with a slight shift only at the lowest frequencies. This small shift means that the dispersion curve of this apparent mode is, in that frequency range, between those of A0 (2D plate) and F11 (3D cylinder), and can to some extent be interpreted as either one of these.

10:40–11:00 Break

11:00

4aBA5. Time-frequency spectrum segmentation method for separating multimodal guided waves in long bones. Zhenggang Zhang, Dean Ta, and Weiqi Wang (Department of Electronic Engineering, Fudan University, Shanghai 200433, China, zhangzg0311@gmail.com)

Ultrasonic guided waves have great potential for evaluating long bones. However, the measured signal often contains multiple wave modes because of the complicated characteristics of guided waves, which cause difficulties to further analysis. In this study, a hollow cylinder filled with a viscous liquid was used to model the long bone, and the multimodal signals were simulated using finite difference time-domain (FDTD) method. The time-frequency spectrum segmentation method was proposed to separate multiple modes. First, the Gabor time-frequency spectrum of multimodal signals was calculated. Second, a multilevel image segmentation algorithm, including the watershed and region growing, was used to find the corresponding area of each mode in the spectrum. Finally, time domain signals representing individual modes were reconstructed from these areas. The validations of this method were analyzed by simulated multimodal signals under different elastic modulus (EM) of long bones, with or without noise. The results showed that the Gabor time-frequency representations of the individual modes were in good agreement with the theoretical dispersion curves. This study suggests that time-frequency spectrum segmentation method can correctly separate multimodal guided waves, which provides a foundation for feature extraction of individual guided modes.

11:20

4aBA6. Circumferential cortical wave propagation at the proximal femur predicts bone strength. Julien Grondin, Quentin Grimal (University Pierre et Marie Curie, Paris, F-75006, France, grdjulien@gmail.com), Sandra Guérard (Arts et Metiers ParisTech, F-75013 Paris, France), Reinhard Barkmann, Claus Glüer (Universitätsklinikum Schleswig-Holstein, Kiel, Germany), and Pascal Laugier (University Pierre et Marie Curie, Paris, F-75006, France)

The importance of predicting hip fracture risk and the key role of cortical bone to maintain femur neck mechanical integrity both motivate one important aspect of the research presented here which is to focus ultrasound measurements on cortical bone at the femoral neck. Hypothesizing that the circumferential propagation at the femur neck may be predictive of femur strength, this in vitro experiment investigates the relationship between ultrasound circumferential propagation and femur strength. For nine femurs of women we measured: (1) the time-of-flight (TOF) of the first arriving circumferential wave guided by the cortical shell at the femoral neck; (2) structural features and density (BMD) using quantitative X-ray computed tomography; (3) femur strength in one-legged stance configuration with state-of the-art mechanical tests. Significant relationships were observed between TOF and mechanical parameters: failure load: R2=0.79; elastic energy: R2=0.63; apparent stiffness: R2=0.70; TOF was also well correlated with BMD in the inferoanterior quadrant of the neck, consistently with a circumferential propagation path along the thicker inferior cortex. Our results evidencing that circumferential propagation TOF is related to strength and reflects local properties of the femoral neck cortex offer perspectives for enhanced in vivo assessment of bone strength directly at the hip.

11:40

4aBA7. Photo-acoustic excitation and detection of fundamental antisymmetric Lamb mode in coated bone phantoms. Petro Moilanen (Department of Physics, P.O. Box 35, 40014 University of Jyvaskyla, Finland, petro.moilanen@jyu.fi), Pasi Karppinen, Timo Karppinen (Department of Physics, P.O. Box 64, 00014 University of Helsinki, Finland), Zuomin Zhao, Risto Myllyla (Department of Electrical and Information Engineering, P.O. Box 4500, 90014 University of Oulu, Finland), Edward Haeggstrom (Department of Physics, P. O. Box 64, 00014 University of Helsinki, Finland), and Jussi Timonen (Department of Physics, P.O. Box 35, 40014 University of Jyvaskyla, Finland)

Photo-acoustic (PA) imaging was combined with skeletal quantitative ultrasound (QUS) for multi-mode ultrasonic assessment of human long bones. This approach permits tailoring the ultrasonic excitation and detection to efficiently receive the fundamental antisymmetric Lamb mode (A0) through a coating of soft tissue. The method was tested on five axisymmetric bone phantoms of individualized wall thickness (1-5 mm) made of a composite material and coated with a layer (2.5 mm) of soft material that mimics the soft tissue. Signals were excited with a pulsed Nd:Yag laser at 532 nm wavelength and detected on the same side of the coated phantom with (i) a laser Doppler vibrometer (LDV) and for comparison also with (ii) a piezoelectric contact ultrasound receiver, scanning a source-receiver distance of 20-50 mm along the phantom. At a centre frequency of 50 kHz, a phase velocity consistent with that of the theoretically predicted A0 mode was identified in the recorded signals. Our results thus suggest that photo-acoustic quantitative ultrasound enables assessment of the thickness-sensitive A0 mode in bone through a layer of soft tissue. Ultrasonic in vivo characterization of the cortical bone thickness may thus become possible.

Invited Paper

12:00

4aBA8. Multiscale elastic imaging & modeling of musculoskeletal tissues. Kay Raum, Susanne Schrof (Charité-Universitätsmedizin Berlin, Julius Wolff Institute, Augustenburger Platz 1, 13353 Berlin, Germany, kay.raum@charite.de), Sara Tiburtius (Technische Universität Darmstadt, Fachbereich Mathematik, Dolivostr. 15, 64293 Darmstadt, Germany), Quentin Grimal (Laboratoire d'Imagerie Paramétrique, UMR CNRS 7623 - Université Paris 6, 15 rue de l'école de médecine, 75006 Paris, France), and Alf Gerisch (Technische Universität Darmstadt, Fachbereich Mathematik, Dolivostr. 15, 64293 Darmstadt, Germany)

Sophisticated technical materials that are used in everyday life are often inspired by nature. Hard biological tissues, e.g. mineralized tendons, bone and teeth are natural examples of achieving unique combinations and also great variability of stiffness and strength. In order to achieve these goals, bone uses various design concepts, e.g. reinforcing a soft and flexible collagen matrix by stiff, but brittle mineral particles, sandwich compounding of anisotropic (directional) films, weight reduction by directional pores and spongy networks. Although many details of the genetics, biology, pathology and mechanics of bone have been uncovered, we still lack of a detailed understanding of bone structure at the nano- and microscales. Towards this goal, both experimental data of heterogeneous elastic and structural parameters from all length scales (from the centimeter to the nanometer scale) and theoretical models that can simulate the deformation behavior based on these data are required. In this presentation the concept of multi-modal coupled multi-scale assessment of tissue properties (using quantitative ultrasound, synchrotron radiation μCT and vibrational microscopy) and modeling (using various homogenization techniques) will be presented with an emphasis of applications in musculoskeletal research, e.g. bone and cartilage healing.

Contributed Paper

12:20

4aBA9. Dependence of local wave velocity in bovine cortical bone on the decalcification. Kenji Fukui, Ryo Tsubota, and Mami Matsukawa (Doshisha Univ., Kyoto, Japan, kicomry38@gmail.com)

Bone is a composite material, mainly composed of HAp crystallites and type I collagen. It is known that the amount and orientation of HAp crystallites contribute to the "bone quality", which affects the bone elasticity. In this study, using a micro-Brillouin scattering technique which is able to evaluate wave velocity in the minute area, the effect of HAp amount on the velocity was measured. 36-plate-specimens in the plane of bone axis and radial directions were obtained from the middle part of a bovine femur. Wave velocity and HAp amounts were evaluated by the micro-Brillouin and XRD techniques, respectively. The specimens were then decalcified using ethylenediaminetetraacetic acid and measured again. Before decalcification, the average velocity was 5.06×10^3 m/s, and showed a moderate correlation with the HAp amounts (R^2}=0.56). After decalcification, the average velocity dramatically decreased to the value of 3.28×10^3 m/s, showing a strong dependence on the HAp amounts. In addition, the wave velocities except for the lateral part shows the moderate correlation (R^2}=0.30) before and after decalcification, which implies the possible effects of collagen on the wave velocities.

Session 4aEA

Engineering Acoustics: Flow Noise and Mitigation Methods

Randolph Leung, Chair
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Invited Paper

9:20

4aEA1. The comparison between passive and active methods of online cavitation detection. Jin Liu (College of Science, China University of Petroleum-Beijing, No. 18, Fuxue Road, Changping, Beijing, China; Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, liuj1314@gmail.com), Zhaoli Yan, Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China), and Wenxiao Qiao (College of Science, China University of Petroleum-Beijing, No. 18, Fuxue Road, Changping, Beijing, China)

Cavitation is the Achilles' heel of kinetic pumps and propellers. It can lead to performance degradation, structure vibration and noise, and bring about material erosion. Therefore some methods should be taken to detect cavitation. In this work, passive and active acoustics methods of online cavitation detection are set up to recognize cavitation and non-cavitation state. The former uses a hydrophone to receive emitted hydroacoustics signal. The signals from 10 kHz to 60 kHz are analyzed to extract features for pattern classification. The latter applies ultrasound to acquire flow field message. The ultrasound received is demodulated and the modulating signal is also analyzed for pattern classification. Experiments based on the two methods are carried out. Classification accuracy, computational complexity and installation difficulty are compared. Their applicability is also summarized.

Contributed Papers

9:40

4aEA2. Simultaneous measurement of density and viscosity of fluid using vibration of structure constrained by fluid. Deokman Kim (Hanyang University, deokman@hanyang.ac.kr)

Measurement of rheological properties of fluid using vibration of structure constrained by fluid. The fluid density and viscosity is the quantity to be measured and monitored during various manufacturing process. In this study, a real-time experimental method to simultaneously measure the density and viscosity of the fluid is proposed. The effects of fluids on flexural vibration of the beam structure partially immersed in fluid are analyzed. The density and viscosity have effects on the fluid-structure interaction. To analyze the fluid-structure interaction effects, the fluids are modeled as a simple support at one end of the beam. Using the proposed method, the density and viscosity of viscosity standard fluids were measured and its result was verified. The proposed method is advantageous in that the setup is possible to be installed in any fluid undergoing manufacturing process for real-time monitoring.

10:00

4aEA3. Flow noise from the transition region of an axisymmetric body in water. Xuegang Li, Kunde Yang, and Yuanliang Ma (Institute of Acoustic Engineering, Northwestern Polytechnical University, xuegang608@126.com)

Flow noise from transition region of an axisymmetric body is important for predicting the self-noise of a sonar mounted on an underwater platform. Numerical calculation of the flow noise for an axisymmetric body is presented and the diffracted loss on the head surface of the body is calculated by the geometrical theory of diffraction. The main physical features of flow noise are obtained. The flow noise in horizontal symmetry profile of the axisymmetric body is non-uniform, but it is omni-directional and has little difference in the cross section of the body. Based on the simulation, the noise power level increases with velocity to approximately the fifth power at high frequencies, which is consistent with the experiment data reported in the literature. Meanwhile, the flow noise received by the acoustic array on the curved surface has a stronger correlation than that on the head plane at the designed center frequency, which is import for sonar system design. Furthermore, the flow noises of two models with different shapes are compared and a rather optimum fore-body geometric shape is given.

10:20

4aEA4. Modelling and computation of boundary layer flow around body of revolution. Xie Hua, Shen Hong-cui, and Tian Yu-kui (P.O. Box 116, WuXi City, Jiangsu Province, China, xie621@163.com)

The characteristic parameters of boundary layer are inputs for the flow noise calculation, their change influence the power spectra of wall pressure fluctuations, so as to affect the analysis of the flow radiation noise. In this paper Hess-Smith boundary element method was adopted to model thick boundary layer for body of revolution. The corresponding code is developed. The computation of characteristic parameters of boundary layer for a body of revolution is carried out. The computed results including boundary layer thickness, shape factor, momentum thickness and friction coefficient are analyzed. The variation of characteristic parameter is obtained. The result showed that the code developed in this paper can be applied to the analysis of the body of revolution's 3D boundary layer calculation, which can offer input parameter for flow noise calculation. Key words: flow noise, characteristic parameter, boundary layer, body of revolution, Hess-Smith boundary element

10:40–11:00 Break

11:00

4aEA5. On the characterization of acoustic two-port sources using multi-load method. Hao Zhang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, zhanghao@mail.ioa.ac.cn), Tao Feng (Department of Mechanical Engineering, Beijing Technology and Business University, Beijing 100048, China), Chengguang Zhou, Bilong Liu, and Ke Liu (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China)

A multi-load method for determining the source data of acoustic two-port sources is presented. By eliminating the term of source strength, the scattering matrix can then be obtained by solving a set of nonlinear algebraic equations, and the source strength is determined by the scattering matrix and the directly measured spectrum matrix corresponding to one of the acoustic loads. Numerical simulations indicate that the proposed method is effective. The method has been tested on a loudspeaker and an axial flow fan in a duct. The source data obtained by this method show reasonable agreement with that measured by the direct method. The multi-load method avoids using external sources, and it can be used as an alternative method when the external source is not easy to realize in practice.

11:20

4aEA6. Flow-induced acoustic resonance prediction using the transfer matrix method. Fei Liu, Sam Yang, and Lou Cattafesta (University of Florida, Gainesville, FL 32611, U.S.A., lfeicq@ufl.edu)

Acoustic resonance in a flow piping system may trigger an occupational or safety issue and can lead to equipment damage. Computational fluid dynamics (CFD) simulations are often used to predict such resonance phenomenon. However, this approach is generally time consuming and requires specialized training and expensive software. In addition, CFD is not viable for routine design purposes due to its computational expense. In this study, an alternative plane-wave based model for a hydraulic piping system is therefore developed using the transfer matrix method (TMM). Such a model can offer a fast yet reasonably accurate prediction for self-sustained pressure oscillation in a piping system. The advantage of the TMM is the simplicity with which the transfer matrix of a system can be generated from a combination of the TMs of its subsystems via matrix operations. The hydraulic piping system under study consists of a duct with constant cross-sectional area, diffuser, nozzle, bends, valves and orifice plate. The TM of each component is developed and compared to either CFD predictions or available experimental data, the TM of the complete system is derived. Design recommendations are made to reduce and/or avoid resonance in the piping system.

11:40

4aEA7. A numerical study of the effects of a winglet on airfoils. Chao-Nan Wang, Chuan-Cheung Tse (National Taiwan University, No. 1, Sec. 4, Roosevelt Rd., Taipei, 106 Taiwan, wangcn@ntu.edu.tw), and Ya-Ju Chang

The purpose of this research is to investigate the effects of a winglet on aerodynamic noise of an airfoil. For simplicity, Reynolds Averaged Navier-Stokes equations combined with Realizable turbulence model are used to solve the turbulent flow. In order to verify the accuracy of flow field analysis, a uniform flow past a three-dimensional rectangular airfoil is analyzed and tracks the center line of the tip vortex. The agreement between the simulated and measured center line is good. For the sound field analysis, the flow induced noise around a rectangular airfoil is computed by the Broadband Noise Source (BNS) model. Proudman's formula was used to evaluate acoustic power per unit volume of aerodynamic noise. This study focuses on the sound power of aerodynamic noise generated by tip vortex when the flow passes through an airfoil and an airfoil with winglet. In order to understand the effects of the winglet on the aerodynamic noise, the different winglet characteristics are investigated and discussed. It is found that with a winglet the dynamic coefficient is improved and the generated sound power is also reduced by about 5.4 dB in this study.

12:00

4aEA8. Low-dimensional modelling of sound generation by a flow past a bluff body. K. H. Seid, Randolph C. K. Leung (Department of Mechanical Engineering, The Hong Kong Polytechnic University, kahimseid@googlemail.com), and Garret C. Y. Lam (Department of Building Services Engineering, The Hong Kong Polytechnic University)

Searching for a unified methodology for controlling aeroacoustics of common structural discontinuities (e.g. bluff body, open cavity, etc.) has been a major topic among aeroacoustics research community. However, constrained by the complexity and nonlinearity of the flow governing equations, the process of reduced order modeling is usually required in which a low-dimensional, reduced-order model is created for approximating the full high-dimensional dynamics of the flow unsteadiness for control implementation purpose. The present study aims to extend two model reduction approaches, namely Proper Orthogonal Decomposition (POD) and Dynamic Mode Decomposition (DMD) via Galerkin projection method, to develop the reduced-order models for full compressible flows. The versatility of these methods are evaluated and compared by applying them to the aeroacoustics of flow past a square cylinder. It is expected that the outcome of this study could facilitate the development of a unified and versatile closed-loop control methodology for effective aeroacoustics control.

Session 4aHT

Hot Topics: Aeroacoustics I

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Chair's Introduction—9:15

Invited Papers

9:20

4aHT1. Noise source identification in high speed jets based on virtual microphone arrays. Philip Morris (Penn State University, 233 Hammond Building, University Park, PA, 16802, pjm@psu.edu), and Yongle Du (Penn State University, 229 Hammond Building, University Park, PA 16802)

Phased arrays have become a popular experimental technique for noise source identification. These techniques are generally limited in resolution due to the number of available microphones. In addition, noise source models are relatively simple, involving either uncorrelated point sources or simple coherent line sources. The former limitation is not present in numerical simulations, where the number and location of virtual microphones is effectively unlimited. The present paper describes numerical simulations of high speed jet noise using a Detached Eddy Simulation turbulence model. The nozzle, which is included in the simulations, is typical of the geometries found in high performance military aircraft. An Immersed Boundary Method is used simulate the effect of chevrons at the nozzle exit. The far field noise is predicted using solutions of the Ffowcs Williams – Hawkings equation. Phased array results are presented for both a baseline and chevron nozzle and the differences are discussed. In addition, near field virtual arrays are sampled and analyzed to include both radiating and non-radiating components. The implications of the results from the far and near field arrays in terms of noise source characteristics are presented. The results are compared with available experimental observations.

9:40

4aHT2. Investigation of noise radiation from a jet engine inlet by direct numerical simulation. Sarah Parrish and Christopher Tam (Florida State University, Department of Mathematics, Tallahassee, FL 32306-4510, parrish@math.fsu.edu)

In a jet engine, strong tones are produced by the fan and are radiated out of the inlet. Such fan noise is an important contributor to the total aircraft noise during take-offs and landings. Experimentally, it has been found that the sound radiation patterns from in-flight tests are quite different from those measured in static conditions. What accounts for this difference? In the current work, the radiation problem is studied computationally using direct numerical simulation based on the most advanced computational aeroacoustics methods. Both static conditions and flight conditions are reproduced. A thorough study of the computed results involving static and flight conditions leads to a physical explanation of the observed difference in the sound radiation patterns. (Invited for presentation in the Aeroacoustics Session)

10:00

4aHT3. Numerical simulation of grazing incidence of sound waves on an acoustic liner. Christopher Tam and Nikolai Pastouchenko (Department of Mathematics, Florida State University, Tallahassee, FL 32306-4510, tam@math.fsu.edu)

Acoustic liner is extremely effective for suppressing fan noise of jet engines. A resonant acoustic liner consists of a face sheet with cavity backing. Numerous small holes are drilled on the face sheet. When a sound wave is incident on a liner, pressure on the liner surface alternates from high to low. At high pressure, fluid is forced into the liner cavities through the holes. At low pressure, the process is reversed. The oscillatory motion of the fluid masses at the hole-openings is crucial to the damping of the sound waves. However, the hole diameter is typically one millimeter or less. Because the holes are small, experimental measurements of the fluid motion around the hole-openings are difficult to perform. Hence, this task is best carried out by numerical simulation; as small holes are not detrimental to numerical computation. The objective of this investigation is to seek an understanding of the flow physics responsible for acoustic damping by a liner. In all previous investigations, only one resonator is simulated. In this study, a liner with eight resonators is simulated. This allows, for the first time, a study of the aggregated effect of multiple resonators on an acoustic field. (Invited for presentation in the Aeroacoustics Session)

10:20

4aHT4. Simulation of compressible flows using Hermite methods. Thomas Hagstrom (Southern Methodist University, P.O. Box 750156, Dallas, TX 75275-0156, thagstrom@smu.edu), Daniel Appelo (The University of New Mexico, Albuquerque, NM), Tim Colonius, Matthew Inkman (California Institute of Technology, Pasadena, CA), and Chang Youn Jang (Southern Methodist University, Dallas, TX)

Spectral element methods based on Hermite interpolation have a number of unique properties. First of all, the stabilization inherent in the interpolation process is sufficient to suppress nonlinear instabilities observed with other discretization schemes and leads to accurate linear transport of nonsmooth solutions. Second, and most important, they allow purely local time-stepping procedures limited only

by geometric domain-of-dependence requirements. Thus high-order Hermite methods maximize the computation-to-communication ratio and therefore they admit highly efficient implementations on multicore processors. In this talk we focus on the application of Hermite methods to simulate unsteady compressible flows. Examples will include the direct simulation of the aeroacoustics of a low Reynolds number subsonic jet, as well as studies of more basic sound radiating flows. The latter will illustrate the coupling of Hermite methods with more standard discontinuous Galerkin discretizations to handle physical boundaries.

10:40–11:00 Break

11:00

4aHT5. Effect of Riemann flux solver on the accuracy of spectral difference method for CAA problems. Junhui Gao, Xiaodong Li (Beihang University, Beijing, China, gaojhui@buaa.edu.cn), and Qiqi Wang (Massachusetts Institute of Technology, Cambridge, MA 02139, U.S.A.)

The spectral difference (SD) method is a new high-order method for unstructured grids proposed recently by Liu et al. (2006). In this paper, a two dimensional computation aeroacoustics (CAA) tool based on SD method is developed. Five Riemann solvers are implemented in the current code, including Roe scheme, advection upstream splitting method (AUSM), flux-vector-splitting scheme, Rusanov scheme, convective upwind and split pressure (CUSP) scheme. A comparison of these Riemann solvers is carried out with three CAA workshop benchmark problems. The relative error of each solver in simulating of entropy, vorticity and acoustic waves is presented. The accuracy of the SD method with each Riemann solver is obtained. It is found that the usually used Rusanov scheme is less accurate than other solvers. AUSM and CUSP schemes are more accurate than others in simulating acoustic waves. Meanwhile, the effect of mesh quality on the accuracy of SD method is investigated. Gaussian distributed random error is superimposed on a base mesh to change the mesh quality. The accuracy of each solver on the skewed mesh is presented and compared with the results on base mesh. It is shown that mesh quality has little effect on the accuracy of SD method if the mesh resolution is sufficient.

11:20

4aHT6. Assessment of nonlinear perfectly matched layer boundary conditions for CAA benchmark problems. Dakai Lin (Beijing Aeronautical Science & Technology Research Institute of COMAC, lindakai@comac.cc), Xiaodong Li (School of Jet Propulsion, Beihang University, China), and Fang Q. Hu (Department of Mathematics and Statistics, Old Dominion University)

Non-Reflecting Boundary Conditions (NRBCs) are very crucial for accurate numerical simulation of aeroacoustic problems. This paper aims to assess the performances of recently developed nonlinear Perfectly Matched Layer (PML) NRBCs by several Computational Aeroacoustics (CAA) benchmark problems through the comparison with the linearized PML, the characteristic and the asymptotic NRBCs. Numerical results show that the performances of the nonlinear PML NRBCs are tantamount to each other, and there is no substantial difference. But for strongly nonlinear cases, the error caused by using nonlinear PML NRBC is 1~2 orders of magnitude smaller than the one caused by using the linearized PML NRBC. Thus, using nonlinear PML is necessary in strong nonlinear aeroacoustic problems. Numerical tests also demonstrate that the nonlinear PML NRBCs outperform the characteristic NRBCs significantly, and have better performances than the asymptotic NRBCs.

11:40

4aHT7. Control of edge-scattering noise via permeable surfaces. Young J. Moon, Ikhyun Bai, and Seungtae Hwang (Korea University, School of Mechanical Engineering, Seoul 136-701, Korea, yjmoon@korea.ac.kr)

The edge-scattering noise generation mechanism is first studied, in line with the existing thoeries of Howe, Amiet, and others. Then the edge-scattering noise is controlled by attempting various permeable edges such as porous surfaces and slitted edges. The basic underlying mechanism of noise reduction is to be understood, examining the three-dimensional scattering of a line-vortex embedded in the laminar boundary layer over the flat plate with the porous and slitted trailing-edges. More realistic investigations will follow by the large-edge simulation (LES) of a turbulent boundary layer over the flat plate, solving the filtered, three-dimensional, compressible Navier-Stokes equations with the six-order compact finite-difference scheme and the four-stage Runge-Kutta method.

12:00

4aHT8. Control of weak perturbations. J. E. Ffowcs Williams (Emmanuel College, University of Cambridge, Cambridge CB2 3AP, U.K, jef1000@cam.ac.uk), and L. Huang (Dept of Mechanical Engineering, University of Hong Kong, China)

We define sound as being a weak perturbation in the properties of material consistent with the Navier-Stokes and continuity equations. Lighthill's pioneering paper on aerodynamic noise gives an exact theory that enables interesting connections to be made between flow and sound. Aerodynamic noise being caused by quadrupoles is a good point of view, but what caused the quadrupoles? Were they possibly initiated by sound? Conclusions deduced from such a theory are not necessarily helpful, but they are true and might be very helpful indeed. The linear perturbations we call sound obey linear rules and it can be suppressed by anti-sound, a subject now becoming both practical and useful. The same must apply to any weak perturbation of a dynamic system perturbed from rest. Some perturbations are unstable and grow exponentially in their early weak state. They might be eliminated altogether by suppressing their linear form. The Rijke tube experiment shows that to be practical and shows also the close similarity that exists between acoustics and control theory. The lecture will give more examples of that type and suggest others that have yet to be demonstrated.

Contributed Papers

12:20

4aHT9. A new algorithm for deghosting in passive acoustic air surveillance systems. Xuelei Zhang, Jie Feng, and Zhaoli Li (The Third Research Institute of China Electronics Technology Group Corporation, Beijing 100015, China, zhangxuelei2008@gmail.com)

This paper addresses the false association (called ghosts) problem of multi-target tracking in a distributed passive acoustic sensor network. To eliminate these ghosts, a new deghosting scheme based on the generalized triangulation has been proposed. First, the received angle information of different targets is reordered by using the gray theory to obtain the correct sequence. Successively, time span match triangulation based on pretreatment algorithm of angle association is used to get the most-likely position of different targets. Last, the residual ghosts are cancelled by using a reasonable hypothesis and third site notarization. The validity of the proposed scheme is evaluated using both simulation and experimental results.

12:40

4aHT10. A robust selection method of time-delay difference for DOA estimation. Zhiyu Li, Zhiguo Hou, and Zhaoli Li (The Third Research Institute of China Electronics Technology Group Corporation, lizhiyu79@126.com)

DOA estimation method base on time-delay differences is used widely in the field of passive source location because its solidity in interference circumstance. To meet the real-time demands of some location systems, a time-delay differences robust selection method is presented in this paper by analysis of DOA estimation errors. In this method, only part of time-delay differences is estimated for the measuring of DOA to reduce the computational complexity. Computer simulation results indicate the new method has less computational cost by contrast with traditional method, and based on the reasonable time-delay differences the DOA estimation results are more accurate.

THURSDAY MORNING, 17 MAY 2012 S425, 9:15 A.M. TO 12:20 P.M.

Session 4aID

Interdisciplinary: Workshop on Publishing Excellence in the Journal of the Acoustical Society of America

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

Invited Papers

9:20

4aID1. Gauging the likelihood for acceptance of a paper submitted to the Journal of the Acoustical Society of America. Allan Pierce (Acoustical Society of America, P.O. Box 339, East Sandwich, MA 02537, adp@bu.edu)

Authors contemplating submitting papers to the Journal of the Acoustical Society of America (JASA) should first determine whether JASA is an appropriate journal, as it is possible that submissions for which this is not the case will be immediately rejected. A principal criterion is whether the paper will find a wider readership with JASA publication than with an alternative journal. If the appropriateness for JASA may not be manifestly obvious to the editors, then the authors should submit a cover letter explaining why, and they should write their paper so that there is a clear tie-in with articles previously published in JASA, preferably recent articles. Authors are advised to select the Associate Editor who is most likely to be familiar with the subject matter of the paper. Given that the paper is appropriate and an Associate Editor can be identified who is willing to handle the paper, it will be subsequently judged for possible acceptance based on several criteria: the most important being whether the paper is (i) original, (ii) significant, (iii) clearly written, and (iv) suitably limited in scope. The significance criterion is discussed at some length. The talk is illustrated by several disguised examples of recent submissions which were rejected.

9:40

4aID2. Understanding the peer-review process in the Journal of the Acoustical Society of America. Ning Xiang (Graduate Program in Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, New York, 12180, xiangn@rpi.edu)

The Journal of Acoustical Society of America (JASA) is an archival, peer-reviewed journal that has served the acoustics community for over 80 years. A rigorous peer-review process often results in significantly improved manuscripts qualified for publication; it also allows relative academic freedom and fairness. To help prospective authors disseminate their research results and progress, the JASA editorial board regularly publishes detailed, updated information, and guidelines. In addition to following these guidelines, a better understanding of the peer-review process is also critically important for prospective authors. This talk will provide some insights into the different roles and the interrelationship between the author(s), and the reviewers. This talk will also discuss criteria for accepting manuscripts, and engagement of knowledgeable reviewers using some disguised examples.

10:00

4aID3. One person view on publishing in JASA and JASA Express Letters. Whitlow W. L. Au (Hawaii Institute of Marine Biology, 46-007 Lilipuna Road, wau@hawaii.edu)

One of the strengths of the Acoustical Society of America (ASA) is the multi-disciplinary aspects of the society. Engineers, physicists, mathematicians, biologists, psychologists, physiologists, and speech researchers plus others are part of our society. The ASA can be broadly divided into two groups, those in the physical sciences and those in the life or natural sciences with signal processing bridging both groups. JASA reflects this diversity. Manuscripts by those in the physical sciences depend heavily on mathematical and physical modeling and various types of equations, while in the natural sciences spectrograms, various types of statistics and hypotheses testing are often used. Nevertheless, certain factors with regard to quality must be met by all authors. The articles should contribute new knowledge, be written in English and be grammatically correct. The approach or methodology must be sound. The results should be clearly presented whether in tables, graphs or in verbiage. Finally, the discussion and conclusions should be to the point, clearly presented so the authors' arguments and points can be easily understandable. Associate editors should be assisting authors in getting their papers published. Responsibility for the content lies totally with the authors and not one bit with the associate editors.

10:20

4aID4. Publishing in the Journal of the Acoustical Society of America. Zhaoyan Zhang (UCLA School of Medicine, 31-24 Rehab Center, 1000 Veteran Ave., Los Angeles, CA 90095-1794, zyzhang@ucla.edu)

Peer-reviewed journals such as the Journal of the Acoustical Society of America provide researchers the best venue to distribute research accomplishments to a broad readership. While this publishing process may be time-consuming and sometimes frustrating, authors can take an active control of this process by preparing a good-quality manuscript and acquiring a better understanding of the peer-review process. The purpose of this talk is to share the present author's experiences as an JASA author and a current JASA associate editor in preparing a publishable manuscript for JASA. Suggestions are given on how to prepare a scientifically significant manuscript and how to benefit from the peer-review process to further improve the manuscript.

10:40–11:00 Break

11:00

4aID5. Authors' sharing on handling reviewers' comments. S.K. Tang (Department of Building Services Engineering, The Hong Kong Polytechnic University, besktang@polyu.edu.hk), and L. Cheng (Department of Mechanical Engineering, The Hong Kong Polytechnic University)

JASA is a highly reputable journal in acoustic community and publishing in the journal is very important to academia. In the universities in Hong Kong at least, a publication in JASA could affect staff appraisal outcome. The journal has very professional editors and a secretariat office, who screen all the submissions before sending them out to reviewers. To publish in JASA, quality of the work is certainly of prime importance. However, the responses to reviewers' comments can sometimes play an important role in the process, especially on controversial issues. We believe that authors and reviewers are equal in the whole process but comments from both the editors and reviewers must be handled professionally. In this presentation, we would like to offer our views from an author's perspective. In particular, we would like to share some experiences and discuss some issues which we think could affect the final decision on a submission.

11:20

4aID6. Guidelines for prospective authors to submit acceptable manuscripts to the Journal of the Acoustical Society of America. Sean F. Wu (Wayne State University, sean_wu@wayne.edu), and Ning Xiang (Rensselaer Polytechnic Institute, Graduate Program in Architectural Acoustics, 110 8th Street, Troy, New York, NY 12180)

The Editor-In-Chief of the Journal of the Acoustical Society of America (JASA) has recently compiled a list of problems in the manuscripts submitted to JASA that often lead to an outright rejection by the Associate Editors handling their review. These problems often occur during submission and writing of a manuscript. They include selecting a title, listing the authors, composing an abstract, defining the scope of work, presenting the background and significance of the research, reporting and discussing the major discovery, ideas and results, showing the work and results that have been published by others in other journals already, drawing concise conclusions, citing references, displaying equations, figures, tables, etc. Last but not the least is the English writing that should be grammatically correct and easy to understand by someone with a similar background. This talk gives a quick overview of these potential problems, which are frequently shown in the manuscripts submitted by authors overseas. The goal of this talk is to provide helpful suggestions and guidelines to the prospective authors whose native language is not English to submit manuscripts that can pass the initial screening and ultimately get published in JASA. Disguised examples of some problematic manuscripts are discussed and analyzed.

11:40

4aID7. Discussions on publishing excellence in the Journal of the Acoustical Society of America and JASA Express Letters. Ning Xiang (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, Troy, New York, xiangn@rpi.edu), and Li Cheng (Department of Mechanical Engineering, The Hong Kong Polytechnic University, Hung Hom, Hong Kong)

This workshop involves a number of invited speakers from the editorial board members of the Journal of the Acoustical Society of America (JASA, including JASA Express Letters) and from representative JASA paper authors to discuss the JASA peer-review process, the criteria for accepting manuscripts, and successful preparation of manuscripts for JASA publications. Following the presentations by the invited speakers, a panel discussion offers a platform for those in the audience, particularly potential authors, to ask relevant questions and for the panelists to give multiple replies. The panelists also include a liaison from the American Institute of Physics to the Acoustical Society of America for all publishing affairs. [The panelists: Whitlow Au, Li Cheng, Robert Harington, Allan Pierce, S.K. Tang, Sean Wu Ning Xiang, Zhaoyan Zhang]

Session 4aMU

Musical Acoustics and Psychological and Physiological Acoustics: Musical Timbre: Perception and Analysis/Synthesis I

James W. Beauchamp, Cochair
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Andrew B. Horner, Cochair
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Invited Papers

9:20

4aMU1. Real-time segmentation of the temporal evolution of musical sounds. John Glover, Victor Lazzarini, and Joseph Timoney (National University of Ireland, Maynooth, john.c.glover@nuim.ie)

Since the studies of Helmholtz, it has been known that the temporal evolution of musical sounds plays an important role in our perception of timbre. The accurate temporal segmentation of musical sounds into regions with distinct characteristics is therefore of interest to the study of timbre perception as well as to different forms of sound modelling and manipulation. Following on from recent work by Peeters and Caetano et al, this paper presents a new method for the automatic segmentation of the temporal evolution of isolated musical sounds in real-time. We define attack, sustain and release segments using cues from a combination of the amplitude envelope, the spectro-temporal evolution and a measurement of the stability of the sound that is derived from the onset detection function. We conclude with an evaluation and discussion of some potential applications of the method.

9:40

4aMU2. Impact of MP3-compression on timbre space of sustained musical instrument tones. Chung Lee, Andrew Horner (Hong Kong University of Science and Technology, im.lee.chung@gmail.com), and James Beauchamp (University of Illinois at Urbana-Champaign)

MP3 compression is widely used in music sharing and storage. A number of studies have investigated the discrimination of instrument tones after MP3 compression. Additionally, a number of previous studies have evaluated other data reduction methods including frequency modulation (FM) synthesis, wavetable synthesis, and principal component analysis (PCA). However, these studies have not considered the impact on timbre space after data reduction. In this study, listening test subjects were asked to rate the dissimilarity of all pairs of original instrument tones. The same process was done on MP3-compressed tones with various bit-rates. Correlation analysis was done on the dissimilarity data of the original and compressed tones to see if MP3 compression caused a significant impact on the perceptual distance between instrument pairs. The multidimensional scaling (MDS) solutions of the original and compressed tones were also compared to see if the timbre space was significantly altered after MP3 compression (e.g., would a clarinet sound more or less similar to an oboe after MP3 compression?) [This work was supported by RGC grants 613510 and 613111.]

10:00

4aMU3. Investigation of timbre saliency, the attention-capturing quality of timbre. Song Hui Chon and Stephen McAdams (CIRMMT, Schulich School of Music, McGill University, 555 Sherbrooke Street West, Montreal QC, Canada H3A 1E3, songhui.chon@mail.mcgill.ca)

Timbre saliency is defined as the attention-capturing quality of timbre. Saliency differences between timbres were measured using a tapping technique in which the stronger beat in ABAB isochronous sequences was reproduced by the listener, the idea being that the more salient timbre would capture listeners' attention and be chosen more often as the strong beat. A timbre saliency space was defined in which the distance between a pair of timbres corresponded to the difference in timbre saliency. Stimuli were generated with 15 orchestral instruments, equalized in pitch, loudness and duration. Data from 40 participants yielded a one-dimensional CLASCAL solution with two latent classes and specificities. Latent class structure shows no relation with gender, musicianship or age. Testing audio descriptors from the Timbre Toolbox [Peeters et al., 2011, J. Acoust. Soc. Am., 130, 2902-2916], the odd-even harmonic energy ratio explains 51% of the variance along this dimension. A combination of tristimulus (band 3) and odd-even ratio explains 73% of the variance in the mean saliencies of individual sounds across all other comparison sounds. Mean saliency thus seems to depend on the high-frequency harmonic energy and spectral envelope jaggedness, whereas saliency comparisons between timbres depend more on spectral envelope jaggedness.

10:20

4aMU4. Toward an effective use of timbre in data sonification. Hiroko Terasawa (University of Tsukuba/JST-PRESTO, terasawa@tara.tsukuba.ac.jp)

The spectro-temporal structure of a sound determines its timbre, and carries musically interesting information such as instrument type and performance expressions. Using timbre in data sonification can be viewed as an inverse transform of this process: Expressing data with timbre is equivalent to designing the spectro-temporal structure of a sound. Taking that into account, timbre is most effectively used in sonification by projecting time-series data onto the spectro-temporal structure of a sound. The temporal structure of the data often differs from the archetypal spectro-temporal structure of traditional instrumental sounds. But this discrepancy contributes to novel musical expression, based on novel timbre design. From this perspective, some sonification works are presented, such as ones produced by sonification of dynamic motion of genetically-modified worms and dynamic transitions of brain-wave data. Based on these examples, methods for effective and expressive use of timbre in data sonification will be presented.

10:40–11:00 Break

11:00

4aMU5. Wind instrument sound design with centroid-controlled spectral template synthesis. Simon Wun, Andrew Horner (HKUST, simonwun@ust.hk), and James Beauchamp (UIUC)

Most previous sound synthesis research has been oriented to instrument imitation and data reduction, whereas perceptual control has received little attention. In other words, the parameters in traditional sound synthesis are not perceptually meaningful. Using common synthesis techniques such as multiple wavetable synthesis and additive synthesis, we have no obvious way to imitate acoustic tones while allowing perceptual control. An important perceptual parameter is spectral centroid, which strongly correlates with a sound's brightness. Spectral centroid and attack time are two universally-recognized timbral features that strongly influence discrimination and identification of musical instruments. This paper investigates the generality and effectiveness of spectral template synthesis, a perceptually-based technique for synthesizing wind instrument tones. Synthesis from spectral envelope templates is driven by spectral centroid or other control functions. Control by spectral centroid has the advantage of direct manipulation of a perceptually-salient feature. Unlike other synthesis techniques, spectral template synthesis is designed to track changes in spectral centroid and mimic acoustic tones at the same time. This work has application in the synthesis of natural realistic sounds that go beyond the normal timbral boundaries of acoustic instruments. [This work was supported by RGC grants 613510 and 613111.]

11:20

4aMU6. Relating timbre discrimination to perceptual distances between interpolated percussive timbres. William L. Martens and Mark McKinnon-Bassett (Faculty of Architecture, Design and Planning, University of Sydney, NSW 2006, william.martens@sydney.edu.au)

A set of percussive timbres was generated using a hybrid resynthesis that was based upon the analysis of recorded conga and bongo drums. Comprising the set were drum timbres resulting from parametric variation in both damping of the low-frequency resonance associated with pitch of the drum, and variation in a higher-frequency resonance associated with percussive attack transients. Listeners were presented with all pairwise comparisons of the synthetic drum sounds, and were asked first to perform timbral discriminations for each pair, and subsequently to produce pairwise dissimilarity judgments. Underlying perceptual scales values were derived for each timbre from discrimination performance along the two manipulated stimulus dimensions, and these values predicted well the perceptual distances that were fit to the stimulus space coordinates derived from the dissimilarity judgments. Taken together, the results provide a basis for developing a reliable control structure for the synthesis of such percussive timbres.

Contributed Papers

11:40

4aMU7. Individual differences in the relative salience of percussive timbre dimensions. Mark McKinnon-Bassett and William L. Martens (Faculty of Architecture, Design and Planning, The University of Sydney, NSW 2006, mbas4365@uni.sydney.edu.au)

A set of nine percussive timbres was generated by varying parameters of two resonant filters incorporated in a hybrid resynthesis of recorded drum sounds. Three values of damping for a lower-frequency resonance were factorially combined with three center-frequency values for a higher-frequency resonance associated with percussive attack transients. Two groups of listeners were asked to produce dissimilarity judgments for all pairwise comparisons of the nine sounds on a ten-point scale. The dissimilarity judgment data from two groups of subjects were combined to form a single dataset for submission to Individual Differences Scaling (INDSCAL) analysis. A common timbre space of just two dimensions was derived along with a subject space that revealed the different weights placed by each subject on each of those dimensions of the derived timbre space. Individual differences in the relative salience of these percussive timbre dimensions were related to the musical training of the listeners.

12:00

4aMU8. A new sinusoidal model for synthesis of musical instruments. Sudhendu Raj Sharma (Purdue University, School of Electrical and Computer Engineering, Electrical Engineering Building, 465 Northwestern Ave., Mailbox 429, West Lafayette, IN 47907, sharmasr@purdue.edu), Zhenhao Ge, and Mark J. T. Smith (Purdue University, School of Electrical and Computer Engineering, Electrical Engineering Building, 465 Northwestern Ave., West Lafayette, IN 47907)

Many algorithms have been developed over the years to synthesize acoustic sounds and are now used commercially in acoustic synthesizers and digital keyboard products. The issue with these algorithms is the trade-off among sound fidelity, algorithm complexity, and hardware/storage requirements-the last two of which are directly related to the cost of the system. In this reported work, we introduce a new sinusoidal model for digitally synthesizing musical instruments. The new algorithm has unusually high fidelity, minimal memory requirements, and high computational efficiency. The algorithm is based on the "analysis-by-synthesis overlap add" (ABS/OLA) sinusoidal model, which models musical sounds as a short-time weighted sum of constant frequencies, phases, and amplitudes. The new model we introduce incorporates a novel dynamic pitch and frequency control feature in synthesis that allows very high quality instrument sounds to be generated over a wide range of pitches from a very short sampled recording of the musical instrument. Sound modifications can be performed parametrically within the framework all using fast Fourier transforms (FFTs) for high efficiency. Examples of synthetically generated non-western musical instruments will be presented during the conference and contrasted with competing technologies to illustrate the advantages of the new method.

Session 4aNSa

Noise, Engineering Acoustics, Physical Acoustics, and Signal Processing in Acoustics: Active Noise Control II

Siu-Kit Lau, Cochair
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Invited Papers

9:20

4aNSa1. Challenges in the implementation of active noise control technologies. Xiaojun Qiu and Ningrong Li (Institute of Acoustics, Nanjing University, xjqiu@nju.edu.cn)

A number of projects have been carried out in Nanjing University to implement active noise control technologies, which include active control of transformer noise, active sound barrier, active noise control in communication chassis, active noise control in natural ventilation windows, active control of large impulsive noise in headset and active noise control in a train compartment. Unlike our previous research, these projects are all funded by industries and the aim is not for doing academic research, but to make commercial prototypes. Various challenges in the implementation of active noise control technologies in these projects to make commercial products are reported and discussed, and main issues to make successful commercial active noise control products are pointed out.

9:40

4aNSa2. On family of fractional lower order moment (FLOM)-based algorithms for active noise control of impulsive noise sources. Muhammad Akhtar (The Center for Frontier Science and Engineering (CFSE), The University for Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan, akhtar@ice.uec.ac.jp), and Wataru Mitsuhashi (Department of Communication Engineering and Informatics, The University for Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan)

Active noise control (ANC) is based on the principle of destructive interference of propagating acoustic waves; essentially a cancelling signal is generated and combined with the primary noise to achieve acoustic cancellation around location of the error microphone. In this paper, we consider a very challenging application of ANC for impulsive noise. The impulsive noise can be modeled using non-Gaussian stable process for which second order moment does not exist. The most famous filtered-x-LMS (FxLMS) algorithm for ANC systems, is based on minimization of the variance of the error signal, and therefore, becomes unstable for the impulsive noise. It has been shown that filtered-x least mean p-power (FxLMP) algorithm; based on minimizing the fractional lower order moment (FLOM) that does exist for stable distributions; gives robust performance for impulsive ANC. However, the convergence speed of the FxLMP algorithm is very slow. Recently; we have proposed various variants of FxLMP algorithm, so that an improved convergence and noise reduction performance is achieved. In this paper, we propose modifying and employing generalized normalized LMP algorithm (GNLMP) algorithm for ANC of impulsive noise. The computational complexity of proposed algorithm is comparable to the existing FLOM-based ANC algorithms. Extensive simulations are carried out, which demonstrate the effectiveness of proposed algorithm. We observe that, in comparison with the existing FLOM-based ANC algorithms, the proposed algorithm gives best performance for ANC of impulsive noise sources.

Contributed Papers

10:00

4aNSa3. Performance of active noise barrier with a moving sound source. Jiancheng Tao, Yiqing Deng, and Xiaojun Qiu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, jctao@nju.edu.cn)

Active noise control is an effective technology to improve noise reduction performance of traditional passive noise barriers at low-frequencies. Previous researches on active noise barriers are mainly on the assumption that the primary sound source is stationary. Effects of source motion on the performance of an active noise barrier are investigated in this paper. First, an analytical model of a passive barrier with a moving sound source is introduced, which can be used to calculate the primary sound field distribution in time domain. Then, the performance of applying active noise control on such a barrier is investigated numerically. Finally, the experimental results with a practical prototype of active noise barrier will be reported and be compared with the numerical results.

10:20

4aNSa4. Active control of exhaust noise using an air horn. Dongki Min, Deokman Kim, and Junhong Park (Hanyang University, 133-791, dkmin@hanyang.ac.kr)

The noise generated by internal combustion engine is reduced by passive muffling systems. For the passive muffling system, its performance significantly degrades especially when multiple low frequency tonal components exist in the flow. The tonal components occur from explosion process of the engine, and radiates as a monopole from the outlet. In this study, active noise cancelation using FxLMS algorithm is proposed to reduce the exhaust engine noise. Air-horn which is capable of being operated at high temperatures is proposed for cancelation of the radiated noise. The vibration input the diaphragm of the air-horn allowed the active control of the frequency and phase of the radiated sound. The FxLMS algorithm was used to actively control the sound radiation from the air-horn to achieve cancelation of the noise from the muffler. The sound radiation from the air-horn induces dipole-like noise radiation from the exhaust system, and significantly reduced the radiated sound power.

10:40–11:00 Break

11:00

4aNSa5. Research on decentralized adaptive active control for a single-layer vibration isolation system. Fengyan An, Hongling Sun, Xiaodong Li, and Jing Tian (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, anfy@mail.ioa.ac.cn)

In this paper, an investigation on decentralized adaptive active control for a single-layer vibration isolation system is introduced. The vibration isolation system consists of a honeycomb table as a vibration source, four rubber isolators in parallel with electromagnetic actuators, and another honeycomb table connected with a rigid base by four rubber isolators as a flexible base. Both simulations and experiments show that the decentralized controller exhibits good performances at frequencies where the tables could be dealt with as rigid bodies. At higher frequencies, however, the system could not work stably because flexible vibrations of the tables become dominant. An experimentally validated optimization method for internal parameters of the decentralized control algorithm is proposed to improve the stability and convergence of the control system.

11:20

4aNSa6. The design of a multi-channel active noise controller with ultra low latency. Kai Chen, Jing Lu, and Xiaojun Qiu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, chenkai@nju.edu.cn)

For wide band noise control, the performance of active noise control systems depends largely on the latency of the controller. The latency of the controller is usually caused by the AD and DA converter, and the normally used Delta-Sigma audio codec is not suitable for a real time active noise control system because several milliseconds time delay of the codec will lead to non-causality of the whole control system. In this paper, an ultra low latency multi-channel audio input and output system is described. The data of the system can be interacted with a float-point DSP, where a high efficient multi-channel feed-forward control algorithm is embedded. The proposed controller also includes an ARM processor, which is in charge of the friendly user interface.

11:40

4aNSa7. Novel application of PVDF film in active noise control through windows. Jeremy Lane (NZi3, University of Canterbury, 69 Creyke Rd, Ilam, Christchurch 8041, jeremy.lane@pg.canterbury.ac.nz), John Pearse, and Stefanie Gutschmidt

PVDF film has been used widely used in active control solutions for noise and vibration. In this work, due to the transparent property of PVDF film and the proven possibility of transparent electrodes, the feasibility of PVDF film's use in the construction of a second source for active noise control (ANC) through windows is considered. Sound pressure level measurements are described to establish the feasibility of PVDF in this application. Different configurations, using glass and acrylic glass backing materials of varying thicknesses, of PVDF film speaker are reported and compared. Finally comments are made on relative performance and the overall likelihood of use in an ANC application for windows.

12:00

4aNSa8. A broadband active control algorithm without cancellation path modeling. Min Gao and Xiaojun Qiu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, gaomin1221328@163.com)

Recently, an active control algorithm without cancellation path modeling has been investigated, which does not require identification of the cancellation path. The algorithm adopts the standard LMS to update the adaptive filter coefficients, but unlike the FXLMS algorithm, the reference signal does not need to pass through the cancellation path model and the proper update direction of the adaptive filter coefficients is chosen automatically by monitoring the excess noise power. Simulation and experimental results show that the algorithm works well to sinusoidal noise, multi-tune noise and narrowband noise within 40 Hz bandwidth, with similar noise reduction performance to that of the FXLMS algorithm. However, it is found that the algorithm does not work well for broadband noise and wider narrowband noise. Aiming at this, this paper investigates the mechanism of the algorithm for broadband noise and explores potential solution to the problem.

12:20

4aNSa9. An investigation on passive-active absorption system in a water-filled impedance tube. Xiaolin Wang, Bilong Liu, and Jing Tian (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, wangxiaolin@mail.ioa.ac.cn)

A hybrid passive-active sound absorption structure is experimentally investigated in a water-filled impedance tube. The surface impedance of the passive material is manipulated by active parts to match the impedance of medium, and consequently to improve sound absorption performance at low-frequency range. Parameters such as boundary, material thickness and position of secondary source are optimized by numerical analysis to achieve high absorption coefficient. Experimental results show that the developed hybrid structure has potential to improve sound absorption at low frequency range.

Session 4aNSb

Noise, Architectural Acoustics, Animal Bioacoustics, and ASA Committee on Standards: Soundscape and Its Application I

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Invited Papers

9:20

4aNSb1. Why soundscape? The new approach to "measure" quality of life. Brigitte Schulte-Fortkamp and Kay Voigt (Technische Universität Berlin, Germany, bschulte_f@web.de)

It is now about 15 years that Soundscape came into the field of community noise and sound quality. The Soundscape approach has provided essential knowledge for the demanding tasks which are required for the design and planning of sustainable environments to support to wellbeing, health, and quality of life, respectively. The multidimensional Soundscape approach puts emphasis on the way the acoustic environment is perceived, experienced and understood by the individual and by society (ISO/TC 43/SC 1/WG 54). Moreover, it accounts for people's concerns and integrates the exposed people as experts. The process of tuning of noise pollution or sound design with respect to the expertise of people's mind is related to the strategy of triangulation of interdisciplinary data. Moreover, the Soundscape approach provides the frame work to integrate contextual and subjective variables to improve the respective Soundscape with regard to people's expertise. This paper will highlight the process of Soundscape and its application with respect to ISO/TC43/SC 1/WG 54 and the COST network TD0804 on Soundscape and Landscape with regard to its implementation and dissemination in the diverse fields of acoustic environments and its definitive meaning concerning quality of life.

9:40

4aNSb2. Perceived soundscapes in relation to transport related annoyance, context and personal characteristics; psychometric analyses. Irene van Kamp (MGO, RIVM, P.O. Box 1, 3720 BA, irene.van.kamp@rivm.nl), Elise van Kempen, and Danny Houthuijs

Most studies into perceived soundscapes have addressed subjective soundqualities at a (very) low scale level, such as parks, recreational area's and squares. Studies into the effects of transport related noise seldom incorporate perceived soundscapes and are typically focussed on negative effects, such as annoyance, sleepdisturbance and environmental worry. It would be valuable to know how people describe their sound environment in areas with varying levels of road-, air or rail noise. Available data on perceived soundscapes from the two Schiphol surveys in 2002 and 2005 allowed us to perform such analyses. The intercorrelations were studied between dimensions of perceived soundscapes, annoyance, arousal as well as several measures at the contextual, social and psychological level, sometimes referred to as non-acouticsl factors. Results shed light on the construct validity of the perceived soundscape scale and may contribute to further refinement of this instrument.

10:00

4aNSb3. The link between soundscape perception and attention processes. Fiebig André (HEAD Acoustics GmbH, Ebertstr. 30a, 52134 Herzogenrath, Germany, andre.fiebig@head-acoustics.de)

In order to understand the perception and evaluation of soundscapes, it appears mandatory to concentrate not only on constellations of sources and their contributions to the acoustic environment, but also to consider attention processes towards sound sources. It is widely known that a listener can easily focus on a certain source and can suppress the noise of other sources, which is called cocktail-party effect. It is assumed that this effect greatly influences the general appreciation of the whole soundscape. However, the process, why people focus on certain sound sources and how this influences the overall evaluation, has to be explored. A detailed knowledge about the (often subconscious) focussing on sources in multi-source soundscapes would be very helpful for design purposes, to attract deliberately attention to certain sound sources leading to positive feelings for the majority of soundscape visitors. Laboratory results dealing with the effect of source attention and its impact on soundscape evaluation were already published. In these surveys it was found that the processes, in which way the global impression changes due to the attention attraction to certain sources, seem to be complex. The paper will focus now on in-situ assessments and will show new results gained in field experiments.

10:20

4aNSb4. From noise control to sound design: the class room as a soundscape project. Juergen Bauer (Waterford Institute of Technology, Ireland, jbauer@wit.ie)

As part of an overall campus building project, the Department of Architecture in Waterford Institute of Technology in Ireland moved to provisional premises in autumn 2011, in a city centre former warehouse, dating from 1875. While this building is a fine example of historic industrial architecture which was previously used successfully as a museum, as a school venue it is "acoustically seen" inappropriate. The studios are more halls rather than rooms and have an approx. height of 5 meters; two classes share one unit and are subdivided by screens, with lectures and tutorials needing to be scheduled at different times in order to avoid (acoustic) clashes. Most surfaces are hard, and in some cases, the class units are even exposed to open galleries and circulation areas. How can the noise problem be transformed into a soundscape project? How can the current situation be used to develop sound as a design tool that informs the awareness about sound phenomena, strengthen the understanding of sound mitigation and instill the confidence to design it? This paper investigates different approaches as to how to introduce sound as a design tool in early architectural education and summarizes the learning outcomes from using the class room as a sound design lab.

10:40–11:00 Break

11:00

4aNSb5. A case study of soundscape design based on acoustical investigation. Hui Ma and Sen Zhang (School of Architecture, Tianjin University, No. 92, Weijin Road, Nankai District, Tianjin 300072, China, mahui@tju.edu.cn)

Jinwan square is an important part of Haihe river, mother river of Tianjin, a Chinese city. In order to create a lively and comfortable environment for Jinwan square, soundscape design is necessary because the acoustical situation in this square is evaluated to be noisy and boring. Based on the sound investigation held in four seasons including sound level, sound type, the relationship between different sounds and sound expectation, a soundscape design focusing on adding natural sounds and controlling both road traffic noise and construction noise was done. Through this study, the process and the method of how to do a soundscape design in certain area were tried to be concluded.

11:20

4aNSb6. Human-machine interaction as influencing factor of indoor soundscape evaluation. Jochen Steffens (Duesseldorf University of Applied Sciences, ISAVE, Josef-Gockeln-Strasse 9, 40474 Duesseldorf, Germany, jochen.steffens@fh-duesseldorf.de), Brigitte Schulte-Fortkamp (Technische Universität Berlin, ISTA, Einsteinufer 25, 10587 Berlin, Germany), and Joerg Becker-Schweitzer (Duesseldorf University of Applied Sciences, ISAVE, Josef-Gockeln-Strasse 9, 40474 Duesseldorf, Germany)

In many ways people interact with machines like vehicles or household appliances. Thereby they consciously and unconsciously perceive information about the machines' performance and operating state. Apart from visual or haptic feedback important information is transferred via the acoustical input. In order to design well-assessed indoor soundscapes like car interiors or kitchen it is essential to involve people in early product development processes. However, in order to evoke everyday perception processes in listening studies and to ensure ecological valid sound evaluations realistic human-machine interaction has to be reconstructed. Only under quasi-real conditions the user needs acoustical feedback from the used devices. This determines the user's attitude to the sound and thus to its evaluation to a large extent. Within this contribution case studies will be presented which expound the influence of interaction on cognitive, emotional, and motivational aspects within the sound evaluation in soundscape research.

11:40

4aNSb7. Soundscape case study: acoustics, ecology and its anthropological sense. Zhiyong Deng (College of Music, Capital Normal University, No. 105, Xisanhuan North Road, Beijing 100048, China, dzy@cnu.edu.cn), Guowen Zhou, Wei Hua (College of Music, Guangxi Arts Institute, Nanning 530022, China), Yun Wang (College of Music, Capital Normal University, No. 105, Xisanhuan North Road, Beijing 100048, China), and Jian Zhang (The 3rd Research Institute of CETC, Beijing 100015, China)

Base on six soundscape investigation case studies in some small urban or historical areas, included the South Putuo Temple of Xiamen city in 2006, the downtown of Kashkar city in 2006, the local worship music performing area of South Gaoluo village in 2007, the Nanxiangkou local music performing area in Hebei University of Technology Gymnasium of ShijiaZhuang city in 2007, the downtown riverside of Liuzhou city in 2010, the local worship and music performing areas of Qianjuntai village and Zhuanghu village of Beijing city in 2011, a certain relationship analysis between the acoustical parameters and its audience's behavior or subjective assessment is put forward in this paper. Due to the two-dimensional (Leq and the subjective assessments) fuzzy clustering or curve fitting, it shows that the sound ecology would have a critical connection to the anthropological sense, which means some keynotes sound and soundmarks must be keep constantly during the soundscape design or the urban development. Furthermore, a rough concept of sound history and the constant of keynotes and soundmarks in the view of soundscape are also discussed in this paper.

12:00

4aNSb8. Soundscape variation in a historical city centre due to new traffic regulation. Luigi Maffei, Maria Di Gabriele, and Francesco Aletta (Seconda Università di Napoli- Center RiAS- Via S.Lorenzo, 81031 Aversa (Italy), luigi.maffei@unina2.it)

In recent years the number of candidate historic city centers to be included in the World Heritage List is increasing. This inclusion must be supported by a Management Plan programming all intervention to be implemented for the preservation of the "outstanding universal value". So far the Management Plans do not consider the preservation and valorization of the soundscape. Consequently, urban renewal processes are based on conservation and restoration of tangible cultural heritage, in order to increase touristic attraction and to improve the quality of life. All these efforts privilege visual perception and do not take in account the auditory perception. Soundscape of a site can be considered an intangible cultural heritage to be preserved and valorized as it constitutes a peculiar characteristic of the place. It makes the place recognizable and attractive. Recently the historic centre of Naples (Italy), as World Heritage Site, has been under renewal and for sustainable mobility the largest restricted traffic area (ZTL) in Europe has been introduced. The results of soundwalks carried out in the historic center of Naples before and after the implementation of ZTL are presented. The variations of acoustical and other environmental parameters influencing the subjective perception of environmental quality are analyzed.

12:20

4aNSb9. Study on how to create a comfortable soundscape for commercial open space. Jianwei Song and Hui Ma (School of Architecture, Tianjin University, No. 92, Weijin Rd., Nankai District, Tianjin, China, boobu530@163.com)

Commercial open space plays a vital role in urban life and the soundscape design of those areas becomes a new problem worthy of researching. In this study, the sound situation of two famous commercial open areas in Tianjin, China, including sound type, sound expectation, and environmental evaluation was analyzed through physical measurement and social surveys. Finally, ten sound samples were obtained from those commercial open areas. Except noise level, both temporal and spatial factors of the sound samples were analyzed. Combined the laboratory experiments and sound signal analysis, the principle of how to create a comfortable soundscape in commercial open space was explored from block design, architecture style and material selection.

THURSDAY MORNING, 17 MAY 2012 S223, 9:20 A.M. TO 12:20 P.M.

Session 4aPA

Physical Acoustics and Engineering Acoustics: Emerging Technologies and Concepts in Ultrasonics

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Contributed Papers

9:20

4aPA1. Ultrasonic set up for the assessment of the stability of a cylinder inserted in a solid. Vincent Mathieu (CNRS, Université Paris-Est, Laboratoire Modélisation et Simulation Multi Echelle, UMR 8208 CNRS, 61 avenue du Général de Gaulle, 94010 Créteil, France, vincent.mathieu@u-pec.fr), Fani Anagnostou, Emmanuel Soffer (CNRS, Université Paris-Diderot, Laboratoire Bioingénierie et Biomécanique Ostéo Articulaire, UMR 7052 CNRS, 10 avenue de Verdun, 75010 Paris, France), and Guillaume Haiat (CNRS, Université Paris-Est, Laboratoire Modélisation et Simulation Multi Echelle, UMR 8208 CNRS, 61 avenue du Général de Gaulle, 94010 Créteil, France)

The study aims at proposing a new experimental ultrasonic methodology for the estimation of the stability of a cylinder inserted in a solid. Such a technology may have various fields of application: aeronautics, car industry, mechanics or also surgery. The present prototype is dedicated to the study of the stability of dental implants. Cylindrical titanium implants were inserted in four groups of rabbit femurs, each group corresponding to a controlled level of stability of the cylinders. The 10 MHz ultrasonic response of the cylinder is processed to derive quantitative indicators based on the temporal variation of the signal amplitude. Analysis of variance (ANOVA) ($p < 10^{-5}$) tests revealed statistical distributions of indicators significantly correlated with the stability of the cylinders. A numerical finite-difference time-domain model was considered in order to understand the origin of the different echoes and the importance of lateral wave propagation was evidenced. The numerical model also enabled to estimate the sensitivity of the indicators to variations in the material properties of the materials in contact with the cylinders.

9:40

4aPA2. The study of dissipative nonlinearity in oil sand. Jiehui Liu, Jinlin Zhu, Xiaozhou Liu, Xiufen Gong, and Dao Zhou (Key Laboratory of Modern Acoustics, Ministry of Education, Institute of Acoustics, Nanjing University, Nanjing 210093, China, wljh@nju.edu.cn)

The acoustic waves propagating in sand have the nonlinear dissipative phenomenon that the dissipative coefficient of acoustic waves with larger amplitude is smaller than that with smaller amplitude. The results of experimental investigations on the propagation of acoustic waves in oil sand with different oil content are presented in the paper. The nonlinear dissipative phenomenon in oil sand is studied and the analytical description is given to explain the phenomenon. It is found that the relative growth coefficient and the dissipative index are dependent on the oil content in oil sand. According to the dependence relationships between the sensitive coefficients and the oil content, a new prospective approach to measure the oil content in oil sand is provided in the paper for oil exploration.

10:00

4aPA3. Cell structure in waves diffracted by a wedge. Mitsuhiro Ueda (Predio Meguro Science Laboratory, 4-20-13 Meguro, Meguro-ku, Tokyo 153-0063, Japan, ueda-mt@nifty.com)

Waves diffracted by a wedge made of perfectly reflecting material exhibit characteristic spatial pattern depending on an aperture angle of the wedge. For examples, in the wedge of aperture angle π, that is, a perfectly reflecting plane and that of aperture angle $\pi/2$, that is, a corner cube, diffracted waves are identically zero. And in the wedge of aperture angle 2π, that is, a semi-infinite plane, diffracted waves are symmetric with respect to a central axis of the wedge. The relation between the pattern and the aperture angle, however, has not been studied in detail so far since there is no appreciate model for diffracted waves and the rigorous solution for diffracted waves is so complex that only the simplest case can be analyzed. We have proposed the new mathematical model for diffracted waves where they can be expressed as a sum of two more fundamental quantities called elementary diffracted waves. The new model reveals that cell structure exists in waves diffracted by the wedge of aperture angle π multiplied by a rational number less than 2. The cases mentioned above can be explained in terms of the cell structure.

10:20

4aPA4. Measurement of the sound pressure in the focal spot area of line-focus ultrasound field by Schlieren technique. Xue-Ping Jiang, Qian Cheng, and Meng-Lu Qian (Institute of Acoustics, Tongji University, 1239 Siping Road, Shanghai 200092, China, 0720106002@tongji.edu.cn)

Schlieren method is an effective method for studying the sound field in transparent medium which called phase objects. The method is used to research the acoustic field by analyzing the refractive index changes induced

by the acoustic wave. Calculations of the sound pressure distribution radiated by the line-focus ultrasonic transducer are implemented, and the acoustic field in the focal spot area is obtained. Then the diffraction light intensity on the Fourier transform plane in a Schlieren system is calculated with two-dimensional Fourier transformation when the phase object is the ultrasonic wave. Because the light intensity of the different diffraction spots on the back focal plane of the transform lens depends on the sound pressure, the sound pressure in the focal spot area can be determined using Schlieren technique. A Schlieren system is set up. The images of the line-focus ultrasonic field are obtained and the sound pressure in its focal point area is measured non-invasively by measuring and comparing the light intensities of the different diffraction spots. This work is supported by the National Natural Science Foundation of China (No. 10804085)

10:40–11:00 Break

11:00

4aPA5. A pipe-like one-way structure of acoustic wave. Bo Yuan (Key Laboratory of Modern Acoustics, MOE, and Institute of Acoustics, Department of Physics, Nanjing University, Nanjing 210093, China, xillo.yuan@gmail.com)

We proposed a pipe-like structure with linear materials to obtain the acoustic unidirectional transmission. The system consists of a bending pipe and a phononic crystal and designed by extending the idea of frequency selection in nonlinear acoustic diode into the mode selection of a linear acoustic system. The system has a significant transmission efficiency and good rectifying ratio which is designed to work in the air. We also experimentally realized the unidirectional transmission behavior for acoustic waves and the experimental results agree well with the theoretical simulation. This device is expected to have potential applications in ultrasonic devices such as acoustic diodes.

11:20

4aPA6. The surface acoustic waves in the phononic crystal surface based on the locally resonance mechanism. Yong Li (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, kyolee2010@gmail.com)

We proposed a two dimensional structure consisted of a resonant unit stubbed periodically on a phononic crystal plate with finite thickness to investigate the surface acoustic waves (SAWs). Numerical results shown that two types of SAWs, one is the classical Rayleigh SAWs, whereas the other is the scattering SAWs overcoming the limitation that SAWs can only exit below the bulk cone of the substrate, can be found in the structure. Furthermore, the band gaps of the SAWs are obtained, as well as slow modes of the SAWs are also observed in the band gaps of the phononic crystal substrate. It could open the probability to control effectively the propagation of these SAWs. The results should have the impact on the SAWs communications.

11:40

4aPA7. Utilizing negative diffraction effect in phononic crystal to realize resting X-shaped wave localization. Weiwei Kan and Jianchun Cheng (Department of Physics, Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, P.R. China, rdchkww@gmail.com)

Motivated by the promising impacts in acoustic devices and applications that can efficiently trap, guide, and manipulate sonic waves, researches of Photonic Crystals (PCs) have received great attention in the past decade. Many anomalous refractive and diffractive effects, such as superprism effects and negative refraction, have been found in such PCs. In this report, by studying the wave packet superposed by Bloch modes in the isofrequency surface at unique points of the PC band structure, the negative diffraction effect in certain directions of the PCs is exploited to realize X-shaped wave localization settled in one point, which cannot exist at rest in homogeneous media.

12:00

4aPA8. Ultrasonic cleaning of the root canal. Bram Verhaagen, Christos Boutsioukis (Physics of Fluids, University of Twente, P.O. Box 217, 7500AE Enschede, The Netherlands, b.verhaagen@tnw.utwente.nl), Lei-Meng Jiang, Ricardo Macedo (Academic Center for Dentistry, Gustav Mahlerlaan 3004, 1081LA Amsterdam, The Netherlands), Damien Walmsley (School of Dentistry, University of Birmingham, St Chad's Queensway, Birmingham B4 6NN, United Kingdom), Luc Van der Sluis (Paul Sabatier University, 118 route de Narbonne, 31062 Toulouse cedex 9, France), and Michel Versluis (Physics of Fluids, University of Twente, P.O. Box 217, 7500AE Enschede, The Netherlands)

A crucial step during a root canal treatment is the irrigation, where an antimicrobial fluid is injected into the root canal to eradicate all bacteria from the root canal system. Agitation of the fluid using a miniature file oscillating at 30 kHz has shown a significant improvement in the cleaning efficacy over conventional syringe irrigation. However, the exact cleaning mechanisms, being acoustic streaming, cavitation or an enhanced chemical effect, are not fully understood. Here we investigate ultrasonically activated irrigation through experiments and numerical simulations in order to understand the relative importance of each of the three cleaning mechanisms. We combine high-speed imaging and micro-Particle Imaging Velocimetry to visualize the flow pattern and cavitation in a root canal model (sub-millimeter dimensions), at timescales relevant to the cleaning processes (microseconds). Measurements of the acoustic streaming are coupled to the oscillation characteristics of the file as simulated numerically and measured with a laser vibrometer. Comparison between the streaming pattern inside the root canal and in the free field shows the importance of the confinement of the root canal on the acoustic streaming. The results give new insight into the role of acoustic streaming for the cleaning of root canals.

Session 4aPP

Psychological and Physiological Acoustics: Current Issues in Auditory Cortex Physiology

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Invited Papers

9:20

4aPP1. Cortical representations of pitch: theories and experiments. Xiaoqin Wang (Department of Biomedical Engineering, Johns Hopkins University, 720 Rutland Avenue, Traylor 410, Baltimore, MD 21205, xiaoqin.wang@jhu.edu)

Pitch perception is one of the most important auditory perceptual phenomena. Its underlying neural mechanisms have not been well understood. Recent human imaging studies and neurophysiology experiments in non-human primate have begun to reveal possible neural coding mechanisms in the cerebral cortex. These studies have pointed to a specialized area in the rostral region of primate auditory cortex where harmonic pitch is extracted. How pitch-selective neurons in this cortical area extract harmonic pitch at the cellular level, however, is yet known. Moreover, it remained to be explored whether other auditory cortical areas process aspects of pitch that are not processed by this rostral pitch-region. An important issue in the study of cortical representations of pitch is whether pitch embedded in harmonic complex sounds is extracted and uniquely represented by a specific cortical area or a subset of neurons in that area. Simply showing that pitch information exists in neural firing in a cortical area is not an adequate demonstration of pitch processing mechanisms. (Research supported by NIH grant R01-DC003180)

9:40

4aPP2. Acoustic motion processing in auditory cortex. Stephen Lomber (The Brain and Mind Institute, The University of Western Ontario, London, Ontario, Canada, steve.lomber@uwo.ca)

Within extrastriate visual cortex of humans, monkeys and cats, individual cortical areas are specialized for spatial or motion processing. In cat auditory cortex, four regions have been identified to be critical for accurately determining the spatial location of an acoustic stimulus. The purpose of the present investigation was to determine if there is an area in auditory cortex specialized for acoustic motion processing or if areas involved in spatial localization are also critical for acoustic motion processing. Or, is there an acoustic MT? To accomplish this, cats were trained to perform two tasks: a spatial localization task using a static stimulus and a task that required the animals to discriminate leftward from rightward apparent acoustic motion. Focal reversible cooling was used to bilaterally deactivate each of the thirteen areas of cat auditory cortex. Overall, the results show that areas involved in acoustic motion processing are also involved in static spatial localization. An area that is uniquely involved in acoustic motion processing was not identified. These results suggest that spatial localization functions may be a prerequisite for acoustic motion processing in auditory cortex. Supported by the Canadian Institutes of Health Research and the Natural Sciences and Engineering Research Council of Canada.

10:00

4aPP3. Rapid plasticity in auditory and prefrontal cortex during active listening. Shihab Shamma (A.V. Williams Bldg, University of Maryland, College Park, MD 20742, sas@umd.edu)

Humans and other animals often attend to sounds in their environment so as to approach mates and competitors, or to avoid predators. Numerous neural processes orchestrate the performance of these behavioral tasks, including sensory adaptive responses in the auditory cortex and executive control functions in the prefrontal cortex. The multitude of mechanisms observed to be involved in neurophysiological recordings from several auditory and prefrontal cortical fields in behaving ferrets will be reviewed. The findings reveal that rapid changes in auditory receptive fields take place only during task performance, and that these serve to enhance discrimination and detection of target stimuli from their backgrounds. Interestingly, the changes also depend on the meaning of the sounds (aversive or appetitive) and the level of behavioral performance.

10:20

4aPP4. Micro-organization and plasticity of the primary auditory cortex. Patrick Kanold (University of Maryland, Dept. of Biology, 1116 Biosciences Bldg, College Park MD 20742, pkanold@umd.edu)

The auditory cortex is a laminated structure that adaptively processes sensory information from the external environment. The precise nature of the transformation of sensory information at the level of cortical networks is unknown. We use in vivo two-photon calcium imaging techniques to measure response properties and functional organization of primary auditory cortex (A1) neurons in mouse. We find that frequency selectivity in supragranular layers is heterogeneous on small spatial scales and that this heterogeneity is likely created from sampling of diverse inputs to supragranular neurons. The large frequency range of inputs available to each neuron might provide a

substrate for a large degree of plasticity in individual neurons. We tested the capacity of A1 neurons to rapidly change tuning properties by using micro-stimulation of top-down projections to A1 and pairing such stimulation with a particular sound. We find that the frequency tuning of individual neurons can rapidly be changed leading to an increase in the representation of the paired sound. Collectively, these results provide insight into how sensory information is represented and adaptively transformed in auditory cortex.

10:40–11:00 Break

11:00

4aPP5. Auditory cortical plasticity—from synapse to perception. Christoph Schreiner (UCSF, 513 Parnassus Ave., San Francisco, CA 94044, chris@phy.ucsf.edu), and Robert Froemke (NYU, 540 First Ave., New York, NY 10016)

Synapses and neuronal receptive fields of the cerebral cortex are plastic. Enhancements and decreases to auditory cortical excitatory synapses can be induced by pairing acoustic stimuli with activation of the nucleus basalis neuromodulatory system. Similarly, the feature selectivity of individual neurons and cell assemblies can be modified in a manner that depends on the patterns of network activity, the engagement of neuromodulatory systems, as well as by sensory experience. Perceptual performance has been shown to change as a consequence of experience and learning. The relationship between synaptic and receptive field changes and perceptual plasticity, however, is poorly understood. We used in vivo whole–cell recordings and behavioral testing to explore that relationship in more detail. We will discuss how synaptic modifications and receptive field changes are reflected in the encoding of frequency and intensity information in rat auditory cortex and demonstrate that these changes are expressed in the perceptual behavior of animals in sensory detection and classification tasks. It is concluded that direct modification of specific cortical inputs leads to wide-scale synaptic changes, which collectively support improved sensory perception and enhanced behavioral performance. Supported by NIH Grants DC02260 (to CES) and DC009635 (to RCF)

11:20

4aPP6. State-dependent changes in background discharge of auditory core neurons in freely moving guinea pigs. Hisayuki Ojima and Masato Taira (Tokyo Medical and Dental University, 113-8549, Japan, yojima.cnb@tmd.ac.jp)

In natural situations, sounds, such as predator noises and nursing and mating calls, are used as signals in determining an animal's behavior. Shifts in behavior are adaptively controlled by activation of higher-order brain regions. Utilizing forced change in behavior as a trigger, we show that brain state shifts affect the background discharge pattern of neurons in primary auditory cortex (A1). Single unit recordings were made from freely moving guinea pigs, which were impelled to shift from a passive/stationary state to an active/exploratory state by a sudden change in ambient illumination from light to dark. Upon the illumination shift, background discharge was reduced significantly for several minutes. This behavioral shift was induced even when the animals were actively engaged in eating food. Acoustic stimulation during the time of reduced background discharge resulted in an improved S/N ratio of the neuron's response. These neurons were localized almost exclusively in upper layer 5, from which cortical feedback to subcortical stations originate. In naturalistic situations, a subset of A1 neurons may transfer information about the cortical state to subcortical auditory stations. Supported by KAKENHI to H.O. (No. 22500368).

Contributed Papers

11:40

4aPP7. Dynamic binaural-correlation processing in rats' inferior colliculus, medial geniculate body, and primary auditory cortex. Qian Wang (Department of Psychology, Department of Machine Intelligence, Speech and Hearing Research Center, Key Laboratory on Machine Perception (Ministry of Education), Peking University, Beijing, China, aleinwangba@126.com), Shuyang Cao, Jingyu Li, Xihong Wu (Speech and Hearing Research Center, Key Laboratory on Machine Perception (Ministry of Education), Peking University, Beijing, China), and Liang Li (Department of Psychology, Department of Machine Intelligence, Speech and Hearing Research Center, Key Laboratory on Machine Perception (Ministry of Education), Peking University, Beijing, China)

Interaural correlation processing is critical for grouping and segregating auditory streams under reverberant environments with multiple sources. Although detection of dynamic changes in interaural correlation has been extensively studied in the field of psychoacoustics, the underlying neural mechanism remains largely unknown. In this study, frequency-following responses (FFRs) to narrow-band noises were measured at various levels of the auditory system in rats, including the inferior colliculus (IC), ventral division of the medial geniculate body (MGB), and primary auditory cortex (A1). The results of Experiment 1 show that FFRs recorded in the IC were affected by both interaural correlation and the interaural time difference (ITD). Moreover, results of Experiment 2 show that a break in interaural correlation (BIC) could elicit marked FFRs in each of the three central auditory structures, and the BIC-induced FFRs were significantly affected by the ITD. The results suggest that the rat's central auditory system is able to resolve and compare fast changes in fine-structure details of arbitrary noises presented at the two ears.

12:00

4aPP8. Associative visuoauditory memory in the auditory cortex. Jufang He, Xi Chen, Yiping Guo, Zhengli Liao, Xinjian Li, Haitao Wang, and Xiao Li (Laboratory of Applied Neuroscience, Department of Rehabilitation Sciences, The Hong Kong Polytechnic University, Hung Hom, Kowloon, Hong Kong, rsjufang@polyu.edu.hk)

This paper presents direct evidences of the establishment of associative memory traces in the rat auditory cortex and the participation of the entorhinal cortex in the establishment and retrieval of these memory traces. We produced an association between cortical electrical activation and a visual stimulus with classical conditioning. The memory traces were physiologically visualized from auditory neuronal responses to the visual stimulus after conditioning and behaviorally confirmed with a memory recall experiment. Formation of new associative memory in the auditory cortex with classical conditioning was bilaterally blocked when the entorhinal cortex was unilaterally temporarily inactivated, but returned if the entorhinal cortex was not inactivated. Retrieval of the established associative memory in the ipsilateral neocortex was affected by the inactivation of the unilateral entorhinal cortex, while that in the contralateral neocortex was not affected, thus suggesting a less dependence of the hippocampal system in the retrieval than in the formation of associative memory. Supported by Hong Kong Research Grant Council (PolyU 9/CRF/09)

Session 4aSP

Signal Processing in Acoustics, Acoustical Oceanography, and Underwater Acoustics: Model-Based Processing and Analysis III (Poster Session)

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Contributed Papers

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

4aSP1. An analysis of influencing factors for structural damage imaging using Lamb waves. Haiyan Zhang (149 Yanchang Road, School of Communication and Information Engineering, Shanghai University, Shanghai 200072, hyzh@shu.edu.cn)

The use of Lamb waves for the inspection of large plate-like structures and their structural health monitoring (SHM) has been a topic of considerable interest in the development of advanced quantitative nondestructive evaluation techniques. Current interrogation algorithm using Lamb waves can roughly identify the location and severity of potential damages. It is well documented that Lamb waves may have multiple modes at a certain frequency, and accompanied by possible dispersion for each specific mode at different frequencies. In order to improve the accuracy of damage identification, some factors such as frequency, mode and cycle number of excitation signal must be taken into account. This paper presents a probability-based imaging approach for evaluating through-thickness hole damage in an aluminium plate. This method predicts the damage in terms of the probability of its occurrence at a certain spatial position of the structure. The influences of various factors on imaging results are analyzed. The results demonstrate that the imaging algorithm can be used to identify the location and severity of damage, but the identifying accuracy is highly related to frequency, mode and cycle number of exciting Lamb wave signal.

4aSP2. Reduction of computation time using Graphics Processing Unit for the detection of a crack in a large scale concrete structure. Yuhei Katsurakawa, Toyota Fujioka, Yoshifumi Nagata, and Masato Abe (Iwate University 020-8551, h19j12@cis.iwate-u.ac.jp)

This paper describe a method for estimating a crack position in a concrete structure using several accelerometers. An array of accelerometers is installed to the concrete structure and a low frequency vibration is made with a small impulse hammer. A reflection wave is generated from the crack position if a crack exists. Because the concrete structure is elastic, it has three wave-propagation modes. It is difficult to estimate the position precisely because the power of necessary primary-wave mode is weaker than that of surface-wave mode. To estimate the crack position precisely, we have proposed a method for eliminating the unwanted surface-wave and side-wall reflections, in which five parameters are used to estimate an unwanted surface-wave or a side-wall reflection by least mean square technique. Since it takes, however, a very long time to estimate a single unwanted wave, the method did not work if two waves overlap with each other. Therefore, we propose the parallelization of genetic algorithm on GPU using CUDA. As a result, the processing time was shortened dramatically compared to conventional one, and we could distinguish two waves reflected from two close boundaries of a caisson, a huge concrete structure which is used as a breakwater.

4aSP3. Nearly Perfectly Matched Layer (NPML) absorbing boundary condition for elastic waves propagation in solid. YiFeng Li (No. 30, Puzhu Road(S), Nanjing 211816, China, 79 Box number, lyffz4637@163.com), Olivier Bou Matar, and YaPing Bao

In this work, a method named Nearly Perfectly Matched Layer (NPML) using a Complex Frequency Shift (CFS) stretched-coordinate metrics is presented to extend the Perfectly Matched Layer (PML) to simulate elastic wave propagation in solid media. This non-physical layer is used at the computational edge of a Discontinuous Galerkin Finite Element Method (DG-FEM) algorithm and a Pseudo-Spectral (PS) algorithm in time domain, as an Absorbing Boundary Condition (ABC) to truncate unbounded media. The main advantages of NPML is linked to the facts that (a) the obtained system of equations has the same form exactly as the original system of equations and so strongly hyperbolic, and (b) the introduced NPML variables are updated by Ordinary Differential Equations (ODE) in place of Partial Differential Equations (PDE) in classical PML implementation. Numerical results show that the NPML has the same ability of energy absorption as the Convolutional Perfectly Matched Layer (CPML) for attenuating the outgoing waves, moreover, it facilitates implementation in the DG-FEM scheme than CPML and preserves the highly parallelisable capabilities of this numerical scheme.

4aSP4. The study of time-frequency analysis the nocturnal snoring signal based on the wavelet transform. Zhang Yinhong (College of Physics and Information Technology, Shaanxi Normal University, Xi'an, Shaanxi 710062, China, zhangyh@snnu.edu.cn), Li Quanlu (Applied Acoustics Institute, Shaanxi Normal University, Xi'an, Shaanxi 710062, China), and Wu Jing (College of Physics and Information Technology, Shaanxi Normal University, Xi'an, Shaanxi 710062, China)

This paper presents a time-frequency analysis method for non-stationary snoring signal that based on the discrete wavelet transform. The snoring is an important characters of upper airway obstruction and a typical inspiratory sound appearing during sleep. The severe snoring sound leads to the Obstructive Sleep Apnea Syndrome during persistent ventilatory movements and it can result in cessation of breathing. The resulting experimental shows that the characters of different time-frequency domain of the snoring signal and offers value for analyzing the temporal feature of snoring sound in health medical treatment. It is an interesting application of the wavelet transform theory in the medical field.

4aSP5. The whisper sensitive scale in the application of speaker identification. Wei Lin (Nanjing University of Aeronautics and Astronautics, 211100, wlin@nuaa.edu.cn)

In this paper, the frequency characteristics of numerical whispered speech were investigated by a filter bank analysis. It was shown that the first and the third formants were more important than the other formants in the speaker identification of Chinese whispered speech. The experiment showed that the 800-1200 Hz and 2800-3200 Hz ranges were the most significant frequency ranges in discriminating the speaker. Based on this result£, a new feature scale named whisper sensitive scale (WSS) was proposed to replace the common scale, the Me1 scale£, and to extract the cepstral coefficient from whispered speech signal. Furthermore, a speaker identification system in whispered speech was presented based on WSCC (the whisper sensitive cepstral coefficient. And the new system performed better in solving the problem of speaker identification of numerical whispered speech than the traditional method.

4aSP6. A modified weighted overlap and add-based spectral subtraction method. Yang Sun, Meng Yuan, and Haihong Feng (Shanghai Acoustics Laboratory, Institute of Acoustics, Chinese Academy of Sciences, No. 456, Xiaomuqiao Road, Xuhui District, Shanghai, China, sunyang0109@mails.gucas.ac.cn)

The Weighted Overlap and Add (WOLA) method is an optimization of frequency modulated DFT filter banks. However, the decrement in frequency resolution of speech signal caused by WOLA leads to noise estimation error. In this study, a speech power compression and gain compensation method was proposed to solve the low-resolution problem from the WOLA process. Finally, a multi-band spectral subtraction method was developed by combining the WOLA and the spectral subtraction algorithm in frequency domain. The noise reduction degrees were determined by the individual Signal-to-Noise Ratios (SNRs) in each frequency bands. Objective evaluation of the proposed noise reduction method was performed based on Itakura-Saito Distortion Measure under different types of noise. Statistical results showed that this proposed method could improve noisy speech by 5 dB SNR with little distortion. This noise reduction method has an advantage on low computation load, real-time processing and good performance on noise reduction. So this method can be implemented in embedded devices, e.g. hearing aid and cochlear implants. # This work is supported by National Natural Science Foundation of China (11104316). and Shanghai Natural Science Foundation (11ZR1446000).

4aSP7. Improvement on coherent signal-subspace method using several reference frequencies. Dahang Feng (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, Beisihuan West Road, Haidian District, Beijing 100190, fengdh@mail.ioa.ac.cn), Ming Bao, Luyang Guan, Jianfei Tong, and Xiaodong Li

An improved coherent signal-subspace method is proposed using several reference frequencies for wideband direction-of-arrival (DOA) estimation. In this proposed method, the whole bandwidth of the received signal is divided into several parts, then a reference frequency is selected from each part for the coherent signal-subspace method to estimate the DOAs of wideband sources separately, and the results from all frequency parts are averaged to obtain a final estimate. Compared with the conventional coherent signal-subspace method, the proposed method achieves higher resolution and smaller root mean square error, especially when the bandwidth of the source is large. The performance of the proposed method is demonstrated and analyzed through the computer simulations.

4aSP8. The target tracking based on cubature Kalman filter. Yuanyuan Fang (Northwestern Polytechnical University School of Marine Engineering, Room 728, emma6663@hotmail.com)

A new extension of Kalman Filter to nonlinear system– Cubature Kalman Filter is introduced.This algorithm has its theory basis consisted of Gaussian Bayesian theory and Spherical-radial rules.In the light of the unique properties of Cubature Kalman Filter:a derivative-free on-line sequential-state estimator, computational complexity grows as n^3 and it eases the curse-of-dimensionality problem.The paper focuses on studying the CKF algorithm in depth, and applys it to bearing-only maneuvering target tracking.Two typical tracking model about tracking maneuvering ship from both non-maneuvering and maneuvering platform is selected.The Monte-Carlo simulations' results illustrate that the CKF algorithm outperforms EKF in accuracy and calculation efficiency of filter, and which make Cubature Kalman Filter easy and feasible to broad application prospect.

4aSP9. Combining Capon and Bartlett spectral estimators for detection of multiple sinusoids in colored noise environments. Chengshi Zheng and Xiaodong Li (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, cszheng@mail.ioa.ac.cn)

Detection of multiple sinusoids in colored noise environments has many potential applications, such as sonar, radar, and communication. Most of conventional algorithms often use the local signal-to-noise-ratio (LSNR) as a test statistic to detect the sinusoids, where the LSNR is estimated in the frequency domain by using the Bartlett spectral estimator (BSE). Unfortunately, the BSE has a relatively low frequency resolution, which may degrade the detection performance significantly. To solve the frequency resolution problem of the BSE, this paper proposes a two-stage hybrid algorithm to estimate the LSNR. In the first stage, the BSE is used to estimate the noise power spectral density over frequency. After obtaining the noise power spectral density, the second stage employs the Capon spectral estimator (CSE) to estimate the LSNR. The proposed hybrid algorithm significantly improves the detection performance, especially when the sinusoids are closely spaced. Simulation results show that the proposed algorithm performs much better than the conventional algorithms in most cases.

4aSP10. Restoring clipped speech signal based on spectral transformation of each frequency band. Makoto Hayakawa, Masanori Morise, and Takanobu Nishiura (Ritsumeikan University, 1-1-1 Nojihigasi, Kusatsu, Shiga, Japan, is033080@ed.ritsumei.ac.jp)

In recent years, high-quality speech recording is requested for comfortably using a communication system. However, a clipping distortion caused by input exceeding the maximum range of amplifier is one of the problems with a sound quality degradation. Although linear prediction method has been conventionally proposed for restoring a clipped speech signal, it has a problem that the frequently clipped speech signal degrades the restoration performance by increasing the prediction errors. In this paper, we propose a method for restoring a clipped speech signal based on a spectral transformation of each frequency band. In this method, the spectral envelope of target speech signal in each frame is approximated to the spectral envelope of original speech signal to remove the influence of a clipping distortion. In particular, the spectral envelope in higher frequency domain including a static characteristic of the speaker is replaced with the spectral envelope of the unclipped speech signal prepared in advance. Then, the spectral envelope in lower frequency domain including a characteristic of phoneme is approximated with Gaussian Mixture Models. We carried out an evaluation experiment for sound quality of speech signal processed by the proposed method. As a result, we confirmed the effectiveness of the proposed method.

4aSP11. A study on the technique using fractal dimension in the selection of the kind of sound. Kenji Muto, Hideo Shibayama, Yoshiaki Makabe (Shibaura Institute of Technology, Dept. of Communications Engineering, 3-7-5 Toyosu, Koto-ku, Tokyo, Japan, k-muto@sic.shibaura-it.ac.jp), Kikuo Asai, and Kimio Kondo (Center of ICT and Distance Education)

There is a paper which described that the selection of the sound which has a different fractal dimension is possible by aural. The fractal dimension of each short time window of conversation sound was analyzed, referring to the paper. In this paper, we showed about the technique using fractal dimension in the selection of the kind of sound. The analyzed sound was a conversation sound with which the engine sound and the chirp sound mix. The value of fractal dimension in the case of the chirp sound of bird or the engine sound indicated a value different from the fractal dimension in the case of the conversation sound. We thought that our technique has shown the characteristic of sound source by one parameter. It is possible to use the fractal dimension to judge a mixing surrounding sound in the teleconference at the direction of the speaker where the voice was used. To transmit a clear voice in the teleconference, the voice is utilized to the estimation in the direction of the speaker. In the real system, the case that the voice mixes with surrounding sound, it is distinguishable by fractal dimension.

4aSP12. Reproduction of human-phonatory radiation characteristics with a polyhedron loudspeaker. Naoki Yoshimoto, Masanori Morise, and Takanobu Nishiura (Ritsumeikan University, 1-1-1 Nojihigashi, Kusatsu, Shiga, Japan, is046081@ed.ritsumei.ac.jp)

Spoken dialogue systems have been studied for car navigation systems and voice search systems. For evaluation, a loudspeaker is used instead of a human because these systems require various kinds of speech samples. However, the sounds radiated by loudspeaker can not reproduce human-phonatory radiation characteristics. Therefore, the mouth simulator is utilized to reproduce human-phonatory radiation characteristics. Although it is based on the average mouth shape, shapes of mouth are different among phonemes. Therefore, due to the hardware structure, it can not accurately reproduce various human-phonatory radiation characteristics affected by shapes of mouth. In this study, we developed a polyhedron loudspeaker to solve this problem. It consists of eleven loudspeakers which are independently controlled. Controlling eleven loudspeakers makes it possible to reproduce desired radiation characteristics. Besides, we try to reproduce human-phonatory radiation characteristics of each Japanese five vowel with an adaptive algorithm based on the MINT (Multi-input/output INverse Theorem). We carried out an experiment to verify the effectiveness of the proposed method. As a result, it was confirmed that phonatory radiation characteristics of Japanese five vowels could be accurately approximated compared with the mouth simulator.

4aSP13. A spatial domain processing method for the direct signal separation in the reverberant field. Benyu Wu (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, wubenyu@mail.ioa.ac.cn)

A spatial domain processing method for the direct signal separation from the same frequency reflections at different frequencies in the reverberant field is proposed. The reflections with the same frequency contained in the reverberant acoustic signals are distributed uniformly and irregularly in space. With the same direct signal being kept, the method makes the reverberant acoustic signals in the different spatial positions synchronized and superposed. Simultaneously the same frequency reflections are eliminated, and then the direct signal is separated. It overcomes the difficulty that the direct signal can't be obtained because of the effect of reflections. It is important and significant for the acoustic measurement or the other researches based on the direct signal. The results show that the method is verified feasible and effective, by the experiments at different frequencies.

4aSP14. Aircraft flight parameter estimation via multipath delays using ground-based microphone array. Wei Xie, Luyang Guan, Ming Bao, and Xiaodong Li (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, xiewei@mail.ioa.ac.cn)

The acoustic signal from a low flying aircraft received by ground-based microphone array is characterized by the interference fringe pattern which is caused by reflection of the ground. In this paper, a model is developed to estimate the parameters of an aircraft's motion with the assumption that the aircraft flies in a straight line at a constant height. The model estimate the speed and height of the aircraft based on the different time of arrival due to the multipath delay at different locations. And the short-time cepstrum method is adopted to estimate the multipath delay accurately. To evaluate the performance of the model, the experimental results and error analysis are presented.

4aSP15. Digital communication system using beamsteering for difference frequency in a parametric array. Chong Hyun Lee, Jaeil Lee, Jinho Bae, Dong-Guk Paeng (Jeju National University, 690-756, chonglee@jejunu.ac.kr), Seung Wook Lee, Jungchae Shin, and Jin Woo Jung (Hanwha Corporation, 730-904)

Digital acoustic signal processing can be applied to sonar and acoustic communications. Especially, transmitting acoustic signal to the desired direction has many applications in military and industry fields. In this paper, we present a steerable digital communication system using parametric array transducers. To evaluate the proposed system, we build digital communication system by using transducer array, power amplifier and Labview software. The Labview software is composed of two parts. The first part is designed to generate beam to the desired direction by changing parameters such as number of sensors, complex weight of each sensor, type of transmit data and etc. The second part is to generate modulated signal of ASK, FSK and PSK, and to demodulate the received signals. With laboratory experiments, we verify the performance of the proposed communication system. Experimental results show that the system can be used to mitigate multipath effect in shallow water and can achieve high data-rate transmission

4aSP16. Interoperability of heterogeneous cores on acoustic signal processing and communication system. Shengchen Cao, Zhaoli Yan, Tianhao Cui, Xiaobin Cheng, and Xiaodong Li (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, caoshengchen@mail.ioa.ac.cn)

An acoustic signal processing and communication system is typically constituted with FPGA, DSP and application processors. These heterogeneous cores have different speed, throughput and peripheral interfaces, which results in barriers of data sharing among them. Texas Instruments' open-source component DSPLINK is an applicable solution for ARM and DSP communication. In this paper, DSPLINK architecture is optimized, FPGA is designed to act as part of the memory pool and SPI based synchronization protocol between the three cores is implemented. As an example of application, a system for cavitations monitoring based on OMAP and FPGA platform is introduced subsequently. Linux OS and DSP/BIOS are ported to ARM end and DSP end respectively, based on which the interoperable architecture is implemented. The developers can use interoperable APIs for data transfer so that more applications can be easily derived.

4aSP17. The realization of precision time protocol for distributed acoustic and vibration measurements. Longhua Ma (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, malonghua@mail.ioa.ac.cn)

Distributed acoustic and vibration measurement based on Local Area Network (LAN) becomes a hot topic, recently. The performance of this measurement system is affected dramatically by the synchronization precision of different measurement nodes. Traditional synchronous methods, such as Network Time Protocol (NTP), Simple Network Time Protocol (SNTP) doesn't meet the precision needed for distributed acoustic and vibration measurements, because it can only achieves accuracy of microseconds. In addition, sync cable is not suitable for long distance distributed acoustic and vibration measurement due to the inconsistent delay. To sovle this problem, a method based on Precision Time Protocol (PTP, IEEE1588) is proposed in this paper for synchronization of distributed acoustic and vibration measurement device. A Field Programmable Gate Array (FPGA) is employed between Medium Independent Interface (MII) and PHY, which monitors all ingress and egress data packets. Only PTP packets are unpacked and proceeded. According to these packets, the time offsets are calculated. Then time offsets between master clock and slave clock are filtered, and the output of the filter is used to compensate the drift of crystal oscillator. Synchronization experiment show the proposed method can achieve synchronization accuracy of few hundred nanoseconds.

4aSP18. A distributed array processing using multi-channel signals over a network with an embedded time code by the network time protocol. Yoshifumi Chisaki, Tomohisa Mashima, and Tsuyoshi Usagawa (Kumamoto University, chisaki@cs.kumamoto-u.ac.jp)

A conventional microphone array system uses a conductor to wire from a microphone to an input via an amplifier. While, a wireless transmission for an array system makes the configuration flexible, and it is expected to provide novel applications widely, such as measuring of impulse response in wide area. One of the issues is a synchronization of time between channels. This paper proposes multiple signals transmission system over a network with a time code embedding to synchronize those signals. The system

consists of clients and a server. Each client at a receiving point runs a network time protocol client daemon, and a packetized audio signal with a time code is sent to a server. The signal from a client is reconstructed at a server based on the time code. Since the time differences between clients affects to performance of the multichannel signal processing, small error in time at a client is preferred. This paper discusses how the error in time between channels affects to performance of the distributed microphone array system.

4aSP19. Loudspeaker compensation using mixed phase technology. Chao Ye, Ming Wu, Shuaibing Wu, and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, cye@mail.ioa.ac.cn)

In a sound reproduction system loudspeakers always introduce distortions. The compensation of loudspeaker responses using digital signal technology is becoming an important part of improving sound reproduction quality. Several FIR and IIR filter design methods have been proposed to equalize the response of loudspeaker systems. However, high order filters are needed to obtain excellent resolution at low frequencies. From a psychoacoustic point of view, warped filters have been employed to improve the resolution at low frequencies, but at the expense of poor resolution at high frequencies. On the other hand, the use of warped filters increases the complexity of the filter structure. In this paper, a mixed phase technology is proposed to equalize the response of loudspeaker systems. Loudspeaker response can be decomposed into minimum and excess phase components, which are inverted respectively to construct the equalization filter. An all pass filter based on position-independent excess phase zeros is cascaded with a minimum phase filter to improve the phase response in the region. The experimental results demonstrate that the time-domain response of the loudspeaker systems is improved by using the presented mixed phase technology.

4aSP20. Estimation of demodulation ratio for the parametric loudspeaker based on spectral envelope. Daisuke Ikefuji, Masanori Morise, and Takanobu Nishiura (Ritsumeikan University, Nojihigashi 1-1-1, Kusatsu, Shiga, Japan, cm000074@ed.ritsumei.ac.jp)

A parametric loudspeaker with the higher directivity transmits the sound wave to only particular area. It emits an AM (Amplitude-Modulated) wave demodulating into an audible sound wave by nonlinear interaction in the air. Therefore, longer distance is required for fully demodulating the AM wave into the audible sound wave. On the other hand, a power of the audible sound wave decays depending on the distance. Thus, the parametric loudspeaker must be utilized on the suitable distance for reproducing the fully demodulated wave with an enough power. However, no criterion has been conventionally proposed to measure the demodulation ratio in the past. Accordingly, the criterion should be formulated to measure demodulation ratio for appropriately utilizing the parametric loudspeaker. Thus in this paper, we aim at formulating it for estimation of demodulation ratio. We therefore propose the criterion based on a spectral envelope of the reproduced TSP (Time-Stretched-Pulse) with parametric loudspeaker. The proposed criterion is defined based on the correlations between the spectral envelope in every distance and that in the target with the reproduced TSP. As a result of the objective experiment, we confirmed the availability of the proposed criterion.

4aSP21. Harmonic distortion measurement for a parametric loudspeaker with logarithmic time stretched pulse. Shohei Masunaga, Daisuke Ikefuji, Masanori Morise, and Takanobu Nishiura (Ritsumeikan University, 1-1-1 Nojihigashi, Kusatsu, Shiga 525-8577, Japan, is037089@ed.ritsumei.ac.jp)

A parametric loudspeaker which utilizes the ultrasound can transmit the audible sound to a particular area. However, the sound reproduced by a parametric loudspeaker contains the harmonic distortions because the sound is demodulated by the nonlinearity in the air. Thus, measuring the harmonic distortions is required to evaluate the sound quality of a parametric loudspeaker. Sinusoidal wave method has been used as the harmonic distortion measurement. In it, the harmonic distortion is measured by analyzing the integral multiplication frequency of a reproduced sine wave. Many measurements by using sine waves with each different frequency are required to measure the wideband harmonic distortions. Therefore, measuring the wideband harmonic distortions with sinusoidal wave method requires much more time. Recently, using Log-TSP (logarithmic time stretched pulse) signal was proposed to measure the wideband harmonic distortions in a short time for non-parametric loudspeakers. Thus in this paper, we attempt to measure harmonic distortion of a parametric loudspeaker by using Log-TSP signal. We carried out an objective evaluation experiment in a soundproof room. The result by using Log-TSP signal was compared with that by using sinusoidal wave method. As a result, we confirmed the result with Log-TSP is equivalent to that with sinusoidal wave method.

4aSP22. The realistic- reverberation sensation in 3-D acoustic sound field reproduction with parametric loudspeakers and indirect non-parametric loudspeakers. Hideya Tsujii, Masanori Morise, and Takanobu Nishiura (Ritsumeikan University, 1-1-1, Nojihigashi, Kusatsu, Shiga, Japan, cm007077@ed.ritsumei.ac.jp)

In the field of virtual reality, mixed reality (MR) has recently been drawing attention as a technology to experience the virtual space by superimposing the computer graphics (CG) objects in real space. The technologies to reproduce 3-Dimensional (3-D) acoustic sound fields in conjunction with MR can especially experience a higher realistic sensation than the conventional MR. A parametric loudspeaker based on an ultrasound with the higher directivity can reproduce 3-D acoustic sound fields by designing a sound image. However, it is difficult to present the realistic sensation depending on the reverberation, because an acoustic wave emitted by it shouldn't diffuse in the room. In this paper, we newly proposed to reproduce 3-D acoustic sound fields utilizing multiple parametric loudspeakers and indirect non-parametric surround loudspeakers for realizing the realistic- reverberation sensation. It could realize a higher realistic- reverberation sensation without affecting the sound image localization. The effectiveness of the proposed method was assessed by the subjective evaluation experiments for the realistic- reverberation sensation and the sound image localization. As a result, we confirmed the effectiveness of the proposed method.

4aSP23. Distant-talking speech enhancement based on spectrum restoring with phoneme labels. Naoto Kakino, Takahiro Hukumori, Masanori Morise, and Takanobu Nishiura (Ritsumeikan University 1-1-1, Nojihigashi, Kusathu, Shiga, Japan, is012085@ed.ritsumei.ac.jp)

With the development of communication and speech recognition technology, remote conference and robot communication systems have been developed in recent years. Although these are valid in close-talking speech, distant-talking speech degrades the performance of the systems. The reason is that it is affected by energy decay, noise and reverberations depending on the distances. To solve this problem, many noise reduction and speech enhancement approaches have been proposed. In the conventional approaches, the example-based speech enhancement is one of the effective noise reduction methods in distant-talking conditions. The most similar example of noisy speech to input signal is detected from model examples of clean and noisy speech signals. Thereby it can estimate and suppress the noise based on the clean speech. Although it can effectively reduce noise, it is not clear the performance in the distant-talking speech condition with the decay of speech energy. Thus in this paper, we proposed a distant-talking speech enhancement based on spectrum restoring with phoneme labels. It restores the clean spectrum from decayed spectrum based on spectrum envelopes for each phoneme label. We demonstrate the experiment to evaluate the effectiveness of proposed method. As a result, we confirmed that it can effectively enhance the target speech.

4aSP24. The determination of dynamic subtraction for spectral subtraction towards musical tone reduction. Keisuke Horii, Takahiro Fukumori, Masanori Morise, Takanobu Nishiura, and Yoichi Yamashita (Ritsumeikan University, 1-1-1, Nojihigashi, Kusatsu, Shiga, Japan, cm010077@ed.ritsumei.ac.jp)

We should reduce the unwanted noise from noisy speech. Spectral Subtraction (SS) which is one of the noise reduction methods has been proposed by S. F. Boll in 1979, and it can effectively reduce the unwanted noise by utilizing only the observed signal. SS however has a problem that dissonant noise called musical tone is generated after noise reduction. SS estimates the noise with non-speech part and subtracts the estimated noise from observed signal. Flooring process is also performed depend on estimated

noise power for supporting the excessive subtraction. Since the musical tone is generated by it, the improvement of SS is required to reduce it. In the past, we proposed SS with the weighted subtraction coefficients in each frequency band for controlling flooring process. In this method, the equal-weighted subtraction coefficients were utilized in every frame, however speech and noise power are different in each frame. To overcome this problem, we newly propose an advanced SS with the color-weighted subtraction coefficients in each frame for effectively reducing the musical tone. Both objective and subjective experiments were carried out for verifying the effectiveness of the proposed SS. As a result, the proposed SS could subjectively reduce the musical tone.

4aSP25. Realtime face recognition system with ultrasonic sensing. Benxi Cao (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, xixiwelcome@gmail.com), Jingyao Wang, Yang Wang, Yong Xu, and Jun Yang

Unlike vision-based approaches, ultrasonic sensing systems have the ability to obtain the object distance and echo energy information. With the development of airborne ultrasonic detection, ultrasonic face recognition has been discussed. However, the existing ultrasonic face recognition systems store the echo waveform data and make processing and analysis afterwards. Seperated acquisition and analysis procedures make these systems non-realtime. In this paper, a realtime ultrasonic face recognition system is proposed. The system has the following functions: signal generation, transmitting, receiving, amplification, demodulation, spectral analysis, and feature extraction. A continuous wideband ultrasonic signal is transmitted, then the geometrical information can be extracted from the echo signal, and suitable pattern recognition methods are used to recognize human face. The system can implement realtime face recognition with ultrasonic sensing, and a considerable recognition rate is achieved.

4aSP26. Robustness of the ultrasonic face recognition method to age variation: analysis and experimentation. Jingyao Wang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, xuancao2009@hotmail.com), Benxi Cao, Yang Wang, Yong Xu, and Jun Yang

Ultrasonic human face recognition systems transmit ultrasonic Continuously Transmitted Frequency Modulated (CTFM) signal, extract facial geometry characteristic information from the received echo signals, and process the information for face recognition. Compared to vision-based approaches, ultrasonic human face recognition systems reduce the influence of illumination and avoid privacy leak. The existing approaches showed the feasibility of ultrasonic face recognition and achieved acceptable recognition accuracy. However, the aging problem, which equals to performance degradation caused by facial geometry change with time going on, is not considered in the previous researches. In this paper, robustness of the ultrasonic face recognition method to age variation is analysed. A database is built, in which each subject's face information was acquired at intervals of months during the last two years. Based on the database, various recognition experiments are conducted using the different pattern recognition algorithms. Aiming at the experimental results, the analysis to age factor that influences face recognition performance is implemented. The feature extraction methods and the pattern recognition algorithms are developed to increase the recognition rate.

4aSP27. Multi-stage identification for abnormal/warning sounds with onomatopoeia models. Junpei Ogawa, Kohei Hayashida, Masanori Morise, Takanobu Nishiura, and Yoichi Yamashita (Ritsumeikan University, 1-1-1 Nojihigashi, Kusatsu, Shiga, Japan, cm002074@ed.ritsumei.ac.jp)

In recent years, the methods utilizing environmental sounds have been increasingly employed for monitoring the safety of the elder who lives in distant place. Environmental sounds should consist of various sounds in daily life, and identified ones enables to detect abnormity. To detect abnormity, it is therefore required that abnormal/warning sounds are accurately identified among environmental sounds. In the past, environmental sound identification method has utilized acoustic models constructed by each sound source for all environmental sounds. However, as it only stores a few training abnormal/warning sounds, it is difficult to accurately design each abnormal/warning acoustic model. To overcome this problem, we propose a multi-stage identification method for abnormal/warning sounds. In the first stage, it utilizes an abnormal/alarm acoustic sound model and each normal acoustic sound model for detecting the abnormal/alarm sounds with a few training abnormal/warning sounds. In the second stage, it utilizes some acoustic models specializing in onomatopoeia to accurately identify abnormal and alarm sounds. We conducted an evaluation experiment to confirm the effectiveness of the proposed method on identification accuracy of abnormal/warning sounds. As a result, we confirmed that the proposed method was superior to the conventional method in identification accuracy.

4aSP28. A general audience rating surveying system and channel retrieval algorithm based on TV set audio features. Jingru Huang, Xin Ma, Lan Tian, and Shibin Du (School of Information Science and Engineering, Shandong University, Shanda South Road #27, Jinan 250100, hjingru2007@126.com)

A general surveying system for audience rating based on audio information are introduced. In this system, the audio signal is sampled from the audio outlet of TV set and high compressed into a special digital package which includes watching timing, audio signal features, and the marks of the TV channel switching. The packaged audio features are high robustness for different types of television sets and different audio volume and have no disturbance of surroundings. In the channel retrieval algorithm, multi-thresholds are used for calculating the correlation coefficients of audio features and matching the audio pattern, and the second searching is adopted for combining match when the retrieved TV channel of the head package is different from the one of the end package. The simulation test results show that the audio recognition rate is above 93%, and for other audios without the TV standard channels can be detected steadily. Keywords: audience rating; audio retrieval; robustness; recognition rate

Session 4aUWa

Underwater Acoustics and Signal Processing in Acoustics: Time Series Analysis and Data Processing in Underwater Acoustics I

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Chao Sun, Cochair
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Invited Papers

9:20

4aUWa1. The pattern of echoes in feature space. Xiukun Li (Acoustic Science and Technology Laboratory, Harbin Engineering University, Harbin, China, xiukun_li@yahoo.com.cn), and Zhi Xia (Acoustic Science and Technology Laboratory, Harbin Engineering University, Harbin, China)

In order to increase the performance of recognition in underwater bottom object detection, the distribution pattern of different kinds of echoes is researched in this paper. The past literatures more focus on the characteristics of object echoes in feature space, but the research of joint distribution pattern of object echoes, reverberation and other interference is few. In this paper, reverberation and fake object echoes are taken as the mainly interference for recognizing object echoes, and they are assumed have steady characteristic in time-frequency feature space, respectively. There are two problems are discussed: the separability between different kinds of echoes in feature space, and the stability of them. To deal with these two problems, feature compression and cluster analysis are adopted. And the feature space is generated by FDWT. The data acquired from a lake experiment is processed in this paper, and the processing results prove that object echoes, fake object echoes and reverberation within limited time scale have steady distribution pattern in FDWT feature space.

9:40

4aUWa2. Non-Rayleigh reverberation statistics. Nicholas Chotiros (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, Texas 78713-8029, chotiros@arlut.utexas.edu)

In sonar reverberation, the superposition of numerous scattered sound waves tends to produce a Rayleigh distribution, consistent with the Central Limit Theorem. Causes for deviation from the Rayleigh model are identified as sonar configuration and environmental variability. In the former, the sonar configuration determines the number of scatterers in each resolution cell, and when the number is too small, the Central Limit Theorem is violated. In the latter, the total environment may be considered as a patchwork of local environments that are resolvable by the sonar system, but not reliably distinguishable due to positional inaccuracies and overlap in the range of reverberation amplitude values. In that case, the resulting ensemble may have a probability distribution function that is a mixture of the probability distribution functions of the local components. The patchiness of the environment determines the number of components and their proportions in the mixture. The issue of stationarity in the context of a patchy environment is an important concern. Although the reverberation from a patchy environment is, strictly speaking, non-stationary, the perception of stationarity may be achieved. [Work supported by the Office of Naval Research, Ocean Acoustics].

10:00

4aUWa3. Acoustic forward scattering by a moving object and its range estimation in littoral experiment. Bo Lei, Kunde Yang, and Yuanliang Ma (Institute of Acoustic Engineering, Northwestern Polytechnical University, lei.bo@nwpu.edu.cn)

When an object crosses the source-receiver line, the sound field aberration can be caused by the forward scattering. However, the aberration is difficult to be seen because of the direct blast overwhelming. An experiment was conducted in littoral environment with several hydrophones deployed at different depths. A repeated wideband LFM pulse is transmitted and data is processed with correlation. The experimental results show that the sound field aberration takes minimum values if the object is located mid-point along the source-receiver line, whereas it attains its maximum if the object is close to the source or receiver. The total field is either enhanced or suppressed if the object crosses different Fresnel zones. In addition, the duration of shadow-induced aberration is dependent on the width of the first Fresnel zone, which is longest at the mid-point of the source-receiver line. Furthermore, a range estimation scheme is proposed. The scheme uses two-point field aberrations of the stable arrival caused by forward scattering of moving object. The ranges of intruder are estimated with a prior knowledge of the moving speed, which agree well with the measurements.

10:20

4aUWa4. Diversity combining for long-range acoustic communication in deep water using a towed array. Hee-Chun Song (UCSD 9500 Gilman Drive, La Jolla, CA 92093-0238, hcsong@ucsd.edu), and William Hodgkiss (UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238)

A recent experiment showed that coherent long-range acoustic communication is feasible in deep water over a ~550-km range between a source towed slowly at 75-m depth and a horizontal line array towed at 3.5 knots at 200-m depth. This paper further demonstrates that diversity combining mitigates channel fading and increases the output SNR. Using sparse channel-estimate-based

equalization, three transmissions are combined successfully to decode a 40-Hz bandwidth (230-270 Hz) 8-PSK (phase-shift-keying) communication signal, achieving an effective data rate of 17 bits/s at ~550 km range.

10:40–11:00 Break

11:00

4aUWa5. Fluctuations of arrival time and amplitude for short-range experimental data. Rui Duan, Kunde Yang, and Yuanliang Ma (Institute of Acoustic Engineering, Northwestern Polytechnical University, ykdzym@nwpu.edu.cn)

The arrival time and amplitude fluctuations of about 230-m propagation were analyzed for various source-receiver configurations using experimental data. The minutes-scale and seconds-scale fluctuations of the arrival time were observed on both the direct, surface-reflection and bottom-reflection arrivals while the minutes-scale fluctuation of the arrival amplitude was only observed on the direct arrivals. Cross-correlation coefficients and frequency spectrum of the fluctuation were calculated to explain the causes of the fluctuations. It shows that the movement of the source is an important cause for the fluctuations of the arrival time and the seconds-scale fluctuations of the arrival amplitude. The variability of the ocean structures contributes to both the arrival time and the amplitude fluctuations of the direct arrivals while the sea-surface scattering is the dominant cause for the surface-reflection arrivals. The fluctuation amplitudes of the bottom-reflection and the surface-reflection arrival amplitudes are around 2dB and 7dB, respectively. The fluctuation amplitudes of the direct arrival amplitude range from 1dB to 10dB for different source-receiver configurations.

Contributed Paper

11:20

4aUWa6. Temporal coherence of acoustic signals in range dependent background. Yin Quan Zhang and Ning Wang (Ocean University of China, Qingdao, zhyq_ouc@126.com)

Temporal coherence of acoustic signals is important for many practical applications, which has been studied both theoretically and experimentally. In previous theories, the fluctuation of sound speed field is assumed to be caused by linear random internal waves and the background is assumed to be range independent. This assumption has two obvious shortcomings: (1) actual fluctuations in shallow water are usually modulated by internal tides; (2) seabed has significant effects on acoustic signal propagation in shallow water, the properties of which is, in general, range dependent (such as topography). In the present presentation, internal tides and seabed properties are involved as the range dependent background, in which the background acoustic field is expressed as coupled mode matrix. A semi-analytic formalism for the temporal coherence of acoustic signal is presented. The dependence of temporal coherence on range and frequency, and the impact due to range-dependence is analyzed. The result is compared with the case range-independent.

Invited Paper

11:40

4aUWa7. Robust adaptive beamforming based on convex optimization. Lianghao Guo, Xin Guo, and Feng-Xiang Ge (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-Huan-Xi Road, Beijing, China, glh2002@mail.ioa.ac.cn)

Adaptive beamforming is an efficient way of spatial filtering in the presence of interference and noise. However, the conventional adaptive beamforming, e.g., MVDR, may degrade significantly due to the poorly estimated covariance matrix or steering vector errors. Convex optimization has now emerged as a major signal processing tool and made a significant impact on numerous problems because of its foundational nature and potential ability in signal processing. Thus in this paper, several robust adaptive beamforming algorithms based on the convex optimization are presented and evaluated, where a novel mathematical tool called "Yalmip" is used to solve the second-order cone problem in these algorithms. Numerical simulations and comparison show that these robust adaptive beamforming algorithms still approach robust to the above-mentioned problems.

Contributed Papers

12:00

4aUWa8. The study about through signature of underwater target based on focused beam-forming. Xue-li Sheng (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001, shengxueli@yahoo.com.cn), Yan-qiong Liu (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001 and Dalian Scientific Test and Control Technology Institute, Dalian 116013), Chun-ping Zhai (Dalian Scientific Test and Control Technology Institute, Dalian 116013), Shi-cai Zhu (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001 and Dalian Scientific Test and Control Technology Institute, Dalian 116013), Yu-dong Liu, Zuo-Xi Tian (Dalian Scientific Test and Control Technology Institute, Dalian 116013), and Jun-ying Hui (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001)

In shallow water, the peak and energy of through signature change because of the stack of reflected sound from the sea bottom and sea surface, it will cause much error in signal process. In this paper, the focused beaming-forming method is introduced in through signature processing. Factors of influencing through signature based on focused beaming-forming method are analyzed and the parameters describing through signature are presented. Processing results of actual measurement signal show that if the parameters describing through signature are not affected significantly, it can restrain noise effectively to smooth the through signature curve based on focused beaming-forming method. It also enhances the gain of through signature. Key words: multi-path effects; underwater acoustic image; beaming-forming; through signature; the gain;

12:20

4aUWa9. High-resolution direction finding without coherent signals number based on multibeam system. Jie Zhuo (Institute of Acoustic Engineering, Northwestern Polytechnical University, jzhuo@nwpu.edu.cn), and Bing Li (State Key Laboratory for Manufacturing System, Xi'an Jiaotong University)

Multibeam acoustic imaging systems are widely utilized for both large- and small-scale underwater investigations, especially for detecting underwater target. Processing backscattered echoes from the targets, the system can plot an acoustic image of target, and then estimate the parameters, such as direction and range. The backscattered echoes of target are coherent signals, and can be modeled as multiple highlights. Usually, the distance between the highlights is close, and the number of the highlights is unknown. Then, the directions cannot be directly estimated due to these highlights are not distinguished in the acoustic image. In this paper, an high-resolution method is proposed for estimating the directions of arrival (DOAs) of coherent highlight signals without signals number. The multibeam underwater acoustic imaging technology and the beamspace MUSIC algorithm are combined together. In this method, eigen-decomposition is skipped, so that coherent signals' DOA can be estimated by high-resolution MUSIC method under the situations with unknown signals number. Due to estimating the DOAs of coherent signals in beamspace, the precision of DOA estimation is enhanced, and the computation complexity is less than the element-space MUSIC method. The method's feasibility and robustness are analyzed and verified by the simulation data.

THURSDAY MORNING, 17 MAY 2012 S426 + S427, 9:20 A.M. TO 12:40 P.M.

Session 4aUWb

Underwater Acoustics and Signal Processing in Acoustics: Waveguide Invariance Characterization and Processing

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Invited Papers

9:20

4aUWb1. Group speed versus phase speed analysis of sound speed fluctuations in a shallow water ocean. W. A. Kuperman, Bruce D. Cornuelle, W. S. Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238, wkuperman@ucsd.edu), and Philippe Roux (ISTerre, Universite Grenoble 1, CNRS UMR 5275, Grenoble, France)

Data collected during eight consecutive hours between two source-receiver arrays in a shallow water environment (Roux et al, J. Acoust. Soc. Am. 124, 3430-3439, 2008) are analyzed using the physics associated with the waveguide invariant. In particular, the use of vertical arrays on both the source and receiver sides provide launch and receive angles in addition to the travel times associated with each eigenray path in the waveguide. From travel times and source-receiver angles, each eigenray amplitude is projected into group velocity-phase velocity (vg-vp) space for each acquisition. The time evolution of the vg-vp representation during the 8-hour long

experiment is discussed. Group speed fluctuations observed for a set of eigenrays with upper turning points at or near the thermocline are compared to independent sound speed measurements obtained by a CTD chain at two locations in the area.

9:40

4aUWb2. Autocorrelation-function-based estimation approach and variability of waveguide invariant in fluctuated shallow water. Zhou Shihong, He Li, Ren Yun, and Zhang Renhe (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, shih_zhou@yahoo.com.cn)

Waveguide invariant, which describes the broadband acoustic field interference striations in underwater acoustic spectrograms due to multipath phenomena of sound propagation, is beneficial to be applied to temporal and spatial array signal processing specially for large-aperture horizontal line array, etc. However, the dependence of waveguide invariant on fluctuated water-column environment due to internal waves or internal tides is often complicated and shows different interference patterns, which influents the performance of temporal and spatial array signal processing. This presentation focuses on variability of waveguide invariant in fluctuated shallow water environment. One novel estimation approach of waveguide invariant based on phase-shift-compensation of autocorrelation functions at two hydrophones with horizontal range separation is presented. The experimental airgun-emitted pulse data acquired in iso-speed shallow water environment is used to verify the estimation approach. Based on it, the variability of waveguide invariant in fluctuated shallow water with internal wave effects are analyzed using long-term observed LFM signals obtained by one sea-floor located horizontal-line-array and one fixed sound source. The contribution of low- and higher- order filtered modes to interference patterns is given for explaining the mechanism of mode interference. Some interesting experimental results are shown.

10:00

4aUWb3. Variability of the waveguide invariant in a range independent shallow-water environment. Kevin Cockrell (Massachusetts Institute of Technology, Dept. Mech. Engineering, Cambridge, MA 02139, kevincockrell@gmail.com)

While it is generally true that the waveguide invariant β is approximately equal to one in many shallow-water environments, the value of β observed in acoustic intensity striation patterns often deviates from the assumed value of one by 30% or more. The precise value of β, like the acoustic field itself, depends on source frequency, source and receiver depths, sound speed profile, range, etc. For example, previous work has found that the observed value of β in shallow water can deviate from the typical value of one if the source and receiver are located below the thermocline, because in that case the acoustic intensity is strongly affected by lower order modes which do not have $\beta \approx 1$ [D. Rouseff, Waves in Random Complex Media, 2001]. This talk will further explore the dependence of β on the source and receiver depths, and on the sound speed profile by using the WKB approximation. Because the group of modes which dominates the acoustic intensity depends on range and frequency, the dependence of β on range and frequency will also be explored.

10:20

4aUWb4. The waveguide invariant β and bottom reflection phase-shift parameter P. Erchang Shang, Jinrong Wu, and Zhendong Zhao (Key Laboratory of Underwater Acoustic Environment, Institute of Acoustics, Chinese Academy of Science, 100190, ecshang32@aol.com)

It is known that the waveguide invariant β is affected by the shallow-water environment. In this paper the effect due to bottom sediment is investigated. It is found that the effect of sediment bottom can be concentrated on one parameter P – the bottom reflection phase shift parameter. For a Pekeris waveguide, under WKB approximation, a very simple analytic relation is given: $\beta \approx 1+P/(k0Heff)$, here Heff is the 'effective depth', and Heff =H+P/2k0. It is shown that the value of β related to different high-speed sediments (including layered sediment) is ranged in 1.0 and 1.5. Some numerical examples including the layered sediment case are conducted to verify this result. Good agreement between the results calculated by KRAKEN and by WKB with parameter P has been found. The advantage of using parameter P is that it can cover any type of high-speed sediment even including shear wave effect, moreover, it also works for layered sediment provided using P(ømega) instead of constant P. Hence, by using parameter P allows us to have a model-free platform to investigate the sound field in shallow-water including the bottom effect on the waveguide invariant β. [This work was supported by NSFC under Grant No.10874201 and No.11074271]

10:40–11:00 Break

4a THU. AM

Contributed Papers

11:00

4aUWb5. Waveguide invariant and dedispersion transform. Gao Dazhi (Ocean University of China, dzgao@ouc.edu.cn)

The waveguide invariant is described as a single scalar parameter for a given waveguide environment. The notion of waveguide invariant has been applied widely in under water acoustic source ranging, beamforming etc. A Fourier-like transform called to dedispersion transform, which can remove the dispersion of multi-modes at the same time, was proposed in our previous paper ("Ning Wang, 9th. Western Pacific Acoustic Conference Beijing 2009"). In this presentation, we refine the relationship between the waveguide invariant and the dedispersion transform, and the dedispersion transform is extended to low frequency region and typical summer waveguides. Numerical simulations and several real-data processing will be also reported.

11:20

4aUWb6. Acoustical monitoring of the second mode internal solitary wave on oceanic shelf. Andrey Lunkov (A.M.Prokhorov General Physics Institute, Russian Academy of Sciences, 38 Vavilov st., 119991 Moscow, Russia, landr2004@mail.ru), Hwung-Hweng Hwung (National Cheng Kung University, 5th F., 500, Sec.3, Anming Rd., Tainan 70955, Taiwan), Valeriy Petnikov (A.M.Prokhorov General Physics Institute, Russian Academy of Sciences, 38 Vavilov st., 119991 Moscow, Russia), Yu-Huai Wang (National Sun Yat-Sen University, No. 70, Lien-hai Rd, Gushan, Kaohsiung, Taiwan, 804, R.O.C), and Ray-Yeng Yang (National Cheng Kung University, 5th F., No. 500, Sec. 3, Anming Rd., Tainan, Taiwan 709)

The possibility of monitoring second gravitational mode internal solitons using interference pattern frequency shifts is discussed. The investigation is carried out for the shelf of the northern South China Sea near the

Dongsha Atoll by means of numerical modeling. We used the data of the in-situ internal solitons measurements in this area. The "vertical modes and horizontal rays" approach is implemented to calculate the low frequency sound fields in the 3d environment. Stationary acoustic path is oriented at the right angle to the preferred internal wave propagation direction. A sound source and receivers are deployed at the sea bottom. Sound receivers are located at the different ranges where horizontal refraction is pronounced, and where it is insignificant. Numerical experiments demonstrate that some of acoustic waveguide modes are focused, and others are defocused in the horizontal plane when the second mode soliton propagates across the acoustic path. It is shown that the second mode internal solitary wave parameters can be successfully reconstructed from the frequency shifts in the spectrum of received signals only if horizontal refraction effects are weak. [Work supported by Russian Foundation for Basic Research and National Science Council of Taiwan # 10-02-92005]

11:40

4aUWb7. Generalized array invariant and its application on broadband source ranging. Qi Chun Shang, Shuang Zhang, and Ning Wang (Ocean University of China, 238 Songling Road, Qingdao, China, maymayed2007@126.com)

A passive source ranging method based on the array invariant in shallow water is discussed in this paper. The arriving time and elevation angle of sound are used to describe the multi-modal propagation and single-mode dispersion. Based on the two parameters, we rederive the array invariant in a different way, which allows simple physical interpretation. The array invariant in its original form is not exact invariant, but depends (weakly) on mode number and frequency. The accuracy of the method based on this notion is limited when the sound speed of seabed is different significantly from that in the water column. A modified technique (generalized array invariant) is proposed in this talk to improve the problem provided when the sound speed of seabed is known. The proposed method is testified by simulation and experimental data.

12:00

4aUWb8. Impulse signal reconstruction using bi-receiver data. Haozhong Wang, Ning Wang, and Dazhi Gao (College of Information Science and Engineering, Ocean University of China, 238 Songling Road, Qingdao 266100, China, coolicejiao@hotmail.com)

Signal reconstruction is used widely in target identification and communication in underwater. A novel method for impulse signal reconstruction using the observed data of two receivers that are arranged in the same depth with a certain horizontal interval, is proposed in this talk. This method needs no a prior environmental information but the ranges between the source and the receivers. Although the Green's function depends on the range, frequency and on the source/receiver depth, the spectrum of signal is only dependent on the frequency. The waveguide invariant notion of shallow water provides a compensation mechanism between the frequency and range shift. According to this mechanism, the amplitudes and phases of Green function spectral ingredients can be extracted respectively. The impulse signal is then obtained by employing the deconvolution. The method is applied to the signal reconstruction of a high S/N ratio real data, the correlation coefficient between the reconstructed and original signals is over 0.95.

12:20

4aUWb9. Time-reversal focusing stability in the presence of background internal waves in shallow water. Valeriy Petnikov and Andrey Lunkov (A. M. Prokhorov General Physics Institute, Russian Academy of Sciences, 38 Vavilov St., 119991 Moscow Russia, petniko@kapella.gpi.ru)

Effect of background internal waves on spatial and temporal low frequency sound focusing stability is investigated for an open oceanic shelf by means of numerical simulations. Focusing is achieved with the time-reversal procedure at a single source-receiving element at 10km range. Calculations are performed in terms of normal mode coupling theory. Internal wave field modeling is carried out using an averaged experimental power spectrum of vertical thermocline displacements measured in the Shallow Water'06 experiment. The results of numerical experiments show that the focal spot is stable for about 1 hour in the presence of typical internal waves on an open shelf. Two adaptive time-reversal algorithms are proposed to increase this period up to 12 hour. [Work supported by RFBR 11-02-00779.]

THURSDAY AFTERNOON, 17 MAY 2012 HALL A, 1:55 P.M. TO 6:00 P.M.

Session 4pAA

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

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Chair's Introduction—1:55

Invited Papers

2:00

4pAA1. Syllables intelligibility in relation to the autocorrelation and cepstrum model: the case of Chinese in Taiwan. Chiung Yao Chen (Chaoyang Univ. of Technology, chychen@cyut.edu.tw)

The articulation of some special pronunciations of vowel inconsistently raises with the rapid speech transmission index (RASTI) tested using monosyllables for Chinese phonics. In the researching of speech intelligibility in room, the factors being considered have to include not only the qualities of sound field but the pronunciation characteristics of syllables as well. Therefore, with regard to the defect of RASTI measurements, we utilized the autocorrelation and the cepstrum of monosyllables recorded in rooms to compare the articulation with the physical phenomena of intelligibilities. Thus, we found that the minimum effective duration of autocorrelation function (τ)