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 ?jgoing 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.

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

4a THU. AM

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

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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 tradeoff 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)

of all testing syllables signals recorded in rooms, associated well with the articulation for each individual room. Specially, for individual syllable signals, they are further shown significant correlation between the cepstral energy of monosyllables and the articulation collected in all rooms. The cepstrum acts as a spatially intelligible detector for syllables, and the autocorrelation is a good response of the pronunciation characteristics of syllables.

2:20

4pAA2. The role of early reflections for definition and source separation. Antti Kuusinen, Jukka Pätynen, Philip Robinson, and Tapio Lokki (Aalto University School of Science, P.O. Box 15400 00076, Aalto, Finland, Antti.Kuusinen@aalto.fi)

The density and spatial distribution of early reflections change the perception of individual instruments in an orchestral concert. Here, we present results of a listening test that aims to find the role of early reflections to perceptual phenomena related to source separation. The stimuli are anechoic symphony orchestra recordings convolved with different impulse responses that comprise of simulated direct sound and early reflection patterns combined with the late reverberation measured from a real hall. The differences between samples are investigated with various listening test methods, enabling the simultaneous comparison of samples. Quantitative data are also collected with the applied test methods. The results are expected to confirm our hypothesis that listeners can distinguish individual instruments better if the early reflection patterns in conjunction with individual sources differ more from each other.

2:40

4pAA3. Does listener weighting of binaural cues take advantage of the binaural statistics of reverberant environments? G Christopher Stecker (University of Washington, 1417 NE 42nd St, Seattle, WA 98105, cstecker@uw.edu), and Andrew D Brown (University of Washington, 1417 NE 42nd St, Seattle, WA 98105)

A series of experiments quantified listeners' weighting of auditory spatial information conveyed by interaural differences of time (ITD) and level (ILD) across cue type (ITD-ILD "trading") and over the durations of brief sounds ("temporal weighting"). Results demonstrated the dominance of cues, especially ITD, carried by the onsets of rapidly modulated or continuous tones. Whereas post-onset ITD information received little weight for such sounds, post-onset ILD was more influential, especially near sound offset [Stecker and Brown, *JASA* 127:3092-103. 2010]. As a consequence, the relative weighting of ITD and ILD changes systematically with modulation rate, with greater weighting of ILD in cases where post-onset ITD is unavailable [Stecker, *Hear. Res.* 268:202-12. 2010]. Greater weighting of post-onset ILD than ITD is consistent with observations of dramatic ITD distortion by echoes [Rakerd and Hartmann, *JASA* 78:524-33. 1985] and of a greater role for ILD in dynamic aspects of the precedence effect [Krumbholz and Nobbe, *JASA* 112:654-63. 2002] resulting from changes in the acoustic environment. Evidence regarding temporal weighting of ITD and ILD will be reviewed and compared to the statistics of binaural cue values across a variety of reverberant recordings. [Supported by NIH R03-DC009482, R01-DC011548, F31-DC010543, T32-DC000033]

3:00

4pAA4. Binaural room acoustics II: Distributions and consequences of interaural differences. William M. Hartmann (Michigan State University, 4208 BPS Bldg., East Lansing, MI 48824, hartmann@pa.msu.edu), Brad Rakerd, and Eric J. Macaulay (same)

Binaural room acoustics attempts to generalize the acoustical properties of rooms as they appear at the two ears of a listener. Because of the importance of sound localization, interest has focused on interaural differences in intensity level and phase. Distributions of these interaural differences, as measured with an artificial head in different rooms, are in reasonable agreement with spherical-head computations having either the direct-to-reverberant ratio or the reverberation time as the main parameter. The psychological relevance of these distributions was tested in experiments using rooms with different reverberation times where listeners were required to report the source azimuth for steady-state pure tones having frequencies between 200 and 1200 Hz. Simultaneously probe microphones in the ear canals recorded interaural differences. Interest centered on the choices made by listeners between plausible and implausible interaural differences resulting from the sound fields in the room. Particular attention was given to measurements made near the binaural critical distance [Hartmann and Rakerd, J. Acoust. Soc. Am. **130** 2352 (2011). [Work supported by the AFOSR, grant 11NO002]

3:20

4pAA5. The contribution of interaural time and level differences to the precedence effect at high frequencies. Bernhard U. Seeber (MRC Institute of Hearing Research, University Park, Nottingham, NG7 2RD, UK, seeber@ihr.mrc.ac.uk)

The precedence effect (PE) allows us to locate sound sources correctly in rooms despite the presence of interfering reflections. It has been shown to function at high frequencies with highly modulated stimuli. These studies were done in the free-field where interaural time (ITDs) and level (ILDs) differences are in their natural combination. The present study investigated the relative contribution of ILDs and ITDs to the PE with high-frequency zero-phase harmonic complex tones. A localization dominance task was used in which participants indicated the location of the lead-lag stimuli and judged if sounds were perceived as fused. A preliminary analysis indicates that the PE emerged when either ITDs or ILDs were applied to lead and lag stimuli while the other binaural cue was held at zero. Patterns for localization dominance and fusion were nearly identical for ITD and ILD conditions, suggesting that ITDs and ILDs were equally effective for these highly modulated stimuli. Fusion of lead and lag extended to somewhat longer delays with smaller cue magnitudes, i.e. the more binaural cues differed between lead and lag the more likely they were to be segregated. The results support the idea that PE mechanisms are similar for ITDs and ILDs.

3:40

4pAA6. Echo thresholds for reflections from acoustically diffusive architectural surfaces. Philip W. Robinson (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, Troy, New York, philrob22@gmail.com), Andreas Walther, Christof Faller (Audiovisual Communications Laboratory, école Polytechnique Fédérale de Lausanne, Switzerland), and Jonas Braasch (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, Troy, New York)

Diffusive architectural surfaces play an important role in performance venue design for architectural expression and proper sound distribution. However, previous psychoacoustic research on perception of reflections and the precedence effect has focused on specular reflections. This study compares the echo threshold of specular reflections, measured using an adaptive up-down method with music and

speech stimuli, against those for reflections from realistic architectural surfaces, and against synthesized reflections that isolate individual qualities of reflections from diffusive surfaces, namely temporal dispersion and spectral coloration. It is found that temporal dispersion up to 16ms in the reflection response, and peak amplitude reduced as much as 18.5 dB results in an echo threshold shorter than that for a specular reflection of comparable amplitude. Rather, the threshold is comparable to that of a specular reflection of similar energy. This indicates that the auditory system integrates the temporally dispersed energy into a single stream.

4:00–4:20 Break

4:20

4pAA7. A binaural model that uses head-movements to evaluate acoustical spaces. Jonas Braasch (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180, braasj@rpi.edu), Samuel Clapp, Anthony Parks, and Ning Xiang (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180)

Binaural models have a long tradition in the instrumental analysis of acoustical spaces. Room acoustical parameters such as the Binaural Quality Index (BQI) are derived directly from the measured Binaural Room Impulse Response (BRIR) of a concert space. The BRIRs are measured using an artifical head with a fixed head position and consequently cannot show the effect of head movements, which are essential for human listening performance. Based on a novel model architecture that utilizes head movements [Braasch et al., 2011, J. Acoust. Soc. Am. 129, 2486 (A)] and psychoacoustic experiments, the effect of head movements on the perceptual judgment of room acoustical parameters will be discussed. In addition, BRIRs for different azimuth angles are calculated from higher-order ambisonic microphone data that were obtained in several concert halls in the Northeastern United States. [Clapp et al., 2011, J. Acoust. Soc. Am. 130, 2418 (A)]. It will be demonstrated how the standard acoustical parameters change with head orientation and how dynamic head-movement cues can be utilized to better predict the perceived quality of concert spaces.

4:40

4pAA8. Three dimensional representation method for a public indoor soundscape with multiple sound sources. Yasushi Shimizu (Tokyo Institute of Technology, 226-8502, shimizu.y.ai@m.titech.ac.jp), and Hiroshi Furuya (Shibaura Institute of Technology, 135-8548)

The author has been investigating an evaluation method of a reproduced sound such as paging sound, background music and so on in a public space. The current acoustical descriptors for an evaluation of such sounds, which are played to deliver information and attention to the people, have a difficulty to apply to multiple sounds environment. This report describes a new evaluation method which utilizes drawing an indoor soundscape with multiple sounds from listening experience, based on major aural impressions such as Loudness, Timbre, Apparent Sound Source Width, horizontal and vertical Sound Localization, Distance Perception, and Perspicuity, “KIWADACHI”. This will be applied to describe the Indoor Sound Environmental Character with multiple sound sources. The evaluation tests with this tool are carried out in the indoor sound environments of a retail shop, regarding aural impression of Perspicuity for the sound reproduction. And the results of both the subjective representation in the indoor sound environment and the acoustical descriptor of speech intelligibility are presented for the reproduced sound.

5:00

4pAA9. Using time-varying loudness to model the reverberance of rooms. Densil Cabrera, Doheon Lee, and William L. Martens (The University of Sydney, NSW 2006, Australia, densil.cabrera@sydney.edu.au)

The reverberant decay of sound over time is one of the most perceptually salient features of reverberation in rooms. This paper examines the concept that the perception of reverberation decay can be modelled using dynamic loudness. As well as being a plausible application of time-varying loudness modelling, this approach helps to explain why higher sound pressure level stimuli are more reverberant than otherwise identical reverberant stimuli – because the slope of the loudness decay function depends on the stimulus level. Loudness decay parameters are derived in analogy to conventional reverberation parameters (reverberation time and early decay time), which provide a better match to subjective experimental data concerned with: impulsive, running, and overall reverberance; and using artificial and measured room impulse responses.

5:20

4pAA10. Loudness asymmetry in real-room reverberation: cross-band effects. Andrew Raimond and Anthony Watkins (Reading University, Reading RG6 6AL, UK, andrew_raimond@hotmail.co.uk)

Although room reverberation adds sizeable “tails” at the ends of sounds, they are not prominent for listeners. Evidence for this comes from loudness judgements of stimuli with envelopes having fast onsets and slow offsets, thereby resembling sounds with reverberant tails. Such sounds are less loud than their reversed counterparts, and this difference is more substantial when the test-sound is preceded by a “standard” sound that has a similarly “tail-like” offset. A perceptual constancy may be responsible; one where sounds with decaying tails are “parsed” to separate source characteristics from effects of reverberation, and to discount energy in tails from listeners’ judgements. Indeed, this “loudness context effect” is even more substantial when real-room reverberation is used. Here we ask whether the effect is restricted to the frequency region occupied by the context; conditions where standard and tests occupy the same narrowband frequency region are compared with “cross-band” conditions, where the standard and test have widely separated frequency regions. Results show that the effect is markedly reduced in cross-band conditions, indicating that the perceptual constancy responsible is a “within band” phenomenon. Similar within-band characteristics are also evinced by a form of constancy in speech perception, where the salience of “tails” from reverberation is also reduced.

5:40

4pAA11. The desire for decorrelation—applications from the recording studio. Alexander Case (University of Massachusetts Lowell, 35 Wilder St, Lowell, MA 01854, alex_case@uml.edu)

Multitrack production solves challenges of masking, localization, and intelligibility while pursuing aesthetics associated with reverberance, distance, envelopment, and source width through any means available in the signal processing-rich environment of the recording studio. Contemporary sound recording techniques are presented that might influence the design for achieving similar results through architectural signal processing.

Session 4pAB

Animal Bioacoustics: Tropical and Sirenian Bioacoustics

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Peter Narins, Cochair
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Invited Papers

2:00

4pAB1. Ultrasonic hearing in frogs: inner ear morphological correlates. Peter Narins (UCLA, 621 Charles E. Young Drive S., Los Angeles, CA 90095-1606, pnarins@ucla.edu)

Three species of anuran amphibians (*Odorrana tormota, O. livida* and *Huia cavitympanum*) have recently been found to detect ultrasounds. We compared morphological data collected from the ultrasound detecting species with data from *Rana pipiens*, a frog with a typical anuran upper cut-off frequency of ca. 3 kHz. In addition, we examined the ears of two species of Lao torrent frogs, *O. chloronota* and *Amolops daorum* that live in acoustic environments resembling those of the ultrasonically sensitive frogs. Our results suggest that the three ultrasound-detecting species have converged on small-scale functional modifications of the basilar papilla (BP), the high-frequency hearing organ in the frog inner ear. These modifications are also seen in the ears of *O. chloronota*, suggesting that this species is a candidate for high-frequency hearing sensitivity. These data form the foundation for future functional work probing the physiological bases of ultrasound detection by a non-mammalian ear. Supported by NIDCD DC-00222, Paul S. Veneklasen Research Foundation, and the UCLA Academic Senate (3501).

2:20

4pAB2. High-frequency sound communication in the concave-eared frog. Jun-Xian Shen (Institute of Biophysics, CAS, 15 Datun Road, Chaoyang District, Beijing 100101, China, shenjx@ibp.ac.cn)

The concave-eared torrent frog, Odorrana tormota, is an arboreal, nocturnal frog living near noisy fast-flowing streams in Huangshan China. Recordings in the field show that males produce diverse melodic calls containing spectral energy extended to the ultrasonic range. Playbacks of the audible as well as the US components of a male call can evoke males' vocal responses. Auditory evoked potentials from the auditory midbrain confirm that males possess the US hearing capacity. Before ovulation, gravid females produce high-frequency short calls, which elicit vocalization and precise positive phonotaxis from males. Acoustic playbacks of male's calls also evoke vocal responses and phonotaxis from females, but the females show no ultrasonic sensitivity. This suggests that the high-frequency sound communication system has evolved in the frog species. [Work is supported by the National Natural Science Foundation of China (NSFC grants Nos. 30570463 and 30730029 to J.-X.S.]

Contributed Papers

2:40

4pAB3. Anatomical changes in the inner ear of the bullfrog across metamorphic development. Erika E. Alexander (Brown University, Campus Box 1821, Providence, 02912, Erika_Alexander@brown.edu), Andrew M. Tarr, and Andrea M. Simmons (Brown University, Campus Box 1821, Providence, 02912)

Metamorphic development in the bullfrog, Rana catesbeiana, is characterized by widespread changes in peripheral transduction pathways and in the auditory brainstem, in preparation for the transition from a fully aquatic to a semi-terrestrial existence. The time course of development of the inner ear organs has not been as extensively examined. A combination of immunohistochemical, cresyl violet and trichrome staining to were used to delineate the development of the saccule, an otolith organ sensitive to particle motion and to seismic stimuli, across metamorphosis. From early embryonic to metamorphic climax stages and extending to the froglet period, the saccule increases linearly in area, correlated with the growth in body size. Myosin VI label indicates that hair cell density in the central region of the saccule remains relatively stable in tadpoles, but then decreases between froglet and subadult stages. From these results, it is hypothesized that hair cell proliferation occurs more extensively in tadpoles than in froglets.

3:00

4pAB4. Photolyses of carboxy-hemoglobin of bar-headed goose studied by photoacoustic calorimetry. Jin-yu Zhao (Lab of Modern Acoustics, College of Physics, Nanjing University, Nanjing 210093, China, jyzhao04118@gmail.com), Jia-huang Li (Lab of Pharmaceutical Biotechnology, College of Life Sciences, Nanjing University, Nanjing 210093, China), Zheng Zhang (Nanjing First Hospital Attached to Nanjing Medical University, Nanjing 210006, China), Min Qu, Shu-yi Zhang (Lab of Modern Acoustics, College of Physics, Nanjing University, Nanjing 210093, China), Zi-qian Hua (National Laboratory of Protein Engineering and Plant Genetic Engineering, College of Life Sciences, Peking University, Beijing 100871, China), and Zi-chun Hua (Lab of Pharmaceutical Biotechnology, College of Life Sciences, Nanjing University, Nanjing 210093, China)

As a specialized species native to high altitude, bar-headed goose can fly annually over an altitude of 9000m, which means that its hemoglobin has a higher oxygen affinity than its lowland relatives, such as goose and chicken. To study the mechanism of the phenomena, laser ultrasonic calorimetry is used to study dynamic processes associated with photolyses of carboxy hemoglobin (HbCO), including the enthalpy and conformational volume changes, of bar-headed goose and its lowland relatives. Considering the

time scales of the reaction lifetimes in the photolyses processes of HbCO, two kinds of piezoelectric transducers, a PVDF film and a PZT ceramic, are used as acoustic signal detectors. For evaluating the relative enthalpy change and the relative conformational volume change in the process, the quantum yield of the photolysis must be taken into account, which has been measured by pump-probe technique. The results show that the enthalpy and conformational volume changes of bar-headed goose are obviously smaller than that of its lowland relatives and human. Some analyses and discussions on the differences of the amino acid sequences of Hb, the tetramer structures, as well as the salt bridges between subunits of Hb and HbCO among them are presented.

3:20

4pAB5. Design of a home-made, low-cost system for studies of vibratory courtship signals on *Pardosa Sierra* (Areneae: Lycosidae) spiders. Eduardo Romero-Vivas, Emiliano Méndez Salinas, María Luisa Jiménez Jiménez, and Francisco Javier García De León (Centro de Investigaciones Biológicas del Noroeste, S.C., Mar Bermejo 195 Col. Playa Palo de Santa Rita, La Paz, BCS, 23090, México, evivas@cibnor.mx)

Spiders possess peculiarities that make them attractive for the study of evolutionary phenomena such as adaptation and specialization. Among these processes, reproductive behavior (particularly courtship) is a main factor, allowing or preventing recognition between potential partners. Spiders sense their environment and communicate using chemical, visual and acoustical/vibrational signals. The study of the nature, variation and content of these signals, provides useful information to understand the role of communication in the formation of species. Vibrational signals excel in importance in the majority of spider families and have been previously studied, especially in leaf-living spiders, using non-contact laser Doppler vibrometers or accelerometers (adding extra mass to the system) coupled to charge amplifiers. Unfortunately, cost and availability of this equipment have limited the widespread of studies in this area. This paper describes how to build an alternative low-cost system for the study of vibrational signals on spiders, and presents the analysis of the acquired vibratory courtship signals of Pardosa sierra, a rocky substrate-dwelling lycosid spider.

3:40

4pAB6. Intraspecific variation in vocal repertoire among dugong populations. Kotaro Ichikawa (Research Institute for Humanity and Nature, 603-8047, Kyoto, Japan, ichikawa.kotaro.dugong@gmail.com), Tomonari Akamatsu (National Research Institute of Fisheries Engineering, 314-0408 Ibaraki, Japan), Kanjana Adulyanukosol (Phuket Marine Biological Center, 83000, Phuket, Thailand), Giovanni Damiani, Janet Lanyon (University of Queensland, St Lucia, Queensland, 4072, Australia), and Hiroshi Nawata (Research Institute for Humanity and Nature, 603-8047, Kyoto, Japan)

Previous studies have demonstrated that vocal signals facilitate acoustic communication of dugongs. We recorded wild dugong calls from around Talibong Island, Thailand (n = 586) and in Moreton Bay, Australia (n = 331). We also recorded vocalizations of a newborn calf (n = 315) kept at Phuket Marine Biological Center, Thailand, a 19 year old female (n = 73) at Toba Aquarium, Japan, and a 7 year old female (n = 203) at Underwater World, Singapore. Dominant frequency, duration and coefficient of frequency modulation were compared across populations and age. Statistical differences were found for almost all pairwise comparisons ($p < 0.05$) except between the captive dugongs kept in Japan and also between wild dugongs in Thailand and in Australia. A negative correlation was found between variance of the dominant frequency and dugong age, and a positive correlation was found between variance of the duration and age. The average dominant frequency of wild dugong calls collected in Thailand and in Australia were 5205.4 and 5760.2 Hz, respectively. These acoustic characteristics ranged between those of the 7 and 19 year old female. Our results suggest that dugongs change their vocal repertoire as they grow.

4:00–4:20 Break

4:20

4pAB7. Analysis of passive acoustic recordings made during a three month survey of cetaceans off the Northern Mariana Islands in the western North Pacific. Thomas Norris (Bio-Waves Inc., thomas.f.norris@bio-waves.net)

Passive acoustic monitoring using was used to complement a line-transect survey of marine mammals for a large (~580,000 km2) study site centered on the Northern Mariana Islands in the western North Pacific. A two-element towed hydrophone array was used to monitor and record during daylight hours. Sonobuoys were deployed opportunistically on sightings and areas of interest. Extremely poor sighting conditions hindered visual efforts but not the passive acoustics effort. Over 70 days of survey effort was completed from mid-January to April, 2007. Approximately 220 'unique acoustic detections' were made, of which 155 (70%) were preliminarily identified to 14 different species. The most frequent whale detected was the sperm whale (65), followed by minke whales (30) and humpback whales (12), respectively. The first recordings of calls from Sei whales in this region are characterized. Post-processing of minke and sperm whales recordings resulted in approximately 30 and over 70 localizations, respectively. We present the first acoustic-based estimates for minke whales abundance in this region. Numerous unidentified odontocete whistles were analyzed using ROCCA, a semi-automated whistle classification program, with promising results. We provide recommendations for additional analyses and improvements to methods of collecting and post-processing passive acoustic data on marine mammals.

Invited Paper

4:40

4pAB8. The Lombard effect in humpback whales. Michael Noad, Rebecca Dunlop (University of Queensland, Gatton, Qld 4343, Australia, mnoad@uq.edu.au), and Douglas Cato (Defence Science & Technology Organisation, Eveleigh, NSW 1430, Australia, and Univeristy of Sydney, NSW 2006, Australia)

The Lombard reflex is an increase in the subject's vocal levels in response to increased noise levels. While it has been demonstrated in humans and a small number of mammals and birds including some whales, it has not been demonstrated in humpback whales. During their southward migration off eastern Australia humpback whales were tracked visually from an elevated land station. An array of calibrated hydrophone buoys was used to simultaneously track vocalizing whales acoustically and to measure ambient noise. Two hundred and ninety two social vocalizations were recorded and analysed from 15 passing groups of whales when there was no detectable boat noise or singing whales in the area. Vocalization source levels increased significantly by a mean of 0.75dB per 1dB increase in background noise (broadband 40Hz – 2kHz). Unlike most previous Lombard studies, however, the vocal level increased even though the background noise was much lower than the vocal level. Thus the whales maintained a signal excess of approximately 75dB which suggests that these social vocalizations may function as signals over distances of several kilometres.

5:00

4pAB9. Variation in the songs of humpback whales (*Megaptera novaeangliae*) wintering in the Northwestern and Main Hawaiian Islands. Jessica Chen, Marc Lammers, and Whitlow Au (Hawaii Institute of Marine Biology, University of Hawaii, 46-007 Lilipuna Rd., Kaneohe, Hawaii 96744, jchen2@hawaii.edu)

A study of the humpback whale song in the Northwestern Hawaiian Islands (NWHI) and the Main Hawaiian Islands (MHI) during the 2009 season suggests that humpback whale song may be more variable than previously suggested. Data from five autonomous acoustic recorders deployed at locations in the NWHI and MHI were analyzed to compare the frequency of occurrence of song units by whales in the island chain. There appears to be a gradient of differences in song units throughout the Hawaiian Island chain, rather than the previously assumed, more discrete differences between breeding populations. Recordings from each site were randomly selected. Song units were classified as one of 23 units and counted to compare between sites. Changes in the frequency of occurrence in a few of the most abundant units suggest a gradual change throughout the island chain. However, this may be confounded by changes that occur throughout the season throughout the ocean basin. Further work examining the amount of variation both between and within humpback whale breeding populations should be conducted.

5:20

4pAB10. Comparison of automated and aural/visual techniques to classify humpback whale (*Megaptera novaeangliae*), song units. Adrienne M. Copeland, Whitlow W. L. Au (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, P.O. Box 1106, Kailua, HI 96734, acopelan@hawaii.edu), Marc O. Lammers (Hawaii Inst. of Marine Biology, Kaneohe, HI 96744), Adam A. Pack (Univ. of Hawaii at Hilo, Hilo, HI 96720), Julie N. Oswald (Bio-Waves, Inc., 517 Cornish Dr, Encinitas, CA 92024), and Jessica Chen (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, P.O. Box 1106, Kailua, HI 96734)

Humpback whale song research has focused on analyzing the full song structure rarely describing individual song units. Even less progress has been made in automatically distinguishing and classifying these individual units. Two different techniques were employed to study their call units, visual/aural and automated/statistical. Humpback whale songs were recorded in the Hawaiian Islands both remotely with an autonomous acoustic recorder and by a snorkeler with a portable digital tape recorder. Humpback whale song units collected by the autonomous acoustic recorders were aurally separated into 23 distinct units in a companion study. Song units collected by a snorkeler using the portable recorder off Maui were analyzed using a specialized Matlab script that defined 48 frequency and temporal parameters for each unit. From the 48 parameters, the units were separated into distinct categories using a multivariate categorical analysis. The distinct units were compared between the different techniques to gage if automated methods could be used in future humpback whale studies. After this comparison was made, a principal component analysis (PCA) determined which of the aforementioned 48 parameters were important in statistically distinguishing between the distinct units furthering our understanding of frequency and temporal importance in categorizing song structure.

5:40

4pAB11. Acoustic issues in studies of behavioral response of humpback whales to seismic ramp-up and hard start. Douglas Cato (Defence Science & Technology Organisation and University of Sydney, P.O. Box 44, Pyrmont, NSW 2009, Australia, doug.cato@sydney.edu.au), Robert McCauley (Curtin University of Technology, Perth, WA 6845, Australia), Michael Noad, Rebecca Dunlop (University of Queensland, Gatton, QLD 4343, Australia), Hendrick Kniest (University of Newcastle, Newcastle, NSW 2308, Australia), Nicholas Gales (Australian Antarctic Division, Kingston, TAS 7050, Australia), Chandra Salgado Kent (Curtin University of Technology, Perth, WA 6845, Australia), David Patton (Blue Planet Marine, Canberra, ACT 2614, Australia), John Noad (University of Queensland, Gatton, QLD 4343, Australia), Curt Jenner (Centre for Whale Research, Fremantle, WA 6959, Australia), Alec Duncan, and Amos Maggi (Curtin University of Technology, Perth, WA 6845, Australia)

Two large behavioral response studies (BRS) have been conducted with humpback whales migrating along the east Australian coastline (in project BRAHSS: Behavioural Response of Australian Humpback Whales to Seismic Surveys). Whales were exposed to four stages of ramp-up with nominally 6 dB increase in level at each step, and a hard start nominally 12 dB above the first stage. Observations of behavior were made by theodolite teams ashore and small boats following specific whale groups, DTAGs, and binoculars from the source vessel. The sound field throughout the area was recorded using five buoys that radioed data back to the shore station, four autonomous receivers and two drifting systems with a vertical array of four hydrophones. Measurements show that the propagation loss at the site is variable and includes patches of anomalously high loss. This complicates estimation of the sound levels received by whales, but may not be unusual in near shore environments. This paper presents preliminary results of the project to illustrate acoustic issues involved in designing and executing comprehensive BRS, including characterization of sources and the acoustic environment experienced by the whales, and monitoring cumulative exposure at individuals for mitigation.

Session 4pBA

Biomedical Acoustics: Bone Quantitative Ultrasound II

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Contributed Papers

2:00

4pBA1. Nonlinear resonant ultrasound spectroscopy is sensitive to the level of cortical bone damage. Sylvain Haupert (University Pierre et Marie Curie, F-75006 Paris, France, sylvain.haupert@upmc.fr), Sandra Guérard (Arts et Metiers ParisTech, F-75013 Paris, France), David Mitton (IFSTTAR, F-69675 Bron, France), Françoise Peyrin (INSA Lyon I, Lyon, France), and Pascal Laugier (University Pierre et Marie Curie, F-75006 Paris, France)

The objective of the study was to evaluate the sensitivity of nonlinear resonant ultrasound spectroscopy (NRUS) measurements to the accumulation of damage in cortical bone by fatigue or by controlled crack propagation. Two groups of human cortical bone specimens were prepared from the femoral mid-diaphysis. The specimens from the first group were taken through a progressive fatigue protocol consisting of four steps of cyclic four-point bending. The specimens from the second group were taken through a toughness protocol consisting of initiation and controlled propagation of a stable crack induced by 4-point bending mechanical loading. Our results evidenced a progressive increase of the normalized nonlinear elastic parameter during fatigue testing or during toughness experiments. While in specimens subjected to mechanical fatigue cycling the relative variation of nonlinear elasticity was significantly related to the relative variation of the number density of small cracks assessed with micro-computed tomography, in crack propagation experiments a significant relationship was found between the level of nonlinearity and total crack length. These results strongly suggest that NRUS measurements are sensitive to damage accumulation and can be used as a marker of bone damage.

2:20

4pBA2. Assessment of soft and mineralized tissue formation in a rat bone healing model using quantitative ultrasound (QUS). Daniel Rohrbach, Bernhard Hesse, Bernd Preininger, Carsten Perka, and Kay Raum (Charite, Julius Wolff Institut, Augustenburger Platz 1, 13353 Berlin, Germany, daniel.rohrbach@charite.de)

It is hypothesized that QUS is a promising candidate for the assessment of early stages of bone healing. 5-MHz QUS measurements in through transmission mode were conducted in vitro on a 2-mm osteotomy rat model (N=10). 2D parametric QUS images of speed of sound (SOS), ultrasound attenuation (UA), and broadband UA (BUA) were registered to histology sections and to projections obtained from 3D μ-CT images. Based on histology two groups of healing (N(A)=5: early healing stage and N(B)=5: early reparative phase) were defined. Parameter variations (medians and integrals) evaluated within the osteotomy gap region were compared between the healing groups. ANOVA revealed significantly higher attenuation and SOS values in group B (UA(A)=14.5$\pm$4.5 dB, UA(B)=35$\pm$10.1 dB, F=18.5; BUA(A)=4.09$\pm$1.7 dB/MHz, BUA(B)=9.6$\pm$4.4 dB/MHz, F=6.8; SOS(A)=1543$\pm$9 m/s, SOS(B)=1590$\pm$34 m/s, F=8.7). ROC analysis with AUC values between 0.84 and 1 confirm a good predictive power of US parameters. The bone mineral density (BMD) based variance assessed from μCT was less pronounced (F=9.5). Moderate correlations of UA and SOS with BMD were observed (R2<0.7). These results demonstrate that ultrasound parameter variations are sensitive to tissue alterations that are not depicted by BMD, but coincide with cartilage formation in the early reparative phase.

Invited Paper

2:40

4pBA3. Therapeutic ultrasound on bone cellular and in vivo adaptation. Yi-Xian Qin, Shu Zhang, Suzanne Ferreri, and Jacky Cheng (Department of Biomedical Engineering, Stony Brook University, Stony Brook, NY, Yi-Xian.Qin@sunysb.edu)

Objective: It is well documented that ultrasound, as a mechanical signal, can produce a wide variety of biological effects in vitro and in vivo. The purpose of the current study was to (1) develop a methodology to allow for in-vitro manipulating osteoblastic cells using acoustic radiation force generated by ultrasound, (2) use this methodology to determine the morphological and biological responses of bone cells to ultrasound, and (3) mitigate bone loss under estrogen deficient osteopenia. Methods: In Vitro Cellular Manipulation: We used a therapy focused transducer, which has spherical cap with 64 mm diameter and 62.64 mm focal length. A laser guide MC3T3-E1 osteoblastic cells were cultured in α-MEM containing 1% penicillin-streptomycin and 10% decomplemented newborn calf serum. In Vivo OVX Model: 72, 16 w.o. Sprague-Dawley rats were divided into six groups; baseline control, age-matched control, OVX control, OVX + 5 mW/cm^2 ultrasound (US), OVX + 30 mW/cm^2 US and OVX + 100 mW/cm^2 US. Low intensity pulsed ultrasound (LIPUS) was delivered transdermally at the L4/L5 vertebrae, using gel-coupled plane wave US transducers. The signal was applied 20 min/day, 5 days/week for 4 weeks. Results: In Vitro Cellular Response: The developed methodology allowed manipulation of MC3T3-E1 cells by acoustic radiation force. The deformation of cell membranes was observed by the US manipulation, which appeared after 15s treatment of pulsed ultrasound in 6W. We also imaged the movement of primary cilia, which showed corresponding movement when subjected to pulsed ultrasound. In Vivo Response: LIPUS treatment significantly increased BVF compared to OVX controls for the 100mW/cm^2 treated group. Interestingly, the 100mW/cm^2 treated groups showed a significant improvement over the 5mW/cm^2 treated group. Discussion and Conclusions: Pulsatile focused ultrasound can create local fluid flow nearby cells. The observed primary cilia can be triggered

to dynamic movement by the acoustic force as a mechanobiologic effect. In vivo results suggest that low-intensity pulse ultrasound can induced mechanical wave in tissue and initiate bone adaptation. These findings support the hypothesis that LIPUS can inhibit bone loss and preserve bone strength under conditions of estrogen deficient osteopenia. Keywords: quantitative ultrasound, therapeutic ultrasound, bone mechanotransduction, osteoporosis, bone remodeling

Contributed Papers

3:00

4pBA4. Effect of trabecular material property on ultrasonic backscattering in cancellous bone. Chengcheng Liu, Dean Ta, and Weiqi Wang (Department of Electronic Engineering, Fudan University, Shanghai 200433, China, estonelau@163.com)

Ultrasonic backscattering technique, used to assess cancellous bone status, is investigated using numerical simulations based on two-dimensional finite-difference time-domain (FDTD) method. High resolution microstructure mappings of bovine cancellous bone, provided by micro-CT, are used as the input geometry for simulations. This paper focused on the effects of material property parameters (density, lamé coefficients, viscosities, and resistance coefficients) of the trabecular on ultrasonic backscattering measurements at 1MHz. Simulations of ultrasonic backscattering in cancellous bone for different trabecular parameters were carried out and the backscatter coefficient (BSC) were measured and discussed. The results demonstrate that BSC is a nonlinear function of trabecular density and increases with trabecular density. While BSC is affected little by the viscosities, the first and second lamé coefficients have a complex effect on BSC. BSC is almost a linear function of normal resistance coefficient (NRC) and decreases with increases of NRC. On the other hand, BSC is practically independent of the shear resistance coefficient, just because there is little shear wave in backscattered signals. The results demonstrated that ultrasonic backscattering is affected by trabecular density and other material acoustic properties, as well as the bone mineral density and microarchitecture.

3:20

4pBA5. Correlation of ultrasonic backscatter parameters with transmission parameters and BMD in cancellous bone. Haijie Han and Dean Ta (Department of Electronic Engineering, Fudan University, No. 220, Handan Rd, Yangpu District, Shanghai 200433, China, haijie861017@126.com)

Quantitative ultrasound (QUS) has been suggested to have a performance equal to dual-energy X-ray absorptiometry (DXA) for the assessment of bone. In this paper, human cancellous bone is investigated in vivo using QUS and DXA. The ultrasonic backscatter method and its parameters (spectral centroid shift (SCS), apparent integrated backscatter (AIB)) are also introduced. The experimental ultrasonic backscatter signals are collected from 300 volunteers' calcanea in two hospitals, and the two parameters are calculated. Finally, correlation between ultrasonic backscatter parameters and transmission parameters (speed of sound (SOS), broadband ultrasonic attenuation (BUA) and stiffness index (SI)), as well as correlation between backscatter parameters and bone material density (BMD) are analyzed. The results showed that correlations between backscatter parameters and SI are better than that of SOS and BUA. SI correlates positively with SCS ($r=0.613$, $p<0.05$), as well with AIB ($r=0.463$, $p<0.05$). According to the size of SCS and AIB of ultrasonic backscattered signals, the status of cancellous bone may be assessed.

Invited Paper

3:40

4pBA6. Effects of boundary conditions on the two wave phenomenon in cancellous bone. Mami Matsukawa, Katsunori Mizuno (Doshisha University, Kyotanabe, 610-0321, Japan, mmatsuka@mail.doshisha.ac.jp), and Yoshiki Nagatani (Kobe City College of Technology, Kobe, 651-2194, Japan)

After the successful observation of two wave phenomenon in cancellous bone, wave characteristics have been investigated by in vitro studies. However, there still exists strong necessity to understand the effects of bone boundary conditions on the phenomenon, because the cancellous bone is always covered by a cortical layer in vivo. This paper is dedicated to the experimental and simulation studies of the effects on the two wave phenomenon. A sample of cancellous bone ($76.1 \times 45.2 \times 12.9$ mm^3) and two cortical plates (thickness: 1.2 and 1.9mm) were obtained from the left radius of a racehorse. Longitudinal pulse waves around 1MHz were measured by a conventional ultrasonic pulse technique using PVDF transducers. With 3-D CT image of the sample, wave propagation was also investigated by a 3-D elastic FDTD method. We then compared wave propagation in the sample covered with cortical layers or not. In both experimental and simulation approaches, two wave phenomenon was observed in the covered sample. However, the slow wave amplitude was very sensitive to the interface conditions between the cancellous bone and cortical layers. In addition, the waves often became small due to the conditions, telling the importance of sensitive wave observation in some cases.

4:00–4:20 Break

Contributed Papers

4:20

4pBA7. Bayesian-derived fast and slow waves correlate with porosity obtained from microCT. Joseph Hoffman, Amber Nelson, Mark Holland, and James Miller (Washington University in St. Louis, hoffman@wustl.edu)

It has previously been shown by our laboratory that Bayesian probability theory permits separation of ultrasonic fast and slow waves in cancellous bone even when the modes overlap substantially in time. The goal of the current study was to determine whether the fast and slow waves obtained from Bayesian separation of an apparently single mode signal individually correlate with porosity. The Bayesian technique was applied to data from cancellous bone samples from 8 human heels insonified with a broadband 500 kHz ultrasound pulse in the medial/lateral direction. The phase velocity (SOS), slope of attenuation (nBUA), and relative amplitude were determined for both the fast and slow waves. The porosity of the samples was measured by X-ray microCT. The phase velocity and slope of attenuation for both the fast and slow wave modes showed an inverse correlation with porosity. The fast wave amplitude decreased with increasing porosity. Conversely, the slow wave amplitude increased with increasing porosity. These results indicate that the properties of the individual fast and slow waves correlate with bone porosity, an important determinant of fracture risk. Supported, in part, by NIH/NIAMS grants R01AR057433 and P30AR057235.

4:40

4pBA8. Ultrasonic properties of fast and slow longitudinal waves propagating in the cancellous bone. Fuminori Fujita, Keisuke Yamashita, Katsunori Mizuno, Mami Matsukawa, and Takahiko Otani (Doshisha University, 1-3 Tatara Miyakodani, Kyotanabe City, Kyoto, Japan, bmi1009@mail4.doshisha.ne.jp)

Longitudinal wave in cancellous bone separates into fast and slow waves depending on the alignment of trabeculae. Here, the fast wave mainly propagates in trabeculae, whereas slow wave propagates in the soft tissue (bone marrow). Because these two waves usually overlap, the evaluation of each wave has still remained difficult. In this study, then, we have tried to evaluate the wave properties (attenuation and velocity), making use of the plane wave in an acoustic tube. 3-D image of the bone specimen was obtained by X-ray micro CT. In an acoustic tube, a cancellous bone specimen was set between home-made PVDF transducers. A function generator delivered a single sinusoidal signal in the range of 0.5 to 1.5 MHz to the transmitter. By filing a part of bone specimen away, we have tried to obtain the attenuation and velocity of the specimen. The two wave phenomenon clearly occurred in our specimen. In some trabecular bones of big animals, the wave separated perfectly. We have then tried to obtain ultrasonic properties of both waves and investigated the reflection and transmission at the interfaces.

5:00

4pBA9. Modeling ultrasound interaction with cancellous bone: investigation on the nature of the two compressional waves. Fabien Mézière, Marie Muller, Emmanuel Bossy, and Arnaud Derode (Institut Langevin, ESPCI ParisTech/CNRS UMR 7587/Université Paris 7/INSERM ERL U979 10 rue Vauquelin, 75005 Paris, France, fabien.meziere@espci.fr)

Although ultrasound might be useful to assess bone quality, the mechanisms of ultrasound propagation in trabecular bone are still poorly understood. For example the propagation of a short pulse that leads, under some conditions, to two transmitted longitudinal waves, a fast and a slow one, is not well explained yet. The objective of this work is to further investigate the nature of these two longitudinal waves in simplified bone model media. The approach is to determine if the fast wave could result from a guided propagation in the solid matrix, the slow wave being due to the propagation in the surrounding fluid (bone marrow). In this context, we study the propagation of the coherent waves through simplified and customizable binary structures, obtained by a random addition of scatterers, with characteristic dimensions, material properties and anisotropy similar to those of bone. The benefit of such simplified structures is that ultrasound propagation in the entire medium can be theoretically studied from the properties of a single scatterer.

5:20

4pBA10. Modeling and simulation of acoustic scattering in poroelastic materials. M. Yvonne Ou (408 Ewing Hall, University of Delaware, Newark, DE 19716, mou@math.udel.edu)

In this talk, we will present a model based on Biot's equation and numerical simulation of wave propagation in a fluid-elastic-poroelastic system, with scattering of a bone sample of cancellous bone surrounded by cortical layer and muscle layer in mind. The numerical methodology is based on the finite volume method and the operator splitting technique, combined with the grid-mappings technique and CLAWPACK. This is joint work with Randall LeVeque and Grady Lemoine form University of Washington.

5:40

4pBA11. Direct comparison of single mode versus fast and slow wave modes analyses of calcaneal bone data. Amber Nelson, Joseph Hoffman, Mark Holland, and James Miller (Washington University in St.Louis, Physics Dept., 1 Brookings Dr., St.Louis, MO 63130, nelsonam@wustl.edu)

Background: We previously demonstrated that a signal transmitted through cancellous bone might be comprised of two interfering (fast and slow wave) modes even though it appears to consist of only a single mode. Objective: The goal of this study was to compare the results of single mode analysis and two mode analysis to quantify the effects of interfering waves on the measured speed of sound (SOS) and broadband ultrasound attenuation (BUA). Methods: A series of human calcaneal samples (bone volume fractions = 0.09 to 0.21) were measured medial-laterally using 500kHz broadband focused transducers. Phase velocity was determined by phase spectroscopy and attenuation was determined by log-spectral subtraction. The original radiofrequency signal and the Bayesian-separated fast and slow wave radiofrequency signals were analyzed. Results: For each of the specimens, the slope of attenuation determined by single mode analysis was larger than that for either the fast or slow wave, and the speed of sound determined by single mode analysis lay between that of the fast and slow wave. Conclusion: The additional information provided by analyzing individually fast and slow wave modes might enhance the effectiveness of bone sonometry. Supported by NIH/NIAMS grants R01AR057433 and P30AR057235.

6:00

4pBA12. Bone ultrasound transducer. Masahiro Okino, Katsunori Mizuno, Daisuke Suga, and Mami Matsukawa (Doshisha University, Kyoto, Japan, dml0126@mail4.doshisha.ac.jp)

Low-intensity pulsed ultrasound (LIPUS) is used on bone-healing. One expected healing mechanism of this technique is the contribution of piezoelectricity. Actually, the piezoelectricity in dry bone at low frequencies has been reported by many investigators. However, the ultrasonic investigation of the piezoelectricity in bone has still remained very few. We have then made original transducers with bovine cortical bone samples, which were fabricated into plates (diameter: 10 mm, thickness: 0.5~1.0 mm). Using a conventional ultrasonic pulse technique in MHz range, we confirmed the observation of ultrasonic wave by the transducer. Here, the bone transducer was set in water at the focal position of a PVDF transmitter, which was excited by a single sinusoidal pulse of 70 Vp-p in the MHz range. The maximum voltage of the received wave was about 200 μVp-p. It was almost 1/2000 of the homemade PVDF transducer, which was made by the same procedure. The observed wave amplitude changed due to the sound wave frequency and bone thickness. Considering the inhomogeneous bone microstructure and the thickness dependence of the bone transducer, the piezoelectricity of bone in the MHz range is discussed.

Session 4pEAa

Engineering Acoustics and Physical Acoustics: Civil Non-Destructive Testing with Ultrasound or Other Non-Contact Methods I

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Invited Papers

2:00

4pEAa1. New designs of air-coupled ultrasonic transducers using micro-stereolithography. David Hutchins, Duncan Billson, Simon Leigh, and Chris Pursell (School of Engineering, University of Warwick, Coventry CV4 7AL, UK, D.A.Hutchins@warwick.ac.uk)

Micro-stereolithography, a form of high resolution rapid prototyping, has been used to design and construct novel designs of ultrasonic transducers for use in air. This approach allows for designs to be available that are difficult to produce by other means. The work will describe experiments that have been conducted with capacitive devices in two configurations: planar devices that are based on capacitive micromachined ultrasonic transducers (CMUTs), and those with a spiral geometry backplate for the generation of modified wavefronts. In addition, the fabrication technique has also been used to make structurally-modified materials (so-called metamaterials) to change the emitted characteristics in air. Examples are given of the type of applications for which these ultrasonic transducers could possibly be used.

2:20

4pEAa2. Air-coupled sensing of leaky rayleigh waves and ZGV modes in concrete slabs. Jinying Zhu (The University of Texas Austin, 1 University Sta., Austin, TX 78712, jyzhu@mail.utexas.edu), Yi-Te Tsai, Xiaowei Dai, and Michael R. Haberman (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, Texas 78712-1024)

Rayleigh waves and zero-group velocity (ZGV) Lamb modes (the impact-echo test) are commonly used for non-destructive evaluation (NDE) of concrete structures to extract information about material property and interior delaminations. Traditional methods in civil engineering employ point impactors to generate the waves while accelerometers measure the resulting out-of-plane motion. Though well-established, this methodology does not lend itself to efficient scanning of large areas, which is critical for monitoring the safety of infrastructure. The ability to detect these two wave types through in-air sensing of Leaky Rayleigh or ZGV Lamb waves can greatly accelerate NDE for large structures. Unfortunately, air-coupled sensing suffers from significantly decreased signal amplitude. This paper presents theoretical analysis and experimental validation efforts to amplify in-air signals resulting from Leaky Rayleigh waves and ZGV Lamb modes using a parabolic focusing mirror. A time-domain impulse response to the Kirchhoff-Helmholtz integral equation is presented that permits the analysis of discrete arrivals from each wave type and estimation of the expected focusing gain and depth of field based on the geometry of the parabolic dish. Analytical and numerical solutions in time-domain are compared with time-harmonic solutions taken from the literature and experimental results. This work is supported by NIST Technology Innovation Program (TIP).

2:40

4pEAa3. Toward a high power non-contact acoustic source using time reversal. Pierre-Yves Le Bas (Los Alamos National Laboratory, Geophysics group EES-17, MS D443, Los Alamos, NM 87545, pylb@lanl.gov), TJ Ulrich (Los Alamos National Laboratory, Geophysics group EES-17, MS D443, Los Alamos, NM 87545), Brian Anderson, and John Esplin (Department of Physics and Astronomy, Brigham Young University, N377 Eyring Science Center, Provo, UT 84601)

Over the last decades, nonlinear acoustic techniques have been developed to detect mechanical damage in solids. They have been proven to be far more sensitive to early damage than standard linear acoustic techniques. Unfortunately, they often require high amplitude waves to propagate within the sample. To be practical in industrial applications, signals have to be generated without contact. Currently available non-contact transducers are generally not powerful enough. A first step toward the creation of a high amplitude non-contact acoustic source will be described. This source is based on the principle of focusing energy on the surface of a sample using Time Reversal in a hollow cavity. By using a laser vibrometer for the necessary calibration of the system we are able to use a full non-contact process. New development in signal processing allows a much cleaner signal generation than usually achieve with time reversal. This source is proven to be much more energetic than current off the shelf non-contact transducers. It is a broad band source with an adjustable standoff distance and also has the capability to selectively generate in-plane waves. This work is supported by the U.S. Department of Energy through the LANL/LDRD Program.

3:00

4pEAa4. Non contact long distance exploration method for concrete using SLDV and LRAD. Tsuneyoshi Sugimoto, Ryo Akamatsu (Toin Univ. of Yokohama, tsugimot@cc.toin.ac.jp), Noriyuki Utagawa, and Syuichi Tsujino (Sato kougyo Co., Ltd.)

The hammering test is a representative method in inspection for cavities and delaminations at shallow area of concrete surface. Although this method is used widely because it is not expensive, efficiency of the defect-judging largely depends on the tester's experience and long measurement time is necessary for wide area inspection. Other methods have been developed, however, it is necessary to contact or approach to the inspection object during a measurement. Therefore, we propose a new non-contact acoustic imaging method for nondestructive inspection using scanning laser Doppler vibrometer (SLDV) and long range acoustic device (LRAD). In this method, Surface vibration, which is generated by air borne sound, is measured using SLDV. This time, the styrofoam board was buried at shallow depth in the concrete are used as a substitute of a cavity in the concrete. As an experimental result, a styrofoam board is clearly imaged by the vibration velocity of the concrete surface. Furthermore, we confirmed that our proposed method can apply even 10 m away distance, and the measurement distance is about within 20 m under the present conditions. It means that the non destructive inspection for concrete from a long distance is possible.

3:20

4pEAa5. Wavelength measurement of guided waves in plates with electromagnetic acoustic transducers. Guofu Zhai (Harbin Institute of Technology, P.O. Box 401, Harbin Institute of Technology, No. 92, West Dazhi Street, Harbin, China, gfzhai@hit.edu.cn), Tao Jiang, Jiapeng Gong, and Lei Kang (Harbin Institute of Technology, P.O. Box 404, Harbin Institute of Technology, No. 92, West Dazhi Street, Harbin, China)

For an EMAT (electromagnetic acoustic transducer) with a meander-line coil, the interval spacing between adjacent wires of the coil is generally designed to be the half wavelength of guided waves excited by the EMAT. However, the actual value of the wavelength deviates from the double interval spacing (double spacing) because of the variation spectrum of the spacing and the bandwidth of excitation frequency. So it is difficult to obtain the actual value of the wavelength of guided waves in plates. Based on dispersion equation and group velocity equation of guided waves, we find that the actual value is double spacing when the frequency of the excited signal is equal to the frequency of the received one; and the actual value deviates from the double spacing when the EMATs' operating points are not on the dispersion curves. Taking SH wave EMATs for instance, this paper proposes two wavelength measurement methods with different initial conditions for SH waves in plates. The validity of the proposed methods is verified by experiments.

3:40

4pEAa6. In-situ characterization of hedges with a parametric acoustic transducer. Kirill Horoshenkov, Amir Khan (School of Engineering, University of Bradford, Bradford, BD7 1DP, UK, k.horoshenkov@bradford.ac.uk), Hongseok Yang, Chris Cheal, Julija Smyrnowa, and Jian Kang (Sheffield School of Architecture, University of Sheffield, Arts Tower, Western Bank, Sheffield S10 2TN, Sheffield, UK)

This work reports on the results of outdoor measurements which were carried out using a highly directional parametric acoustic transducer and a 2-D array of microphones. The purpose of these measurements was to determine the frequency and angular dependence of the acoustic transmission loss of hedges (hedgerows). The experimental setup and procedure used in this work enabled us to simulate an approximately plane wave propagation regime and to reduce considerably the effect of ground on the recorded transmission loss data. The results show that a 1.5-2.5m thick hedge can provide a considerable (up to 20 dB) transmission loss which is comparable to that expected from an artificial noise barrier structure. The theory developed by Aylor (JASA 51(1), 197-205, 1972) is used to explain the observed dependencies as a function of the leaf density, size and hedge thickness. An alternative equivalent fluid theory for porous absorber (Horoshenkov et al, JASA 104(3), 1198-1209, 1998) is also used to explain the observed attenuation for sound propagation through the leaf mass.

4:00–4:20 Break

4:20

4pEAa7. Characterizing the properties of adhesive in bonded materials by laser ultrasonic method. Xiaodong Xu, Cong Liu, and Xiaojun Liu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, xdxu@nju.edu.cn)

In order to characterize the quality of adhesive in bonded materials, a metal/bonding/metal assembly model named as "sandwich" model is used to characterize the propagating properties of Lamb wave generated by laser ultrasonic method and studied in theory and experiment. In our experiment, a Nd:YAG ns laser is focused as line source and scanned on the surface of sample to supply signals in time and space, the signals of traveling lamb wave is detected by a beam deflection system. The dispersion curve of Lamb wave is deduced based on 2D FFT of obtained signals. According to the theoretical "sandwich" model, the real part and imaginary part, which is corresponded to the attenuation coefficients of sample, of elastic parameters of sample in wideband frequencies can be determined by theoretical fitting to the dispersion curve, and the determined parameters are reflected the changing properties of the adhesive in bonded materials.

4:40

4pEAa8. A quantitative imaging of bonding strength at the bonded solid-solid interface obtained by nonlinear test. Jianjun Chen (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, jjchen@nju.edu.cn)

As well known, when a longitudinal wave propagates through an interface with micro-cracks and micro-defects between solids, the contact acoustic nonlinearity (CAN) will be generated dramatically and the nonlinear parameter can be used to contour the bonding state of the interface. However the contour can only show the relative state of bonding strength and can not be used to judge weather the multilayered composite materials in use is safe because the safe judgment is not by the relative state while by the absolute value of the bonding strength. Therefore characterization of quantitative bonding strength at the interface is very important for judging a multilayered material in safe use. In this paper, how to get the quantitative bonding strength from the CAN parameter is studied. After the vibration amplitude of incident focusing wave at the bonded interface was calculated, the standard bonding strength with complete bonding state was established by tension test and CAN parameter is calibrated, the quantitative imaging of the bonding strength is obtained by CAN microscope in experiments. From the imaging, the positions with weak bonding strength could be easily located, which can be used to decide whether the material could be employed continuously.

5:00

4pEAa9. Nonlinear spring model for adhesive bonding interface. Zhiwu An (Beisihuanxi Road 21, Haidian District, Beijing, China, anzhiwu_111@163.com), Xiaomin Wang, Mingxi Deng, Jie Mao, and Mingxuan Li

The boundary conditions for an interface between two solids are analyzed to model second harmonic generation in a thin elastic adhesive layer. The approximate boundary condition models termed as "nonlinear spring models" are rigorously developed using an asymptotic expansion of the exact solutions in the limit of small ratio of interface layer thickness to wavelength. The applicability of such boundary conditions is analyzed by comparison with exact solutions for ultrasonic wave transmission. Numerical calculation indicate that as the acoustic nonlinearity increases, the model tends to be more accurate, meanwhile the second-harmonic amplitude increases in direct ratio. The present nonlinear spring models may provide a potential to evaluate the nonlinear mechanical behavior of bonding interface. Acknowledgment: This work is supported by the National Natural Science Foundation of China (Grant No.10834009)

5:20

4pEAa10. Noncontact MASW methods for near surface soil/infrastructure assessment. Zhiqu Lu (National Center for Physical Acoustics, The University of Mississippi, 1 Coliseum Dr. University, MS 38677, zhiqulu@olemiss.edu)

In near surface geophysics, a multi-channel analysis of surface wave (MASW) method has been increasingly applied for underground infrastructure assessments, in which conventional contact sensors like geophones and accelerometers were mostly employed to detect surface vibrations. In this study, noncontact sensors technology such as a laser Doppler vibrometer (LDV) and a microphone were used to measure Rayleigh wave and leaky Rayleigh waves respectively. These noncontact sensors were installed in a scanning platform that was driven by a stepper motor. The scanning MASW system consisted of two excitation sources: an electromechanical shaker and steel-balls to generate frequency sweeping (chirp) signals with frequency from 30 Hz to 500 Hz and high frequency (up to 40 kHz) impulsive signals respectively. The LDV-shaker-MASW was developed for near surface soil profile exploration and the microphone-steel ball-MASW was built for pavement assessment. The details of the system and several case studies will be addressed.

5:40

4pEAa11. Investigation of simultaneous signal transmission in non-destructive inspection of steel billet. Yoko Norose, Koichi Mizutani, and Naoto Wakatsuki (University of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8577, norose@aclab.esys.tsukuba.ac.jp)

A non-destructive method of steel billet, which uses time-of-flight of ultrasonic longitudinal wave and detects defects as decrement of pseudo sound velocity reconstructed by computerized tomography, has proposed. The remaining problem is long measurement time, owing to step-by-step measurement corresponds to each sound path whose number becomes huge. One of the solutions is simultaneous measurement, which can be achieved by transmitted signals simultaneously. To avoid interference among signals, choice of the transmitted signal becomes one of the most important points to be considered. In the case of non-destructive inspection of steel billet, many reflected waves from boundaries may cause adverse influence. Therefore, in this study, the parameters of the signal; modulation scheme, modulated signal, signal length, and signal frequency are investigated. Simulation-based study suggested that the signal is required to be short and independent each other. From those conditions, some experiments about defect detection by the several kinds of modulation schemes using pseudo-random noise were conducted. As a result, defects detection by 5ch simultaneous transmission was successful. It was suggested that the total measurement time could be decreased to 1/5.

6:00

4pEAa12. A grinding acoustic emission monitoring method using wavelet transforms and fuzzy logic. Yang Jing, Zhang Zhong-Ning, Cheng Jian-Chun (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, Jiangsu Province, China, yangj@nju.edu.cn), and Liu Xiang-Xiong (Hiecise Machines Co., Ltd, Kunshan 215337, Jiangsu Province, China)

It is known that the acoustic emission (AE) signals contain potentially valuable information for grinding wheel and work conditions during automatic grinding process. However, AE signals produced in the grinding zone are usually much more complicated and it is difficult to obtain enough effective information directly from raw AE signals. This paper presents an efficient grinding AE monitoring method based on a combination of wavelet transforms and fuzzy Logic techniques. First, the discrete wavelet transforms are used to extract the feature vectors from AE signals with different grinding wheel and work states. Secondly, a fuzzy classifier is employed to build a nonlinear relationship between the feature vectors of AE signals and the various grinding states. This method requires less computation and has a low sensitivity to the changes of grinding conditions. Experiment results show that the proposed intelligent monitoring system can identify the different grinding wheel and work states accurately and is valuable for industrial applications.

6:20

4pEAa13. Research on target-matching method in ultrasonic testing. Han Zhang, Zhiwu An, Xiaomin Wang, Jie Mao, and Mingxuan Li (Institute of Acoustics, Chinese Academy of Sciences, zhanghan81@126.com)

Ultrasonic echoes contain ample flaw information in nondestructive testing. However, bad coupling or small flaw often leads to low SNR (signal-to-noise ratio), so the result provided by conventional amplitude detection may not be accurate enough. To solve this problem, the finite element method is used to investigate the frequency domain characteristics of flaw echoes. Furthermore, a target-matching method based on the maximum output SNR criteria is proposed. The transducer is excited by a certain electrical signal obtained by adaptive filtering deconvolution algorithm for each flaw, so we get a maximum SNR when the excitation signal and flaw match. This is further verified in ultrasonic test for a series of samples containing different sizes of flat-bottom holes. This work is supported by the National Natural Science Foundation of China (Grant Nos. 10834009, 11074272)

Session 4pEAb

Engineering Acoustics, Underwater Acoustics, and Biomedical Acoustics: Acoustic Sensors and Actuators I

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Contributed Papers

2:00

4pEAb1. A passive dispersive wave amplifier for high-intensity broadband acoustic pulses. Steven Dion, Martin Brouillette, and Louis-Philippe Riel (Université de Sherbrooke, 2500 boul. de l'Université, Sherbrooke (Québec), Canada J1K 2R1, steven.dion@usherbrooke.ca)

The acoustical power output of piezoelectric ultrasonic transducer is limited by the material breakdown voltage or the available driving electrical power. While there are well known ways to passively amplify monofrequency acoustic waves generated by a single transducer, e.g., with an exponential horn, there is no obvious way to similarly pump energy into a structure to produce high-intensity broadband acoustic pulses. It was found that the frequency dependant phase velocity inherent to dispersive waveguides can be advantageously exploited to generate high intensity planar pulse waves using a single transducer. With this amplification concept, gain factors as high as 15 have been measured, which can be exploited to produce shock waves in water with a conventional ultrasonic transducer and low power electronics. The paper will present the theoretical underpinnings of this method, as well as its experimental validation. Some potential biomedical applications of this technology will also be discussed.

2:20

4pEAb2. Instantaneous tracking of a monopole sound source using algebraic localization method and circular array. Tsukassa Levy and Shigeru Ando (Dept. of Information Physics and Computing, University of Tokyo, Tokyo Bunkyou-ku Hongou 7-3-1, levy.t@alab.t.u-tokyo.ac.jp)

The purpose of this study is to localize instantaneously and to track sound source using circular array. Localization is done using a novel explicit inversion formula of direction and distance of a monopole source [S. Ando, ASA Seattle Meeting, 2011]. This method consists in applying the weighted integral method (IEEE Trans. SP, 57, 9, 2009, Inverse Problems, 26, 015011, 2010) on a partial differential equation (PDE) satisfied by the sound wave. In this study, we consider the location-constraint PDE (ASA/ASJ meeting, 2006) that describes the unilateral propagation of wide-band waves from a single source. We also take into account reverberation in this study by using a simplistic model of room reverberation. In the experiments, loci of instantaneous localization results are displayed on a motion image captured by a TV camera mounted on the 16-element circular array. Several experimental results are shown to demonstrate time-resolved localization capability of a multiply reflected source. Some theoretical and experimental comparisons with and without reverberation will be given.

2:40

4pEAb3. A mutual radiation impendence measurement method for 2-element transducer array considering nonlinear interaction. Xuesen Zhang, Peifeng Ji, and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, zhangxuesen@mail.ioa.ac.cn)

Mutual radiation impedances represent the acoustical interaction between transducers in an underwater array, which are inherent in the distance between transducers and the geometry of the array. The influence of mutual radiation impedances, which are important to evaluate the performance of the underwater array, especially needs to be intensively studied on condition that the power of transducers is high enough because of nonlinear interaction. A method for determining mutual radiation impedances of transducers in a 2-element array is proposed in this paper, which applies to estimation of mutual radiation impedances for an underwater array with high radiation power. This method derives from V method in which the mutual radiation impedances are considered as the acousto-motive force generated on the surface of each transducer. By using nonlinear acoustic analysis, theoretical foundations of this method based on equivalent circuits are described. Detailed experimental procedures to implement the method are illustrated.

3:00

4pEAb4. Mass sensing using a functionalized subordinate oscillator array. Joseph Vignola, John Judge, Aldo Glean, and Teresa Ryan (The Catholic University of America, 620 Michigan Ave, NE, Washington DC 20064, vignola@cua.edu)

Vibrometric based mass sensing for detection of trace levels of chemical vapors has been under investigation for a number of years. One implementation of such a sensor uses a vibrating MEMS cantilever that has been functionalized to bind with a specific chemical compound. The additional mass reduces the resonance frequency of the structure by an amount that can be related to the concentration of the compound. This work describes an analytic and numerical study of an array of differently functionalized cantilevers coupled to a primary structure. The dimensions of individual cantilevers are chosen such that there is a distribution of their isolated natural frequencies and the mass of the primary structure is substantially larger than the collective mass of the subordinate cantilevers. The isolated natural frequencies of the cantilevers must be packed around that of the primary structure such that the response curves of neighboring cantilevers cross near their half power points. This overlap ensures an exchange of energy between array elements. Results describing optimized distributions of cantilever

properties will be presented. Both fabrication variations and the sensed mass are considered as disorder in the system. This presentation reports how the relative magnitude of these disorders affects mass declarations.

3:20

4pEAb5. Performance of optimized sound field control techniques in simulated and real acoustic environments. Philip Coleman (CVSSP, University of Surrey, Guildford, Surrey GU2 7XH, United Kingdom, p.d.coleman@surrey.ac.uk), Martin Møller, Martin Olsen (Bang & Olufsen a/s, Peter Bangs Vej 15, DK-7600, Struer, Denmark), Marek Olik, Philip Jackson (CVSSP, University of Surrey, Guildford, Surrey GU2 7XH, United Kingdom), and Jan Abildgaard Pedersen (Bang & Olufsen a/s, Peter Bangs Vej 15, DK-7600, Struer, Denmark)

It is of interest to create regions of increased and reduced sound pressure ('sound zones') in an enclosure such that different audio programs can be simultaneously delivered over loudspeakers, thus allowing listeners sharing a space to receive independent audio without physical barriers or headphones. Where previous comparisons of sound zoning techniques exist, they have been conducted under favorable acoustic conditions, utilizing simulations based on theoretical transfer functions or anechoic measurements. Outside of these highly specified and controlled environments, real-world factors including reflections, measurement errors, matrix conditioning and practical filter design degrade the realizable performance. This study compares the performance of sound zoning techniques when applied to create two sound zones in simulated and real acoustic environments. In order to compare multiple methods in a common framework without unduly hindering performance, an optimization procedure for each method is first used to select the best loudspeaker positions in terms of robustness, efficiency and the acoustic contrast deliverable to both zones. The characteristics of each control technique are then studied, noting the contrast and the impact of acoustic conditions on performance.

3:40

4pEAb6. On frequency characteristics of bone conduction actuators—subjective and objective measurements of inner canal type and head-of-mandible type actuators. Yoshimi Fukuda, Yuki Yae, Fumihiro Nakahara (Kumamoto University, 2-39 Kurokami, Kumamoto City, Kumamoto, Japan, yoshimi@hicc.cskumamoto-u.ac.jp), Hidenori Nakatani (Goldendance Co., Ltd., 3-22-19 Furuichi, Jyoto-ku, Osaka, Japan), Koji Kobayashi (Toyota Tsusho Corporation, 4-9-8 Meieki, Nagoya, Aichi, Japan), Yoshifumi Chisaki, and Tsuyoshi Usagawa (Kumamoto University, 2-39 Kurokami, Kumamoto City, Kumamoto, Japan)

Recently, bone conduction actuators which can reproduce the high frequency signal around 10 kHz are released in the market and those are used not only for speech but also music reproduction. However, the detail characteristics of bone conduction actuators, such as threshold, loudness vibration and frequency characteristics are not yet clear. Because there are difficulties to make clear the relationship between input signal and loudness of perceived sound even if acceleration of actuator is given. In this paper, threshold and loudness of stimuli are measured by means of psychoacoustical test for two types of bone conduction actuators; inner canal type and head-of-mandible type. Frequency characteristics of actuators are measured as acceleration. Comparison between subjective and objective measurements is performed in order to make clear the characteristics of measured bone conduction actuators.

4:00–4:20 Break

4:20

4pEAb7. A micro-machined microphone based on field-effect-transistor and electrets. Kumjae Shin, Yub Je (Department of Mechanical Engineering, Pohang University of Science and Technology, PIRO 416 Hyoja-Dong, Nam-Gu, Pohang, Gyungbuk, 790-784, South Korea, forhim13@postech.ac.kr), Haksue Lee (Agency for Defence Development, P.O. Box 18, Jinhae, Changwon, Republic of Korea), James E. West (Department of Electrical & Computer Engineering, Johns Hopkins University, 3400 N. Charles St., Baltimore, MD 21218), and Wonkyu Moon (Department of Mechanical Engineering, Pohang University of Science and Technology, PIRO 405 Hyoja-Dong, Nam-Gu, Pohang, Gyungbuk, 790-784, South Korea)

Micro-machined microphones are attaching attention of industry because of their benefit of size over conventional ones. Since most of micro-machined microphones are capacitive sensors, the sizes of their electrodes determine the low frequency noise level that increases with inverse of frequency (1/f). Therefore, the size of microphone itself becomes larger than the one that can be fabricated. Here we introduce a micro-machined microphone that can overcome the limit of capacitive microphones. The proposed microphone is composed of a field-effect-transistor (FET) and an electret. The difference between the conventional electret capacitive microphones and the proposed microphone may be the transduction mechanism: the change in the position of an electret causes the change in electric field on the gate of FET. Compared with capacitive transduction, the resistive channel of FET can be designed to have low sensor impedance, and subsequntly have low impedance at low frequency. To make experimental specimen, FET onto membrane and electret were fabricated with conventional metal-oxide-semiconductor fabrication process and micromachining process, respectively. The FET membrane chip and the electret chip were assembled. Simple current to voltage converter was applied as a pre-amplifier. Its feasibility to apply low frequency acoustic sensor will be proved by simulation and experimental results.

4:40

4pEAb8. Study on improving piezoelectricity and PBLG/PMMA composite. Junsik Jeon, Yonghwan Hwang (Department of Mechanical Engineering, Pohang University of Science and Technology, PIRO 416 Hyoja-Dong, Nam-Gu, Pohang, Gyungbuk, 790-784, South Korea, jeonjs@postech.ac.kr), Micheal S Yu (Department of Materials Science & Engineering, Johns Hopkins University, 3400 N. Charles St., Baltimore, MD 21218), James E West (Department of Electrical & Computer Engineering, Johns Hopkins University, 3400 N. Charles St., Baltimore, MD 21218), and Wonkyu Moon (Department of Mechanical Engineering, Pohang University of Science and Technology, PIRO 405 Hyoja-Dong, Nam-Gu, Pohang, Gyungbuk, 790-784, South Korea)

Poly-Gamma-benzyl-L-glutamate Poly methyl methacrylate (PBLG-PMMA) composite polymer, a new piezoelectric transduction material, was invented by Yu et al. In our previous study, we have shown that the body composed of this new material can be produced by simultaneous poling and curing of a homogeneous solution comprising Poly-Gamma-benzyl-L-glutamate(PBLG) and methyl methacrylate(MMA) via detachable molding. And, it was also shown that the piezoelectric coefficient(d33) of PBLG-PMMA composite is dependent on the PBLG concentration and the intensity of electric field applied during curing in molds. In this study, we attempt to improve piezoelectricity of PBLG-PMMA composite because the piezoelectric coefficient of PBLG-PMMA was only about one twentieth of that of PVDF and measure the dynamic behaviors of PBLG-PMMA bodies in order to investigate the feasibility to apply this composite to acoustic transducers. In order to improve the piezoelectricity, the PBLG concentration and the electric field intensity applied during fabrication processes are increased. Our efforts result in making the piezoelectric coefficient about three times higher than it in previous study. We also measured the dynamic behaviors of a PBLG-PMMA disk. Then, a Tonpilz transducer was designed, fabricated using PBLG-PMMA composite disks, and tested experimentally. [Work supported by Grant No. ADDUD080004DD.]

5:00

4pEAb9. High Curie temperature relaxor piezoelectric single crystal broadband transducers. Jindong Xia (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing 100190, China, xiajindong@tom.com), Junbao Li (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing 100190, China), Guisheng Xu (Shanghai Institute of Ceramics, Chinese Academy of Sciences, No. 215, Chengbei Road, Jiading District, Shanghai 201800), Jianxin Xing, Gaolei Zhao, and Zhe Chen (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing 100190, China)

Lead indium niobate-lead magnesium niobate-lead titanate (PIMNT) crystals have been used to fabricate longitudinal vibrator broadband tonpilz transducers as active piezoelectric material. Compared to lead magnesium niobate-lead titanate (PMNT), rhombohedral-to-tetragonal phase transition temperature (Trt) of PIMNT increases to higher than 120° C and coercive field (Ec) can reach 5kV/cm, while possessing the similar electromechanical properties. The PIMNT single crystal transducer can work under high temperature exceeding 80° C and large drive voltage over 0.6MVpp/m with a

DC bias field of 0.27MV/m.Simultaneously, the device shows a smaller size, as much as 10dB higher transmitting voltage response and 8dB higher receiving voltage response than lead zirconate titanate (PZT) longtitudinal vibrator(tonpliz) transducer with a broad bandwidth over 150%. The PIMNT device demonstrates tremendous performance for sonar and ultrasonic technology.

5:20

4pEAb10. The study on vibratory tactile sensor using piezoelectric bimorph resonator. Subaru Kudo (Department of Information Technology and Electronics, Faculty of Science and Engineering, Ishinomaki Senshu University, 1 ShinnMito Minamisakai, Ishinomaki-shi, Miyagi 986-8580, Japan, kudou@isenshu-u.ac.jp)

Piezoelectric vibratory tactile sensors are used for measuring the softness and hardness of an object. In this work, a new construction of vibratory tactile sensor was investigated using a piezoelectric bimorph resonator. The bimorph resonator was driven in one side by constant voltage and constant driving frequency. When the tip of a resonator was contacted with a test piece, the output voltage in the other side of resonator was changed by contact impedance. Then, the contact impedances were calculated with experimental characteristics of output voltage using the equivalent circuit of the bimorph resonator. It was experimentally clarified that the electrical contact capacitance gradually decreased as the load added to test piece increased. The amount of decrease for hard test piece was larger than those of the soft test pieces. Next, the Young's moduli of test pieces were calculated from the experimental results in this method. The calculated values were compared with actual material constants of test pieces. It was examined that the possibility for detecting the softness and hardness of an object by this method on bimorph resonator. These results in this study may be useful for designing on the piezoelectric tactile sensor.

5:40

4pEAb11. Theoretical approach on SAW characteristics of layered structures for gas sensing. Xiao Xie (Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, xiexiao08@mails.gucas.ac.cn), Wen Wang, Minghua Liu, and Shitang He

Integrating the self-assemble sensing film with SAW devices has advantage for organic gas detection. It generally requires a thin metal layer like Au acting as a catalytic base for the sensing film formation. This leads to a double layer SAW device sensing area fabrication consisting of a metal layer deposited on the piezoelectric substrate. Therefore, the SAW characteristics will be affected when propagating through the sensing area. Various layer parameters such as stiffness constants and the thickness are needed for consideration. In this paper, the SAW phase velocities have been calculated theoretically for ST-X Quartz/Gold structure under different normalized thickness configurations.

6:00

4pEAb12. Deposition of ZnO films with c-axes lying in R-sapphire substrate planes. Yan Wang, Shu-yi Zhang (Laboratory of Modern Acoustics, Institute of Acoustics, Nanjing University, Nanjing 210093, China, wangyan8008@126.com), Kiyotaka Wasa (Department of Micro-Engineering, Kyoto University, Kyoto 606-8501, Japan), and Xiu-ji Shui (Laboratory of Modern Acoustics, Institute of Acoustics, Nanjing University, Nanjing 210093, China)

The (110) textured ZnO films with c-axes lying in R-sapphire substrate planes are deposited by RF magnetron sputtering. The focusing investigation is the effect of substrate position in the sputtering on structural and acoustic properties of the ZnO films on the sapphire. The crystallographic characteristics of the films are characterized by X-ray diffraction (XRD) analysis. It is found that the crystalline orientation of the ZnO film varies with the substrate position deflected from the center of the target. The XRD spectra of the films show that there is an optimized substrate position, at which the strongest (110) peak of the ZnO film can be obtained. However, with the decrease or further increase of the deflection, the (110) diffraction intensity decreases and additional diffraction peaks of (002), (102) and (103) emerge. In order to investigate the variations of acoustic properties of these films, the ZnO piezoelectric films are deposited on R-sapphire substrates with different deflection positions, and the bilayered structures are used to fabricate film bulk acoustic resonators (FBARs). The results show that the electromechanical coupling coefficients k_{15} of the shear mode acoustic waves excited by FBARs also vary with the different positions similar to that shown by the XRD spectra.

6:20

4pEAb13. Optimal design of surface wave emats for enhancing their ultrasonic signal strength. Lei Kang, Shujuan Wang, Zhichao Li, and Guofu Zhai (Harbin Institute of Technology, P.O. Box 404, No. 92, West Da-Zhi Street, Harbin, Heilongjiang 150001, P.R. China, victorkang11@126.com)

The strength of the ultrasonic signal transmitted by electromagnetic acoustic transducers (EMATs) is very weak, which severely confines the further expansion of the application scope of EMATs. To solve this problem, the features of the transmission process of a surface wave EMAT are studied based on a 3-D model. Simulation reveals that performances of the transducer are significantly affected by its parameters. Aiming at enhancing the strength of the ultrasonic signal, this paper investigates the influence of EMAT parameters on the ultrasound field of the transducer and accomplishes the optimal design of the EMAT by utilizing orthogonal test method. Experiments indicate that after optimization, the received signal of the EMAT has increased by 162.3%, which suggests the ultrasonic signal of EMATs can be effectively enhanced by utilizing orthogonal test method based on the 3-D EMAT model. Acknoweledgement: Supported by "the Fundamental Research Funds for Central Universities" (Grant No. HIT. NSRIF. 2012008)

Session 4pHT

Hot Topics: Aeroacoustics II

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Contributed Papers

2:00

4pHT1. A circular microphone array design approach for discrete noise suppression. Bo Yang, Jie Feng, and Ming Wen (The Third Research Institute of China Electronics Technology Group Corporation, Chaoyang District, Beijing 100015, China, byang@yahoo.cn)

A broadband detection performance of a circular microphone array has been degenerated by discrete noise interference. In this paper, a circular microphone array optimum design approach is proposed for discrete noise suppression in acoustic detection in the air. This approach utilizes the noise information of acoustic propagation's transfer function to mitigate its effect on broadband microphone array detection processing. It is demonstrated to be feasible to improve passive broadband detection ability of the circular microphone array through the adjustment of array shading weights. This is accomplished by numerically maximizing the deflection coefficient under the assumption of a small signal-to-noise ratio. Under this approach, the conventional beamformer is not redesigned, and only the shading weights of the conventional beamformer are adjusted.

2:20

4pHT2. Noise generation by a cylindrical cavity in subsonic flow. Olivier Marsden, Christophe Bogey, and Christophe Bailly (Laboratoire de M'ecanique des Fluides et d'Acoustique Ecole Centrale de Lyon & UMR CNRS 5509 69134 Ecully Cedex, France, olivier.marsden@ec-lyon.fr)

Following a previous experimental study of the noise generated by cylindrical cavities placed in subsonic flows, this work investigates mechanisms involved in the noise generation by a cylindrical cavity placed in a turbulent boundary layer, via numerical simulation. The cavity has a radius of r =5 cm and a depth of 10 cm. Flows are computed at Mach numbers of 0.2, 0.26 and 0.32, and the numerical boundary layer, of thickness 17 mm, is perturbed upstream of the cavity. Simulations are performed by solving the unsteady compressible Navier-Stokes equations using low-dispersion and low-dissipation finite-difference schemes. The LES approach is based on the explicit application of a selective filtering to the flow variables to take into account the dissipative effects of the subgrid scales. Numerical results are compared to the available experimental data presented in previous work, including mean flow aspects, flow statistics and acoustics. The numerical results are then examined in more detail in particular with regard to the origin of the well-defined tones observed in the acoustic far field. A simple feedback model coupling shear layer dynamics with depth mode acoustic resonance is shown to yield good frequency predictions.

2:40

4pHT3. A new acoustic target recognition method based on gaussian mixture model. Ming Wen, Jie Feng, and Bo Yang (The Third Research Institute of China Electronics Technology Group Corporation, Chaoyang District, Beijing 100015, China, wenming.wm@gmail.com)

Target recognition is one of the most important components of an acoustic detection system. Conventional acoustic target recognition algorithms base the classification results on features from a single frame of signal, suffering from inability to take the dynamic characteristics of acoustic signal into account, which can significantly increase the separability of acoustic targets. In this paper, we present a novel method for acoustic target recognition. Acoustic signal to be classified is divided into a sequence of signal segments. Gaussian mixture model is employed to model the sequence, and EM algorithm is utilized to learn the parameters. This idea is motivated by the interpretation that the Gaussian components can represent some general target-dependent spectral shapes. Experimental results show that the method achieves high recognition accuracy on different data sets, and is highly robust to ambient noise.

3:00

4pHT4. Aeroacoustics of flow merging at duct junction. Garret C. Y. Lam, S. K. Tang (Department of Building Services Engineering, The Hong Kong Polytechnic University Hung Hom, Kowloon, Hong Kong, P.R. China, garret.lam.hk@connect.polyu.hk), and Randolph C. K. Leung (Department of Mechanical Engineering, The Hong Kong Polytechnic University Hung Hom, Kowloon, Hong Kong, P.R. China)

Although merging flow at duct junction is always encountered in fluid-transporting systems, previous works were mainly devoted to study the acoustics of duct junction but the aeroacoustics of flow merging has received little attention. Therefore, this paper aims at revealing the mechanism of sound generation by a merging flow at a 30-degree duct junction. The flow problem is investigated numerically by solving the unsteady compressible Navier-Stokes equations and the gas equation of state simultaneously, thus allowing the acoustic field and the aerodynamic field to be determined without modelling the source terms in the wave equation. The Conservation Element and Solution Element (CE/SE) method is chosen as the solver, which has been successfully applied in tackling many aeroacoustic problems. The results show that a shear layer is created between the two flows at duct junction due to the velocity gradient across the flow. Furthermore, another shear layer is also formed at downstream to the edge of duct junction. However, only the latter one rolls up to form vortices, which are believed to be the acoustic source. The contributions of these flow dynamical processes to sound generation and the identification of dominant sources will be discussed.

3:20

4pHT5. Numerical research on the flow characteristics of confined sonic and supersonic air-modulated speakers. Zhao Yun, Zeng Xinwu, and Gong Changchao (Institute of Optical-Electronic Science and Engineering, National University of Defense Technology, Changsha 410073, P.R. China, sangelboy@sohu.com)

Air-modulated speaker is one of the most famous high-intensity sources. Sound waves with SPL above 160 dB are generated by the modulation of sonic air flow through a time-varying valve. To overcome the source saturation at high chamber pressures, the performance and flow characteristics of a new design, in which the flow speed increases from sonic to supersonic by a converging and diverging nozzle, were investigated. Based on the

analytical results from the widely used quasi-steady model, the source level improvement was verified for the same pneumatic power. The SPL increment of more than 5 dB is obtained at Mach number 2.75. The transient flow inside the source was simulated using a recent developed numerical model in which the compressible flow is governed by the Navier-Stokes equation and the valve movement is modeled by the dynamic mesh method. Compared with the ordinary sonic flow modulation, the source mechanism on the supersonic case has shown several distinct features: the effect of the flow-acoustic coupling could not be neglected; some complex phenomena such as shock wave formation and propagation arise in the transient flow inside the nozzle; the establishment of supersonic flow and the resulting improvement are related to the valve modulation frequency.

3:40

4pHT6. On the adjoint problem in duct acoustics and its solution by the time domain wave packet method. Fang Hu, Ibrahim Kocaogul (Old Dominion University, Norfolk, VA 23529, fhu@odu.edu), and Xiaodong Li (Beihang University, Beijing 100191, China)

The Time Domain Wave Packet (TDWP) method has the advantage of obtaining radiated sound by a given duct mode at all frequencies in one computation. It also makes possible the separation of the acoustic and shear flow instability waves. In this paper, the TDWP method will be applied to the adjoint duct acoustics problem. As a microphone records the sound radiated by all duct modes combined, the adjoint approach has the advantage of obtaining the relative mode strengths of all the propagating modes at all frequencies in one computation. The novelty of the paper is on the formulation of adjoint equations in time domain and the numerical solution using the TDWP method. The theoretical formulation of the duct adjoint problem and derivation of various reciprocal relations are presented. The adjoint equations are then solved numerically in reversed time by the TDWP method, in which a point source in the far field is enforced with a Broadband Acoustic Test Pulse time function. PML absorbing boundary conditions for the adjoint equations are also discussed. By an FFT, the relative strengths of all duct modes at all frequencies to the far field point are computed at once.

4:00–4:20 Break

4:20

4pHT7. Preliminary analysis of acoustic measurement from ARJ21 flyover test. Zhifei Chen, Jianhua Yang, and Hong Hou (Northwestern Polytechnical University, 127 West Youyi Road, Xi'an, Shaanxi, China, chenzhifei@gmail.com)

A flight test was performed on the airport of Yanliang in June, 2010 for the noise source identiofication of ARJ21. The airframe noise was recorded using a ground-based multi-arm logarithmic spiral array with diameter as 30 meters. A data set of five taking off and eight approaching flight with extended landing gears has been acquired during four days. Conventional beamforming and deconvolution approach for the mapping of acoustic sources (DAMAS) are applied for the data analysis. The beamforming results could clearly distinguish two engines and the nose landing gear. Their spectrum characterizations are also presented together with a description of the experimental set-up.

4:40

4pHT8. A nodal discontinuous Galerkin method for computational aeroacoustics and comparison with finite difference schemes. Eryun Chen, Ailing Yang, and Gaiping Zhao (USST, cheneryun@usst.edu.cn)

A nodal discontinuous Galerkin formulation, which is based on Lagrange polynomials basis, is used to directly simulate the acoustic wave propagation. Two test problems of wave propagation with initial disturbance consisting of a Gaussian profile or rectangular pulse are investigated. We evaluate the performance of the schemes in short, intermediate, and long waves. Moreover, the comparisons of numerical results between the nodal discontinuous Galerkin method and finite difference type schemes are performed, which indicate that the numerical solution obtained using nodal discontinuous Galerkin method with a pure central flux has obvious high frequency oscillations for initial disturbance consisting of rectangular pulse, which is the same as those obtained using finite difference type schemes without artificial selective damping. If an upwind flux is adopted, spurious waves are eliminated effectively except for the location of discontinuities. When a limiter is used, obviously the spurious short waves are almost completely removed. The quality of the computed solution has greatly improved.

5:00

4pHT9. The enhancement of pulverized-coal combustion by using sound waves. Genshan Jiang, Yingchao Zheng (Department of Mathematics and Physics, North China Electric Power University, Baoding, Hebei Province 071003, P.R. China, gs_jiang@hotmail.com), Jie Pan (School of Mechanical Engineering, The University of Western Australia, Crawley, WA 6009, Australia), and Jing Tian (Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, P.R. China)

A model for the enhancement of pulverized coal combustion in high-intensity acoustic fields has been developed. In a power boiler pulverized coal particles and char are entrained in the main gas flow for a significant length of time. The slip velocity between the entrained particles and the gas is very low leading to low heat and mass transfer to and from the particles. This results in long combustion times for the particles, particularly if the combustion is diffusionally controlled. When a high-intensity acoustic field is applied to the gas flow, it superimposes high-velocity oscillations on the main flow. If the frequency and amplitude of this acoustic field is just right, the particles are not entrained in the acoustically induced flow oscillations. This creates a periodically varying slip velocity between the particles and the hot gas, leading to higher convective heat and mass transfer to and from the particles, thereby enhancing the combustion. The paper also presents the axial pressure gradient and shear stress in neighborhood of the particles placed in sound field.

5:20

4pHT10. CAA-based optimisation of a sensor array to characterize aft and forward fan noise in a 3D nacelle. D.-C. Mincu (Onera - Office National d'Etudes et de Recherches Aerospatiales, BP 72, 92322, Chatillon, France, daniel-ciprian.mincu@onera.fr), E. Manoha, and C. Polacsek

In the context of the on-going European project JTI-SFWA, the modal acoustic content inside a realistic fan duct, designed by Dassault-Aviation, must be characterized experimentally. The objective is to use Onera's CAA solver sAbrinA-V0 to design a non-intrusive sensor distribution inside the nacelle, upstream/downstream the fan plane, accounting for the heterogeneous mean flow and the complex internal geometry. By computing the propagation of several interacting modes in the duct, a region in which the acoustic pattern is close to the infinite duct analytical solution was characterized. Then several 3D distributions of pressure sensors were tested in this zone. Radial and azimuthal sensor arrays were located at the nacelle wall, whereas a radial sensor array was placed along the bifurcation leading edge, without additional intrusivity. The response of the 3D arrays to a given acoustic field, either analytical or propagated through CAA, was projected on an analytical modal basis. For several array configurations, this projection clearly showed the emergence of the modal injected content. From these simulations, the most efficient sensor distribution will be used to equip an experimental fan noise simulator, and the modal projection process will be applied on measurements to evaluate the actual modal content in the duct.

5:40

4pHT11. Trailing edge noise mitigation investigation for wind turbine blades. Michael J. Asheim, Dave Munoz (Colorado School of Mines, Golden, CO 80401, U.S.A., masheim@mines.edu), and Patrick Moriarty (NREL-NWTC, Golden, CO 80401, U.S.A.)

Wind turbines offer one of the most mature technologies for providing large scale renewable energy to society in an economically viable way. Although not on par with the price of conventional energy sources yet, the cost of energy has been steadily decreasing as the technology continues to develop. Unfortunately, like with all energy sources, there are some problems with this form of generation. Among these, sound emissions from wind turbines are one of the problems people who live close to the installed machines may be exposed to. Past studies show that these noise emissions are dominated by aeroacoustic noise and of the many mechanisms that lead

to aeroacoustic noise, the interaction between the unsteady flow and the trailing edge seems to constitute the largest portion of the overall sound spectrum. Modifications to the trailing edge geometry will change how the fluid interacts with the trailing edge and can be used to change the resulting noise emission. This study will investigate the effect passive trailing edge devices have on the overall noise emission from a wind turbine, in an attempt to reduce the aeroacoustic noise being generated by the turbine.

6:00

4pHT12. Aerodynamic noise prediction model of pantograph for high-speed train. Sukkeun Yi and Junhong Park (Hanyang University, 133-791, sarkkun@hanyang.ac.kr)

In this study, to analyze aerodynamic noise from the pantograph on high-speed train, numerical model was established. We study the mechanism of the noise generation by setting up the simple model and finally analyze entire pantograph model. Based on the Lattice Boltzmann Method, computational fluid analysis was carried out. Through noise analysis for near field, sound radiation for far field was estimated using Ffowcs Williams and Hawkings Equation. Simple model is the square and circular cross-sectional cylinder. This model makes noise by flow separation accompanied by vortex shedding. Considering pantograph have a series of the rods, we made a complex structure which have square and circular cross-sectional cylinder. By analyzing entire pantograph model we suggest the model for predicting the noise generation and radiation.

6:20

4pHT13. Prediction of aerodynamic sound radiation from pantograph of high-speed train. Sukkeun Yi and Junhong Park (Hanyang University, 133-791, sarkkun@hanyang.ac.kr)

Based on the Lattice Boltzmann method, aerodynamic noise from the pantograph of high-speed trains was investigated. The mechanism of the noise generation was analyzed using a simple panhead model, and the numerical procedures were extended to analyze sound radiation from the whole pantograph system. The square and circular cross-sectional cylinders were used to simulate the panheads. The primary noise was generated by flow separation accompanied by vortex shedding. Since the pantograph were consisted of a series of the rods, the primary noise generation characteristics were similar to the simple panhead. From the numerical model, the design of the pantograph for reduced aerodynamic noise generation is proposed and validated.

THURSDAY AFTERNOON, 17 MAY 2012 — S228, 2:00 P.M. TO 3:00 P.M.

Session 4pMUa

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

James W. Beauchamp, Cochair
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Contributed Papers

2:00

4pMUa1. Spatial variability of timbre for an electric guitar amplifier. Alexander Case (University of Massachusetts Lowell, 35 Wilder St., Lowell, MA 01854, alex_case@uml.edu), Agnieszka Roginska, and Jim Anderson (New York University, New York, NY 10012)

Impulse response measurements of an electric guitar amplifier at high spatial resolution reveal frequency response variabilities likely to drive key elements of timbre for the recording engineer. Data collection from the very near field of the driver to a distance of several feet away, gathered in a three-dimensional grid around the open-backed, single-driver, Fender Deluxe electric guitar amplifier quantify the variations in frequency response which may be exploited by the recording engineer using microphone placement to influence timbre.

2:20

4pMUa2. Synthesis of performance expression of bowed string instruments using "Expression Mark Functions". Yuma Koizumi and Katunobu Itou (Hosei University, 3-7-2 Kajino-cho Koganei, Tokyo, Japan, 08k1014@cis.k.hosei.ac.jp)

This paper proposes a method for synthesis of performance expression of bowed string instruments. In order to reflect a creator's intention on a synthetic sound, physical model is efficient because the sound can be synthesized by the performance imagination. The physics of the bowed string instruments are still some issues remain unresolved, bowed string sound synthesis using only the physical model is difficult. In this paper, a method using a transfer function for expression mark is proposed. "Expression mark functions" is estimated from recorded performance sounds, using spectrum of string motion and inverse filter of resonant properties of the instruments. Bowed String motion is a triangular wave called the Helmholtz wave, and it can be determined from bowed string position. Changes of waveform from expression performance were estimated by non-negative matrix factorization. Resonant properties of an instrument were measured as TSP response using a "direct conduction speaker". The expression mark functions of the performance is extracted by decomposing the actual performance. For evaluation of the quality of the synthesized sound, the scale by ten kinds of expression marks was synthesized using expression mark functions, and the questionnaire estimated the degree of agreement in five steps.

2:40

4pMUa3. Data collection for individuality analysis on subjective music similarity evaluation. Shota Kawabuchi, Chiyomi Miyajima, Norihide Kitaoka, and Kazuya Takeda (Nagoya University, Furo-cho Chikusa-ku, Nagoya 464-8603, Japan, shota.kawabuchi@g.sp.m.is.nagoya-u.ac.jp)

Recently there are many studies of subjective music similarity for music information retrieval. To quantify the subjective music similarity, there are many factors to take into account. In this study the individuality of the subjective similarity was focused on. To analyze the individuality of the subjective similarity, subjective evaluation data for 200 pairs of RWC popular music database was collected. 28 subjects listened to the pairs of tracks, and evaluated similarity of each pair by similar or dissimilar. They also selected the components of music (melody, tempo/rhythm, vocal, instruments) that was similar. Each subject evaluated the same 200 pairs, thus the individuality of the evaluation can be easily analyzed. The results of pilot analysis were reported.

Session 4pMUb

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

Thomas Moore, Chair
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Contributed Papers

3:20

4pMUb1. Open-loop control of a robotized artificial mouth for brass instruments. Thomas Helie (IRCAM- CNRS UMR 9912, thomas.helie@ircam.fr), Nicolas Lopes, and René Causse (IRCAM - CNRS UMR 9912)

In order to have reproducible and controllable experiments, a robotized artificial mouth dedicated to brass instruments has been developed (CONSONNES project of the French National Research Agency). The actuators control: (A1) the airflow, (A2) the mouth position (monitoring the lips force applied to the mouthpiece), (A3-4) the water volume in each lip (water-filled latex chamber). The sensors measure: (S1-2) the pressure in the mouth and the mouthpiece, (S3: optical sensor) the area between the lips, (S4) the force lips/mouthpiece, (S5-6) the water pressure in each lip, and (S2bis-S4bis) the position of the moving coils (A2-4). We present an open-loop control obtained from measures, according to the following steps. First, a calibration for the lips control is performed, by analyzing signals (S4-6) w.r.t. positions (S2bis-S4bis), with no airflow. Second, slowly time-varying calibrated commands (A2-A4) are used to obtain quasi-stationary regimes (non oscillating, quasi-periodic, etc), for constant airflows. Third, a sound analysis is processed to generate cartographies of features (energy, fundamental frequency, etc) w.r.t. to the calibrated control parameters. This exploratory tool allows to build a dictionary of relevant control parameters from which basic sequences of notes can be played, and during which all the signals (S1-6) can be recorded.

3:40

4pMUb2. Minimum models for self-sustained oscillations of conical reed instruments. Jean Kergomard (LMA-CNRS 31 Chemin J. Aiguier, F-13402 Marseille Cedex 20, kergomard@lma.cnrs-mrs.fr), Philippe Guillemain, and Fabrice Silva (F-13402 Marseille Cedex 20)

It is now well known that a minimum model for self-sustained oscillations of clarinet-like instruments is the iterated map model, leading to square signals. The reed is assumed to be without dynamics, while losses are ignored (or assumed to be independent of frequency). The generalization to conical instruments is not straightforward. For the present work, the minimum model is used for a truncated cone instrument, but the missing part of the cone is not assumed to be small compared to the wavelength. Thus the result should be a signal without sharp corners. However, without any kind of mouthpiece, no periodic sound can be obtained in a steady-state regime. It will be explained that the choice of a model for the mouthpiece can be done without adding any supplementary parameter (therefore a conical resonator has one parameter only more than a cylindrical one). It is shown that several choices are possible, allowing either the use of: the same inverse nonlinear characteristic than for clarinet-like instruments, or any direct nonlinear characteristic, leading to a great simplicity. Advantages and drawbacks of several solutions are discussed.

4:00–4:20 Break

4:20

4pMUb3. Investigation of optimum shape for an acoustic guitar by electroanalysys of tone quality. Kazutaka Itako (Kanagawa Institute of Technology, 1030 Simo-Ogino, Atsugi, Kanagawa, Japan, itako@ele.kanagawa-it.ac.jp), and Satoshi Itako (B.K. Guitar Craft Center, 217-77 Mibutei, Mibumachi, Shimotugagun, Tochgi, Japan)

The shapes of acoustic guitars are strongly governed by the sensibilities of the craftsmen who make them, and thus, shapes vary widely. Unlike violins and other instruments, no ideal shape has yet been established for guitars. In our research, we conducted an electroanalysis of tone quality with the aim of manufacturing a guitar with enhanced tone quality. In this study, we examined the effects of varying guitar body thickness and chamber cubic volume on tone quality with various playing methods, and quantitatively identified the optimal thickness.

4:40

4pMUb4. Simulation of 'Suikinkutsu' sound considering sound radiation from water surface. Yuki Fujita, Naoto Wakatsuki, and Koichi Mizutani (University of Tsukuba, Tsukuba, Japan, fujita-y@aclab.esys.tsukuba.ac.jp)

This paper is about a 'Suikinkutsu', a kind of Japanese traditional musical instruments. Different from other musical instruments, 'Suikinkutsu' is composed of an upended jar buried underground in the garden. Dropping water into 'Suikinkutsu' works as sound source, and attracting harp-like sound inside the jar, 'Suikin-on', is generated. Although the sounding mechanism of 'Suikinkutsu' is interesting because there are complicated interactions among air, water, and jar made of ceramic to generate harp-like beautiful sound, its analysis has not been achieved yet. To investigate the sounding mechanism of 'Suikinkutsu', simulation of interaction between air and water is performed. The obtained results suggest that the water in the jar contributes to sound generation of 'Suikinkutsu'.

5:00

4pMUb5. Strike notes and their pitches of some kinds of bells. Shigeru Yoshikawa (Graduate School of Design, Kyushu University, 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, shig@design.kyushu-u.ac.jp), Wang-Ho Cho, and Takeshi Toi (Department of Precision Mechanics, Chuo University, 1-13-27 Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan)

The strike notes of western church bells have been studied very long. This paper considers the bells that are different from church bells: (1) simple tubular bells called "wind chime", (2) a downward-stretching tubular bell with hyperbolic surface, which the authors name "Gaudi bell" because Antoni Gaudi envisaged music of such bells coming from the belfries of the Sagrada Familia Church, and (3) a double Gaudi bell stretching downward and upward from the center with different lengths. Strike notes of these bells are yielded from the axial vibrations. However, their pitches do not necessarily correspond to the strongest spectrum (mode) of the bell vibrations. Aluminum tubular bells of our wind chime gave the "spectral pitch" determined by the spectral frequencies (around 1 kHz) of the fourth mode. On the other hand, our Gaudi bell (brass-made, 1.80 m, 50 kg) yielded the fourth mode at 750 Hz. However, many listeners perceived the pitch at 375 Hz. This is the "virtual pitch" based

on the missing fundamental (mode frequencies higher than the 4th were 1082, 1450, 1842, and 2258 Hz). A double Gaudi bell (bronze-made, 0.342 + 0.260 m) gave the "spectral pitch" at 1125 and 2046 Hz.

5:20

4pMUb6. Gabor domain analysis of mambranophones. Robert Ferguson (Department of Geoscience, University of Calgary, 2500 University Drive NW, Calgary, Alberta, Canada, rjfergus@ucalgary.ca)

Advanced analysis methods of exploration seismology are applied to the study of a class of membranophones that are well modelled as three spring damped oscillators. As a first attempt, damping is ignored in this treatment. Analysis using vibroseis-like source sweeps, cross-correlation, Fourier decomposition, and Gabor domain analysis provide insight into how this class is tuned for use in jazz and popular music. It is found that for a particularly good set of three example membranophones, they are tuned to pitches such that they resonate in combinations that are particularly harmonious. Combinations of octave, perfect fourth, and perfect fifth are found and, in particular, the Gabor domain is most useful for discrimination of resonant tones from the ambient noise of the recording system and surroundings.

5:40

4pMUb7. Simulation of 'Suikinkutsu' sound considering sound radiation from water surface. Yuki Fujita, Naoto Wakatsuki, and Koichi Mizutani (Univ. of Tsukuba, fujita-y@aclab.esys.tsukuba.ac.jp)

This paper is about a 'Suikinkutsu', a kind of Japanese traditional musical instruments. Different from other musical instruments, 'Suikinkutsu' is composed of an upended jar buried underground in the garden. Dropping water into 'Suikinkutsu' works as sound source, and attracting harp-like sound inside the jar, 'Suikin-on', is generated. Although the sounding mechanism of 'Suikinkutsu' is interesting because there are complicated interactions among air, water, and jar made of ceramic to generate harp-like beautiful sound, its analysis has not been achieved yet. To investigate the sounding mechanism of 'Suikinkutsu', simulation of interaction between air and water is performed. The obtained results suggest that the water in the jar contributes to sound generation of 'Suikinkutsu'.

THURSDAY AFTERNOON, 17 MAY 2012 — HALL B, 2:00 P.M. TO 5:40 P.M.

Session 4pNSa

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

Siu Kit Lau, Cochair
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Contributed Papers

2:00

4pNSa1. Active noise control in the exhaust port of a vacuum cleaner. Ping Wang, Jiancheng Tao, and Xiaojun Qiu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, wangpingfairy@163.com)

Exhaust port is one of the main sound transmission paths for noise radiation of a vacuum cleaner. Porous sound absorption materials are usually stuck at the inner surface of the exhaust port to absorb the noise; however the noise reduction performance, especially at low frequency, usually is limited due to the limited transmission path length to the exhaust port for airflow and the low acoustical absorption coefficients of porous materials at low frequency. In order to reduce the low-frequency noise which radiates outside through the exhaust port, active noise control technologies are applied in exhaust port, where the configurations of the active control units including the reference sensor, the control loudspeaker and the error microphone are investigated. Finally, experiments are carried out for performance validation.

2:20

4pNSa2. Active control on the scattered radiation by a rigid surface. Ning Han and Xiaojun Qiu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, hanning@nju.edu.cn)

An approach for predicting the scattered sound pressure in one- and three-dimensional sound field is proposed by measuring the total sound pressure on a rigid scatterer surface, which is further used as an error sensing strategy in an active noise control (ANC) system to reduce the scattered radiation. Experiments are carried out to validate the prediction method, and a single channel broadband feedforward ANC system is implemented to suppress the impulsive scatterance at the observation point. It is found that the ANC system based on the proposed prediction method is effective. About 12.2 dB reduction of the impulsive scattered radiation is obtained in one-dimensional sound field, and 8.2 dB reduction in three-dimensional sound field. ACKNOWLEDGMENTS The authors would like to acknowledge the support NSFC (Project 11104141).

4p THU. PM

2:40

4pNSa3. Case study of applying active noise control on communication chassis. Haishan Zou, Di Yu, and Xiaojun Qiu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, hszou@nju.edu.cn)

In some communication chassis, running of the cooling fan causes the noise emission level of the chassis exceed the upper limited level of the standards and regulations. In this case study, various sound transmission paths are indentified, such as the structure borne sound transmission, the air intake and the air outtake, etc. To reduce the direct noise, a metal enclosure which divided into several ducts and with new intake and outtake on its surface is used to cover the chassis. The wall of ducts is equipped with absorbing material to form mufflers, so the middle and high frequency noise is reduced successfully. The key point is that the active control systems are installed in the ducts to reduce the low frequency noise. A noise level reduction of 6-8 dB is achieved with this active-passive hybrid method. However, it is found that it is impossible to achieve such a reduction level without using active noise control because of the abundant low frequency component of the noise.

3:00

4pNSa4. A study on active noise control using multi-pole control sources. Hiroyuki Ichikawa (Kyushu University, 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, ichikawa.hiroyuki.365@s.kyushu-u.ac.jp)

In Active Noise Control (ANC), the difficulty of global control is one of the issues to be tackled. Basically, a lot of control sources and corresponding sensors are needed to realize effective control in large area. In the present study, a new type of secondary source arrangement, the multi-pole source array, is proposed. The idea is based on the Kempton's approach using multi-pole expansion, which was proposed in 1970's. This method is expected to have a capability of realizing more flexible shapes of wavefront compared with conventional control source arrays. The effect of proposed array sources is examined in the free field and in the diffracted field with thin semi-infinite barrier. The results of the numerical simulations and the preliminary experiment indicate the superiority of the proposed method.

3:20

4pNSa5. A robust acoustic feedback suppression algorithm based on spectral subtraction. Feiran Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, feirany.ioa@gmail.com), Ming Wu, Peifeng Ji, and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China)

Acoustic feedback is caused by the coupling between the loudspeaker and the microphone. In this paper a new acoustic feedback suppression (AFS) method is proposed to address this problem. The cross-spectrum of the loudspeaker and the microphone signal and power spectrum of the loudspeaker signal are estimated by the recursive averaging of periodograms. Then, the magnitude spectrum of the acoustic feedback path is obtained by dividing the cross-spectrum by the power spectrum of the input signal. The spectrum of the acoustic feedback signal is calculated using the above magnitude spectrum. Moreover, the spectral modification technique originally proposed for speech enhancement is adopted to remove the acoustic feedback from the microphone signal. The proposed algorithm is less sensitive to the acoustic feedback path changes than the existing adaptive filter approach. Simulation results demonstrate the effectiveness and the robustness of the new method.

3:40

4pNSa6. Assessment of the effectiveness of adaptive active vibration control for the minimisation of radiated sound from panels. Robin Wareing and John Pearse (University of Canterbury, 69 Creyke Road, Ilam Christchurch, robin.wareing@pg.canterbury.ac.nz)

In many practical situations noise is radiated from a noise or vibration source into adjacent areas through vibration of plates. This paper is concerned with the reduction of this radiated noise via active vibration control of such panels. An LMS based adaptive controller is implemented and the experimental results are compared to an ideal model of the noise reduction, this allows the effectiveness of the adaptive controller to be assessed. The plate used in tests is a 1546mm by 946mm simply supported panel. This has been modelled in Matlab allowing the sound field above the plate to be evaluated. Using the theories developed by Nelson et al [1] the theoretical optimal noise reduction can then be calculated. The panel is excited via a point force; the sound pressure at a number of points is then measured. The active control is actuated via a coil based inertial actuator. The sound pressure is measured following the implementation of active control, thus allowing the practical noise reduction to be calculated. 1. Fuller, C.R., S.J. Elliott, and P.A. nelson, Active Control of Vibration. Vol. 1. 1996, London: Academic Press Limited. 332.

4:00–4:20 Break

4:20

4pNSa7. A novel decentralized velocity feedback strategy of plate vibrations using piezoelectric patch actuators. Yin Cao, Hongling Sun, Xiaodong Li, and Jing Tian (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, caoyin@mail.ioa.ac.cn)

Previous studies have shown that, similar feedback gains are required to minimize the kinetic energy of a plate and to maximize the absorbed energy by each force actuator in a wide-band excitation. The energy absorption of a force actuator is easy to measure, but it is hard to get the energy absorption by a moment actuator such as a piezoelectric patch actuator which needs the angular velocity at the mounting point of the actuator. In this paper, the energy absorption of piezoelectric patch actuators for decentralized velocity feedback control of plate vibrations is investigated numerically. A kind of virtual energy absorption of the piezoelectric patch actuator is proposed, which is the multiplication of the torque produced by the piezoelectric patch actuator and the velocity at the mounting point of the actuator. Numerical investigations are performed to explore the relationship of the virtual energy absorption by the piezoelectric patch actuators and the kinetic energy of the structure. The results show that maximizing the wide-band virtual energy absorption is nearly equivalent to minimizing the kinetic energy.

4:40

4pNSa8. Variable step-size μ-law memorised improved proportionate affine projection algorithm for sparse system identification. Longshuai Xiao and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, xls.ioa@gmail.com)

System identification has wide applications in adaptive echo/feedback cancellation, active noise control, adaptive channel equalization, etc. When the system is sparse, recently, a μ-law memorised improved proportionate affine projection algorithm (MMIPAPA) has been proposed to improve the misalignment performance remarkably. However, the MMIPAPA with constant step-size has the conflicting requirement of fast convergence rate and low steady-state error. To solve this problem, a variable step-size version of the MMIPAPA (VSS-MMIPAPA) has been extended by setting each component of the a posterior error energy vector equal the system noise energy. Furthermore, through an alternative method to compute the variable step-size, when the estimated component of the a prior error energy vector is smaller than the system noise energy, further lower steady-state misalignment is achieved. This method leads to an improved VSS-MMIPAPA (IVSS-MMIPAPA). The computational complexity of the IVSS-MMIPAPA is $O(P^2L)$, which may be too high for many applications. Therefore, an approximate algorithm with little performance degradation has been implemented using recursive filtering and dichotomous coordinate descent iteration techniques, by which the complexity has been reduced to $O(PL)$. Finally, the efficiency of both the exact and approximate algorithms developed here has been verified by simulation results.

5:00

4pNSa9. The Fortaleza noise mapping project—a tool for the definition of noise action plans for the airport located in the center of the municipality. Francisco Aurélio Chaves Brito (SER V and UNIFOR, aurelio.semam@hotmail.com), and Jose Luis Bento Coelho (CAPS, Instituto Superior Técnico, TULisbon, Lisboa, Portugal)

Studies of the impact of noise from Fortaleza International Airport, located well within the urban area, based on the noise mapping project that was created for the spatial representation of indicators of ambient noise in the city of Fortaleza, Brazil, which provided a tool essential to analyze and define strategies for the control of noise pollution in the city. This is the first noise map drawn to scale in a large city in Brazil. Noise emissions from major sources that contribute to the sound environment of the city, including road traffic, railway noise, aircraft noise, industrial noise, noise and entertainment areas were included. The method followed a hybrid approach, essentially calculating complemented with experimental measurements for validation and calibration. The study of airport noise allowed detailed results for equip lawsuit against the noise produced during operation of the airport.

5:20

4pNSa10. Building active noise control system decrease MRI driving sound. Shohei Nakayama (Shibaura Institute of Technology, f08091@shibaura-it.ac.jp)

Our purpose is to build Active Noise Control (ANC) system for MRI examination. It is problem for the patients that loud sound which is generates by Magnetic Resonance Imaging (MRI) equipment. Its maximum of the sound pressure level is over 100dB. It does not only make patients sickening, but has possibility of hearing loss. Patients' hearing ability is protected from the loud sound by the ear protectors. The ear protectors decrease around 20dB from the sound. Its performance is unsatisfactory to decrease the MRI driving sound for the patients. Because the patients hear the high level sound which the maximum sound pressure level is over 80dB if any wearing ear protectors. They hope for more the performance. We have an idea that we build the ear protectors whose ANC system controlling low frequency. It is important for the feed-forward ANC system to decide the reference point. We show that the result measured the MRI driving sound for finding the sound source. As a result, we found the sound source is located around the gradient coil. We show computer simulation result that ANC system whose reference point located on the sound source controls the MRI driving sound.

THURSDAY AFTERNOON, 17 MAY 2012 — HALL C, 2:00 P.M. TO 6:20 P.M.

Session 4pNSb

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

Brigitte Schulte-Fortkamp, Cochair
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K. C. Lam, Cochair
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Invited Papers

2:00

4pNSb1. Enhancement of soundscapes using natural sounds in urban spaces. Jooyoung Hong and Jin Yong Jeon (Hanyang University, Seongdong-gu, Seoul, st200045@hotmail.com)

Design elements of urban soundscape were investigated through individual soundwalk and laboratory experiments. Soundwalking was performed in Seoul; thirty subjects selected evaluation sites when they perceived positive and negative soundscape elements in the soundscape route. It was found that soundscape elements are closely related to visual elements. In order to investigate the effects of the design elements on soundscape perception, laboratory experiments were designed: birds' singing and water sounds were selected as acoustic stimuli enhancing the urban soundscapes. Trees and water features were also added as visual stimuli which were connected to the acoustic elements. The experiments consisted of three parts; 1) audio-only condition, 2) visual-only condition, and 3) audio-visual condition. As a result, the contributions of soundscape elements to urban soundscape perception were derived.

2:20

4pNSb2. Tranquility analysis by soundwalks in Pisa's green areas. Gaetano Licitra (CNR-IPCF and ARPAT Tuscany Environmental Protection Agency, g.licitra@arpat.toscana.it), Claudia Chiari (ARPAT, Tuscany environmental protection agency), Irene Menichini (University of Pisa), and Elena Ascari (CNR-IDASC, Institute of Acoustics "O.M. Corbino")

Identification and analysis of main quiet areas in Pisa agglomeration has been carried out by the University of Pisa and Tuscany Environmental Protection Agency (ARPAT). According to suggestions given by the Environmental Noise Directive (END) 2002/49/CE, authors measured noise levels (using different indicators) and perceived tranquility of users at four different park/green areas in the city (Monumental site – Miracle square, Botanic Garden, Riverside park "Le Piagge", City Park "Giardino Scotto"). In order to analyze tranquility, measurements and analysis on site has been carried out including noise levels acquisition, video recording, binaural audio acquisition and users surveys. At first some different methodologies has been tested and compared to identify most suitable measurement instruments and analysis procedures. The survey, based on similar studies carried out by G. Watts in UK, intended to identify most annoying sources and subjective users impression of the park. In the same sites, soundwalks have been recorded to allow a laboratory test to obtain a second evaluation of the park (video and binaural audio reproduction). First results highlighted the negative effects on objective and subjective evaluation of tranquility done by different type of transport (trains, cars, aircraft) in Pisa's parks.

2:40

4pNSb3. Revision of the new orleans noise code in regard to soundscape, enforcement, and cultural impact. David Woolworth (Oxford Acoustics, Inc. 356 CR 102 Oxford, MS 38655, dave@oxfordacoustics.com)

New Orleans is a city with districts of predominately residential historical structures that overlap with entertainment districts that play host to a tourist based economy, in addition to being the home of rich cultural traditions that include live music and Mardi Gras Carnival. As tourist and entertainment venues and their noise levels increase and expand, residents of mixed use areas are becoming less tolerant of the noise and are vacating the French Quarter that depends on their presence for the vitality of these districts. The complex layers and conflicting schedules of neighborhoods, entertainment zones, and frequent music festivals make a difficult case for enforcement. This paper outlines the methods for assessing the existing conditions and making the code revision to begin the process to strike a balance in the French Quarter and keep future development of the rest of the city in check.

3:00

4pNSb4. Validation of the Swedish soundscape quality protocol. Östen Axelsson, Mats E. Nilsson, and Birgitta Berglund (Department of Psychology, Stockholm University, SE-106 91 Stockholm, Sweden, oan@psychology.su.se)

The Swedish Soundscape-Quality Protocol was developed to help non-experts (e.g., officials working for municipalities rather than soundscape researchers) to make informed, accurate measurements of soundscape quality. The Protocol has hitherto been used in England, France, Italy, Spain, Sweden, and The Netherlands; a Korean version is being developed. Based on field studies – soundwalks in urban residential areas, recreational areas, and parks – the present paper reports on the psychometric properties of the scales of the Protocol. Participants were residents, or visitors to the areas and their results support the reliability and validity of the scales in the Protocol. Because high acoustic quality has a greater effect in visually attractive than in visually poor areas, the Swedish Soundscape-Quality Protocol includes scales for cross-sensory tabulation. These are sound source identification – sounds from humans, nature and technology – attribute scales (e.g., eventful, exciting, pleasant, and calm), overall soundscape quality, and concomitant visual impressions. In brief, the Swedish Soundscape-Quality Protocol is an easy to use and practical tool for measuring soundscape quality. It has the potential to help operationalize how soundscapes can be measured in “quiet areas” to meet a future guideline value of the World Health Organization.

3:20

4pNSb5. A soundwalk study on the relationship between soundscape and overall quality of urban outdoor places. Mats E. Nilsson (Department of Psychology, Stockholm University, SE-106 91 Stockholm, Sweden, mats.nilsson@psychology.su.se), Jin Young Jeon (Department of Architectual Engineering, Hanyang University), Maria Rådsten-Ekman, Östen Axelsson, Joo Young Hong, and Hyung Suk Jang

In a field study, we explored the relationship between the soundscape and the overall quality (good - bad) of outdoor open places. Thirty three residents in down town Stockholm participated in soundwalks near their homes. Along the soundwalk route, the participants assessed six places with respect to the soundscape, the visual environment and the overall quality of the place using a questionnaire. The six locations were preselected to vary in acoustic and visual quality. A regression model with pleasantness of the auditory and visual environment as predictors explained a substantial part of the variance in assessments of the six place’s overall quality. To disentangle the specific effects of auditory and visual aspects, the present study will be complemented with laboratory experiments in which visual and auditory aspects are independently manipulated.

3:40

4pNSb6. Soundscape: its theory and praxis in Chinese classical garden. Xiao-mei Yuan, Shuo-xian Wu (College of Architecture, South China Univ. of Tech, GuangZhou, GuangDong, 510640, xmyuan@scut.edu.cn), and Yan Wu (Faculty of Arts, Jinan University, GuangDong, 510642)

In this paper, an unique soundscape concept in Chinese classical garden is explicated from theory and praxis levers, which is based on the traditional “Harmony” culture, with the philosophical background of Confucianism, Taoism and Buddhism, and the character of the aesthetic refining and gardening construction experience together up. The general aim of which is creating a artistic conception with nature aesthetics and poetic culture to conduct people’s life to archive the highest state. This paper firstly introduces the formation and development of the concept of Chinese classical garden soundscape, and its related social and cultural reasons are investigated as well. And then, the technical logic contained in garden construction rules are sorted out. So the unique characteristic of soundscape theory and praxis in Chinese classical garden are explicated. Key Words: Chinese Classical Garden; Soundscape; Theory; Praxis

4:00–4:20 Break

Contributed Papers

4:20

4pNSb7. Thinking soundscapes into the management of country parks. Lawal Marafa, Kin Che Lam (The Chinese University of Hong Kong, lmmarafa@cuhk.edu.hk), and Lex Brown (Griffith University, Australia)

Countryside and Country Parks, commonly used as an area for recreation, conservation and education, accommodates sunlight and allows free air movement, thus providing visual relaxation. In recent years, ways to make country parks more attractive for use have been highlighted worldwide. Among a number of promoted common themes regarding the protection and enhancement of country parks, sound as a component of the biological and social environment should be utilized for improving the general environmental quality. Based on the large scale field recordings and questionnaire surveys carried out in 3 country parks in Hong Kong countryside, the relationship between subjective evaluation of soundscape and landscape quality as well as the particular contribution of sound in the enhancement of country parks have been statistically analyzed. Results of this study demonstrate a close relationship between subjective evaluation of landscape quality and acoustic quality. Identification of different sound sources in different functional zones, their particular properties and influence on subjective evaluation of the acoustic quality suggests an alternative approach for improving the acoustic quality and consequently the general environmental quality at the countryside location. Keywords: Country Park, Country side, Recreation, Soundscape

4:40

4pNSb8. Constructing the ideal soundscape: a practical study on closing the gaps between soundscape and urban designers. Daniel Steele (School of Architecture + CIRMMT, McGill University, 815 rue Sherbrooke Ouest, Montreal, Montreal, Quebec H3A 2K6, Canada, daniel.steele@mail.mcgill.ca), and Nik Luka (School of Architecture + School of Urban Planning, Macdonald-Harrington Building 815, rue Sherbrooke Ouest, Montréal, PQ H3A 2K6, Canada)

Calls are increasingly made for an urban land-use policy that takes non-vision sensory modalities into account, like hearing, but agents capable of making such changes often lack the expertise to do so. The best progress in acoustics so far has been through intentional soundscape design, which considers sound during the urban design process rather than after. Indeed, soundscape designers should understand how complicated factors play out in-situ, such as findings linking increased driving speeds with acoustically treated roads. Armed with this knowledge, they can take action to prevent further harm to the urban landscape. In practice, however, what can happens is 1) papers in soundscape are written in language not interesting to urban designers; 2) research studies examine the current environment without proposing design updates; and 3) different investigators fail to agree on what constitutes wanted and unwanted noise. Each of these shortcomings contributes to a built environment that reflects little of our sophisticated understanding. In response, this presentation will: demonstrate how soundscape research can fit into current urban design frameworks; review the literature to suggest some small and large acoustically-optimized urban designs; and encourage collaboration channels for the direct flow of soundscape research into urban design practice.

5:00

4pNSb9. An aural and visual study on the urban open space in Shenzhen Dongmen shopping district using ANN models. Lei Yu (HIT Shenzhen Graduate School, Room 425, E-Building, HIT Campus, Shenzhen University Town, Xili Nanshan, Shenzhen, leilayu@hitsz.edu.cn), Jiang Kang (School of Architecture, University of Sheffield, Western Bank, Sheffield, S10 2TN, UK), and Huan Liu (HIT Shenzhen Graduate School, Room 425, E-Building, HIT Campus, Shenzhen University Town, Xili Nanshan, Shenzhen)

The Dongmen shopping district is the oldest shopping market in Shenzhen. Since 1995, the district has become one of ten famous commercial pedestrian-streets in China. Located in a bustling area of Shenzhen, the Dongmen shopping district attracts a large number of people in its tight-street spaces. An open space is therefore designed, which is important for customers to have a rest after exhausted shopping. It is also a relaxing place for the nearby residents. However, the space quality does not match its recreation function. According to an on-site investigation, subjective evaluations of physical comfort in the space are rather poor, including the aural and visual evaluations. Therefore, this paper is going to investigate how subjective evaluations of aural and visual "scapes" influence physical comfort. Based on analyses of aural and visual evaluations and physical comfort evaluation, artificial neural network (ANN) models of predicting subjective evaluations of physical comfort of the open space in Dongmen are explored. Using the best-trained model, a design with various aural and visual "scapes" for the space is reviewed. It is expected that this study can provide an optimised design scheme for the Dongmen open space in its refurbishment programme in terms of the physical comfort.

5:20

4pNSb10. Changes of soundscape along rural-urban gradients and their influence on landscape preference: a case study from Xiamen, China. Yonghong Gan (Institute of Urban Environment, Chinese Academy of Sciences, Xiamen 361021, China; Department of Bioscience & Technology, Zhangzhou Normal University, Zhangzhou 363000, China, yhgan@iue.ac.cn), Tao Luo (Institute of Urban Environment, Chinese Academy of Sciences, 361021, Xiamen, China), Holger Behm (Landscape Planning and Landscape Design, Faculty of Agricultural and Environmental Sciences, University of Rostock, Rostock 18059, Germany), Timothy Coppack (Institute of Applied Ecology (IfAÖ), Alte Dorfstrasse 11, Neu Broderstorf 18184, Germany), and Jiang Liu (Institute of Urban Environment, Chinese Academy of Sciences, 361021, Xiamen, China; Landscape Planning and Landscape Design, Faculty of Agricultural and Environmental Sciences, University of Rostock, Rostock 18059, Germany)

Humans percept their environment mainly visually, but the acoustic background may have a strong effect on overall landscape preference. Through urbanization, not only the appearance, but also the acoustic background of a landscape is changed. In this paper, the following questions are addressed: (a) How does urbanization change soundscape, in terms of its composition and intension? (b) To what extent does soundscape perception affect the process of landscape preference? With audio and video data from field investigations carried out in Xiamen, China, the differences in composition and intensity of soundscapes along the rural-urban gradient were identified. In surveys with sixty test persons, human perception of visual landscape and soundscape, as well as the preference of overall landscape, were measured. Visual, acoustic, and overall landscape quality of the study area was mapped on the basis of a land-use map. The influence of visual and acoustic factors on overall landscape preference is discussed in view of potential methodological approaches for mapping overall landscape quality by integrating visual with acoustic qualities of landscape. This may provide the basis for an objective multivariate assessment tool in landscape planning. Acknowledgment:Project supported by National Natural Science Foundation of China(40971111) and Natural Science Foundation of Fujian Province, China (2011J01280).

5:40

4pNSb11. Noisy spring?—bird song as a component of urban soundscape perception. Jiang Liu, Holger Behm (University of Rostock, Justus-von-Liebig-weg-6, Rostock 18059, Germany, jiang.liu@uni-rostock.de), Timothy Coppack (Institute of Applied Ecology, Alte Dorfstrasse 11, Neu Broderstorf 18184, Germany), Yonghong Gan, and Tao Luo (Institute of Urban Environment, Chinese Academy of Sciences, 1799 Jimei Road, Xiamen 361021, China)

Where do we stand 50 years after Rachel Carson's (1962) scenario of "a strange stillness" that crept over "a town [...] where all life seemed to live in harmony with its surroundings"? Today, research on urban acoustics is focused more on noise control than on those factors that contribute positively to overall environmental quality. Thus, it seems that no significant progress has been made in our concept of an ideal urban environment. The soundscape approach, which considers not only unfavourable noises, but also desirable environmental sounds, could help to achieve this goal. Here, this hitherto neglected aspect of the urban soundscape was focused on. A field investigation was conducted on human perception of bird sounds in an urban area in Germany. First, the role of bird song as a positive soundscape element was confirmed. Then a series of analysis were performed, including characterization of daily spatial patterns of bird sounds, their contribution to overall soundscape, their relationship with other urban soundscape components, and the connection between avian sounds and urban landscape functions. At last, how bird song could improve urban environmental quality was discussed and possible strategies from an urban planning and a conservation perspective was suggested. Acknowledgement: Project supported by National Natural Science Foundation of China(40971111), Natural Science Foundation of Fujian Province, China (2011J01280), and German Academic Exchange Service(DAAD).

6:00

4pNSb12. The Swedish soundscape-quality protocol. Ãsten Axelsson (ISO/TC 43/SC 1/WG 54, osten.axelsson@comhem.se), Mats E. Nilsson, and Birgitta Berglund

The Swedish Soundscape-Quality Protocol was developed to help non-experts (e.g., officials working for municipalities rather than soundscape researchers) to make informed, accurate measurements of soundscape quality. The Protocol has hitherto been used in England, France, Italy, Spain, Sweden, and The Netherlands; a Korean version is being developed. Based on field studies – soundwalks in urban residential areas, recreational areas, and parks – the present paper reports on the psychometric properties of the scales of the Protocol. Participants were residents, or visitors to the areas and their results support the reliability and validity of the scales in the Protocol. Because high acoustic quality has a greater effect in visually attractive than in visually poor areas, the Swedish Soundscape-Quality Protocol includes scales for cross-sensory tabulation. These are sound source identification – sounds from humans, nature and technology – attribute scales (e.g., eventful, exciting, pleasant, and calm), overall soundscape quality, and concomitant visual impressions. In brief, the Swedish Soundscape-Quality Protocol is an easy to use and practical tool for measuring soundscape quality. It has the potential to help operationalize how soundscapes can be measured in quiet areas to meet a future guideline value of the World Health Organization.

THURSDAY AFTERNOON, 17 MAY 2012 S223, 2:00 P.M. TO 6:00 P.M.

Session 4pPA

Physical Acoustics: Laser Ultrasonics: Fundamentals and Applications

Che-Hua Yang, Cochair
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Zhong Hua Shen, Cochair
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Contributed Papers

2:00

4pPA1. Laser ultrasonic inspection for water transport through membrane in a PEM fuel cell. Ching-Chung Yin (National Chiao Tung University, 1001 Ta Hsueh Road, Hsinchu 30010, Taiwan, ccyin@faculty.nctu.edu.tw), Min-Hsiu Wu, and Yu-Shyan Liu

The performance of a proton exchange membrane fuel cell (PEMFC) was reported to be strongly influenced by water management. This work presents laser ultrasonic inspection for in-situ detection of water across the membrane electrode assembly (MEA). A laser induced grating or an alternative interdigitated piezoelectric fiber composite based acoustic wave transducer was placed on an extension of the flow field plate to launch fixed-wavelength plate waves propagating parallel to the flow channels. The acoustic waves passing through the flow channels underlain by the moist MEA were detected at the opposite end using knife-edge technique. The phase velocities of the acoustic guided waves do not change much between both moist and moist-free states. The insight of water transport could be gained through the excess attenuation of guided waves. The synthetic aperture focusing technique was utilized to establish clear B-mode images of water transport through the membrane.

2:20

4pPA2. Laser ultrasound technique for determine the guided waves propagation in layered medium with temperature gradients. Sheng-Po Tseng and Che-Hua Yang (National Taipei University of Technology, tseng3392@gmail.com)

This paper focuses on the modeling and measurements on the propagation behaviors of guided waves propagating along plate-like wave guides with temperature gradients along their thickness direction. A theoretical model based on a recursive asymptotic stiffness matrix method (RASM) with recursion computation algorism is used to provide numerical calculations for the dispersion relations. A laser ultrasound technique is used to measure the dispersion relations. For all the experiments, the measured dispersion curves show good agreement with the theoretical calculation, indicating the reliabilities in the measurement and modeling. This study is useful to thermometry in different temperature gradient environment in a non-contact and non-destructive ways. This paper demonstrates a procedure employing a LUT introduced the dispersion curves for the steel plate with different temperature distribution. In the spectra, the higher mode dispersion curve could be observed more significant phase velocity difference by LUT system. The phenomenon of temperature gradient is more sensitive for the higher mode. This method is proved to have better accuracy for plate-like structures with lower thermal conductivity. However, applications for structures with higher thermal conductivity will be also applicable while the measurement accuracy is further improved.

2:40

4pPA3. Laser ultrasound technique for material characterization of thermal sprayed nickel aluminum coatings in elevated temperature environment. Cheng-Hung Yeh, Che-Hua Yang, Cheng-Yuh Su, and Wei-Tien Hsiao (National Taipei University of Technology, chyeh0706@gmail.com)

Thermal spraying processing usually use nickel-aluminum alloy system as major powder due to its strong adhesion to substrates. The contents of powder material and the processing parameters used in the spraying process cause material properties of coating exhibiting a wide variation. It is difficult to investigate mechanical properties of coating layer in nondestructive way. This research focuses on characterizing mechanical properties of thermal spraying coatings at high temperature environment up to 295° C in nondestructive way. A laser ultrasound technique (LUT) is used for the measurements of dispersion spectra of guided waves. Theoretical model for surface waves propagating along a multi-layered structure with coating and substrate is used to model the sprayed coatings. An inversion algorithm based on Shuffled Complex Evolution (SCE-UA) is used to extract mechanical properties from the measured dispersion spectra cooperating with theoretical model. In the results, surface wave dispersion spectra are measured for three different thickness coatings at different temperature environment. The difference thickness of coatings and temperature environment are reacted to dispersion spectrum of guided waves. It also related to mechanical properties of coating in theoretical model. This method is potentially useful to characterize the mechanical properties of thermal spraying coating in a nondestructive way.

3:00

4pPA4. A full-field mechanical property mapping with quantitative laser ultrasound visualization system. Chia-Han Wu and Che-Hua Yang (National Taipei University of Technology, chw0105@gmail.com)

This research employs a quantitative laser ultrasound visualization system (QLUVS) for the full-field mechanical property mapping in plate-like structures. The QLUVS has the advantage of fast, full-field and quantitative inspection. The QLUVS uses a pulsed laser generate acoustic waves with fast scanning mechanism to reach two dimensional scanning goal and then detected with a piezoelectric longitudinal transducer. By utilizing QLUVS, the spatial and temporal information of guided wave can be obtained with further signal processing, the velocity map can be extracted. Finally the analytical model and inversion algorithm will be integrated into QLUVS for the full-field mechanical property mapping purpose.

3:20

4pPA5. All-optical probing of laser-induced modulation of crack parameters by surface Rayleigh and surface skimming longitudinal bulk acoustic pulses. Chen-Yin NI (LAUM, UMR-CNRS 6613, Université du Maine; College of Science, Nanjing University of Aeronautics and Astronautics, Chenyin.Ni@univ-lemans.fr), Nikolay Chigarev, Vincent Tournat (LAUM, UMR-CNRS 6613, Université du Maine, Av. O. Messiaen, 72085 Le Mans, France), Nicolas Delorme (LPEC, UMR-CNRS 6087, Université du Maine, Av. O. Messiaen, 72085 Le Mans, France), Zhong-Hua Shen (School of Science, Nanjing University of Science and Technology, Nanjing 210094, P.R. China), and Vitalyi Gusev (LPEC, UMR-CNRS 6087, Université du Maine, Av. O. Messiaen, 72085 Le Mans, France)

All-optical monitoring of the nonlinear motion of surface-breaking cracks is reported. Crack closing is induced by quasi-continuous laser heating, while Rayleigh acoustic pulses and skimming longitudinal surface acoustic pulses are also generated and detected by lasers. By exploiting the strong dependence of the acoustic pulses reflection and transmission efficiency on the state – open or closed – of the contacts between crack faces, the parametric modulation of ultrasonic pulses is achieved. It is demonstrated that detection of the parametric modulation of the reflected and transmitted skimming longitudinal waves and Rayleigh waves mode converted by the crack from skimming longitudinal waves is a sensitive technique for the evaluation of crack modifications and local closure. It is observed that skimming longitudinal waves can be more sensitive to crack motion than Rayleigh waves, which are probing the crack motion without mode-conversion. In comparison with an all-optical frequency-domain technique, 1 the time-domain technique is potentially faster for the imaging applications. This research is supported by the grant ANR-10-BLAN-092302 and a postdoctoral fellowship from the Région des Pays de la Loire for Dr. C. Ni. Also, the support for travelling and participation to the conference from China Postdoctoral Science Foundation (No. 20110491409) is acknowledged. 1 S. Mezil, N. Chigarev, V. Tournat, V. Gusev, "All-optical probing of the nonlinear acoustic of a crack", Opt. Lett. 36, 3449-3451 (2011).

3:40

4pPA6. Numerical simulation and experimental study on surface acoustic waves interacting with cracks heated by scanning heating laser source. Zhonghua Shen, Jia Li (College of Science, Nanjing University of Science & Technology, Nanjing 210094, China, shenzh@mail.njust.edu.cn), Chenyin Ni (LAUM, UMR-CNRS 6613, Université du Maine, Av. O. Messiaen, 72085 Le Mans, France, College of Science, Nanjing University of Aeronautics and Astronautics, Nanjing 210016), and Vitali Gusev (LPEC, UMR-CNRS 6087, Université du Maine, Av. O. Messiaen, 72085 Le Mans, France)

The influence on surface acoustic waves (SAWs) propagate through the micro-crack with partial closure is presented in this paper. Heating brought by laser irradiation causes the thermal expansion of the sample, which leads to partial closure of the micro-crack. The partial closure of crack impacts the transmission efficiency of acoustic pulses strongly. Detected SAWs signals are different when heating laser irradiates different regions, when the middle region of the crack is heated, the amplitude of SAWs signals reach the maximum value. Based on this, the experimental system for detecting micro-crack is set up. The crack can be detected effectively by scanning the laser heating source. The finite element method is applied to simulate the temperature rise and relative displacement of crack edges caused by laser irradiation. The relative displacement change with different location of heating laser source is also calculated. The results of numerical simulation and experiment coincide with each other.

4:00–4:20 Break

4:20

4pPA7. Influence of acoustic leakage at liquid-solid interface on the interferometric detecting surface acoustic waves. Yan Zhao, Zhonghua Shen, Jian Lu, Xiaowu Ni (School of Science, Nanjing University of Science and Technology, Nanjing 210094, P.R. China, zhaoyan7906@mail.njust.edu.cn), and Yiping Cui (Advanced Photonics Center, Southeast University, Nanjing 210096, P.R. China)

Optical interferometry is an important method to detect the surface acoustic wave due to its advantages, such as without contact and a wide responsive bandwidth. So optical interferometric detecting the surface acoustic wave (SAWs) was widely used to diagnosis the surface and subsurface defection, which senses the SAWs by detecting the instantaneous surface displacement on the interfaces. Commonly, the material are placed in air or a special liquid and the surface acoustic wave is leaky and readiates energy into the liquid under the form of bulk waves. As a results, there has another factor to make the interferometric signal change, that is the refractive index variation of liquid induced by the leakage acoustic wave. In this paper, we will mainly analyze the influence of acoustic leakage at liquid-solid interface on the interferometric detecting surface acoustic waves and discuss the suitable condition of ignoring the influence of leakage wave in liquid on interferometric signal to ensure the measuring accuracy of interferometric sensor.

4:40

4pPA8. Theoretical and experimental study of wedge wave mode transformation and energy attenuation. Jing Jia, Zhonghua Shen, Ling Yuan, and Xiaowu Ni (Faculty of Science, Nanjing University of Science and Technology, Nanjing 210094, China, jiasujing2@126.com)

Wedge Waves are guided acoustic waves propagating along the tip of wedge. In this paper, both theoretical and experimental work are done to research the wedge wave modes transformation and energy attenuation. Firstly, finite-element method (FEM) was used to simulate the laser induced wedge waves and different orders of wedge wave modes and the mode transformation process were clearly observed. Then pulsed laser excitation and optical deflection beam method for detection was built to investigate these characteristics experimentally. Plenty of wedge waves at different locations were recorded by scanning the excitation laser along the wedge tip, and different orders of wedge waves were observed. By fixing the distance between the excitation and detection position and scanning the samples along the direction normal to the wedge tip, the modes transformation process was obtained. Finally, theoretical solution by the method of potentials is used to explain the principle of mode transformation and energy attenuation. By integrating the power spectra of the acoustic waves, the energy distribution of these acoustic waves is calculated. Both theoretical and experimental results were found that the energy of acoustic waves decrease exponentially to a steady energy of Rayleigh wave within the main wavelength range.

5:00

4pPA9. High quality photoacoustic tomography in scattering biological tissue. Dan Wu, Chao Tao, and Xiaojun Liu (Institute of Acoustics, Nanjing University, joywudan@gmail.com)

Photoacoustic tomography is an emerging biomedical imaging method, which has a high image contrast and good spatial resolution in deep tissues. Because of these merits, PAT has broad applications in biomedical imaging. However, acoustic scattering is unavoidable in tissue with inhomogeneous acoustic properties. Acoustic scattering brings a challenge for the applications of PAT. A new PAT method is presented to obtain the photoacoustic images in scattering tissue. This method is based on the time-reversal invariance of acoustic wave propagation during the acoustic scattering process.

Both numerical simulations and experiments are used to validate this method. The results show that the proposed method can provide better PAT images in scattering tissues than the traditional photoacoustic tomography. Therefore, this method could serve as an alternative for imaging inhomogeneous biological tissues with acoustic scattering.

5:20

4pPA10. Photoacoustic measurement of the optical absorption spectra of dark or turbid media. David Birtill, Anant Shah, Michael Jaeger, and Jeffrey Bamber (The Institute of Cancer Research, 15 Cotswold Road, Sutton, Surrey, SM2 5NG, UK, david.birtill@icr.ac.uk)

A photoacoustic (PA) spectroscopy system has been built to study small samples, particularly the differences between the PA spectra of oxygenated and deoxygenated blood, and various PA contrast agents, with view to optimising the indentifying these media, in clinical PA images. Short (ns) pulses of light from one of two OPO lasers are delivered into a 1mm diameter cylindrical sample holder. The wavelength is scanned over the range 400-700 nm or 690-950nm (depending on laser used) using a different pulse for each wavelength. Sensitive measurement of the thermoacoustic pressure wave energy emitted from the end of the sample, which acts like a disc-shaped piston source, is facilitated by placing it at the focus of a strongly focused ultrasound transducer. The resulting optical spectra are corrected for some system variables, such as the wavelength-dependent laser energy. Further corrections are planned, so that the measurement is truly of optical absorption coefficient at each wavelength. Even without these additional corrections however, the measured PA spectra of oxygenated blood and gold nano-rods strongly resemble their published optical absorption spectra. In addition to its intended use this system may have applications as a laboratory spectrophotometer, suitable for use with optically dark and turbid media.

5:40

4pPA11. Broadband propagation and photoacoustic time reversal imaging using k-space methods. Ben Cox (University College London, Gower Street, London, WC1E 6BT, UK, b.cox@ucl.ac.uk), and Bradley Treeby (Australian National University, Canberra, Australia)

Numerical, time domain, models of broadband acoustic propagation using k-space methods will be described and applied to the problem of image reconstruction in photoacoustic imaging. k-space methods are a subset of time-stepping pseudospectral methods, which use FFTs to calculate field gradients, with an additional correction to make the solutions exact in the homogeneous case. Furthermore, they are closely related to a number of methods in the mathematical, engineering and physical sciences literature, including non-standard finite difference schemes, exponential integrators, wavenumber domain reverse time migration, and wave propagator models developed to solve the time-dependent Schroedinger equation. These connections will be discussed and used to illuminate the advantages of the k-space approach for large scale modelling of broadband acoustic waves. To illustrate the method, a fluid-based k-space model will be applied to the inverse acoustic initial value problem encountered in tomographic photoacoustic imaging. It is well known that this can be solved, in the linear case, using a time-reversed numerical model. Extensions from this basic case to absorbing and non-linear media will be explored.

THURSDAY AFTERNOON, 17 MAY 2012 S423, 2:00 P.M. TO 6:00 P.M.

Session 4pPP

Psychological and Physiological Acoustics and Signal Processing in Acoustics: Psychological and Physiological Basis of Tonal Language Processing

Fan-Gang Zeng, Cochair
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Michael Tong, Cochair
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Invited Papers

2:00

4pPP1. Relative contributions of temporal and spectral cues for Mandarin and Cantonese tone recognition. Ying-Yee Kong (Northeastern University, Speech Language Pathology and Audiology, Boston, MA 02115, yykong@neu.edu), Tan Lee (The Chinese University of Hong Kong, Department of Electronic Engineering, Shatin, Hong Kong), Meng Yuan (Bionic Ear and Sound Technology Laboratory, Shanghai, China), and Wilson Yu (The Chinese University of Hong Kong, Department of Electronic Engineering, Shatin, Hong Kong)

Despite good speech perception performance in quiet, pitch perception remains a challenge for cochlear-implant users. The current vocoder-based processing technique in cochlear implants preserves temporal envelope and coarse spectral information. This information, however, is insufficient to support high-level performance for tone perception in tonal languages (e.g., Mandarin and Cantonese), particularly in noise. Mandarin has four lexical tones and Cantonese has six lexical tones. Each lexical tone has different fundamental frequency contours. The temporal envelope of the signal also differs among different tones. In this talk, we will first discuss the relative contributions of temporal and spectral envelope and fine structure cues for Mandarin and Cantonese tone perception. We will then describe a new signal processing algorithm developed at the Chinese University of Hong Kong that aims to enhance temporal periodicity cues for Cantonese tone perception in noise. We will present findings from normal-hearing and cochlear-implant listeners, which demonstrate a significant improvement of Cantonese tone perception in noise with this new algorithm.

2:20

4pPP2. Recognition of Mandarin Chinese in noisy, reverberant environments. Liang Li (Department of Psychology, Peking University, Beijing 100871, China, liangli@pku.edu.cn), Xihong Wu (Peking University), and Bruce Schneider (University of Toronto, Mississuaga)

Cochlear-implant users partially recover their speech intelligibility in quiet but not in a noisy, reverberant environment, particularly for those speaking tonal languages, for which semantic information is also expressed by pitch contour. To improve cochlear-implant algorithms for tonal-language users, we have investigated speech recognition in Mandarin-Chinese speaking listeners under adverse listening condition to address four issues related to perceptual fusion and informational masking. First, to what extent do Chinese speech and non-speech sounds differ with respect to the tendency of perceptual fusion (between direct and reflected waves)? Second, why does perceptual separation provide a smaller release from informational masking in Mandarin Chinese than in English? Third, can the use of the recently developed simulated phase-locking stimulation strategy (SPLS, which extracts both phase and amplitude-envelope information) improve speech perception in Mandarin-speaking cochlear-implant patients compared to the continuous interleaved sampling strategy (CIS) currently in use? Fourth, does the prior presentation of a sentence spoken in quiet, by the same person who immediately afterwards produces a masked target sentence, improve identification of the masked target (voice priming) only for tonal-language speaking listeners?

2:40

4pPP3. The perceptual normalization of lexical tones: effects of surrounding tonal context. Valter Ciocca (School of Audiology and Speech Sciences, University of British Columbia, 2177 Wesbrook Mall, Vancouver, BC, V6T 1Z3, Canada, vciocca@audiospeech.ubc.ca), Alexander Francis (Department of Speech, Language, and Hearing Sciences, Purdue University, Heavilon Hall, 500 Oval Drive, West Lafayette, IN 47907), Elaine Eramela (Wonderworld Speech Therapy Clinic Unit 1703A-1705, Landmark North, Sheung Shui, New Territories, Hong Kong), Teresa Siu Kwan Yau (Professional Speech and Hearing Services Ltd., Room 1133, Pioneer Centre, 750 Nathan Road, Kwoloon, Hong Kong), and Wing Man Shum (Hong Chi Winifred Mary Cheung Morninghope School, 220 Lai King Hill Road, Kwai Chung, Kowloon, Hong Kong)

Previous research has shown that the perceptual categorization of a target lexical tone depends on its surrounding tonal context ("tone normalization"). Native speakers have been shown to take into account both preceding and following tonal context in order to carry out the normalization process. The present study focused on the effects of the duration and type of preceding tonal context on tone normalization. Listeners identified the tone of a syllable in final sentence position, as a function of the number of syllables (1, 2, 3, or 4) and of the tone type (level or contour) of a semantically-neutral precursor sentence. The results showed that: 1. The effect of the precursor sentence reached an asymptote once listeners heard two syllables; 2. The tone type of the precursor sentence did not affect performance. This evidence is consistent with a tone normalization process that operates on the basis of a running F0 average of the surrounding tonal context. When tonal context precedes the target tone, the running F0 average is effectively computed over a two-syllable interval.

3:00

4pPP4. Central nervous system markers for lexical tone learning. Patrick C.M. Wong (Knowles Center for Hearing, Northwestern University, Evanston, IL 60208, pwong@northwestern.edu)

A large degree of individual variability can be observed in language learning in adulthood. This variability is especially prominent in the learning of foreign sounds, including lexical tones. In this presentation, I will report a series of experiments from my research group that examines the central nervous system markers for lexical tone learning success. Our knowledge such as collicular sensitivity to frequency modulation and cortical basis of pitch perception can begin to assist in generating specific hypotheses for these markers. In our experiments, native English-speaking adults learned Mandarin-like tone patterns in a lexical context. Behavioral, electrophysiological, neural hemodynamic (fMRI), and neuroanatomical measures were collected to characterize variability in learning success. Among the markers of success we found were integrity of the frequency following response, activity in the posterior auditory cortex, gray matter volume in the Heschl's Gyrus, and white matter connectivity within the auditory cortex. Some of these markers were used to redesign training in order to optimize learning for all learners, including training strategies such as enhancing auditory pitch cues and reducing trial-by-trial stimulus variability. Taken together, these experiments provide information regarding the neural basis of lexical tone learning and demonstrate the feasibility of using biomarkers for designing training. [Work supported by NIH and NSF]

3:20

4pPP5. Acoustic cues for lexical tone perception in hearing-impaired listeners. Shuo Wang (Beijing Tongren Hospital, Capital Medical University, Beijing Institute of Otolaryngology, Beijing 100730, China, shannonwsh@yahoo.com.cn), Li Xu (Ohio University, Athens, OH 45701), Ruijuan Dong, Jing Li (Beijing Tongren Hospital, Capital Medical University, Beijing Institute of Otolaryngology, Beijing 100730, China), Robert Mannell (Macquarie University, Sydney, Australia), and Luo Zhang (Beijing Tongren Hospital, Capital Medical University, Beijing Institute of Otolaryngology, Beijing 100730, China)

This series of studies was aimed to investigate how listeners with sensorineural hearing loss (SNHL) and with auditory neuropathy syndrome disorder (ANSD) achieved lexical tone recognition using either the temporal envelope (E) or the fine structure (FS) cues. Five groups of Mandarin-speaking subjects, including (1) 22 normal-hearing subjects, (2) 8 moderate, (3) 13 moderate to severe, (4) 10 severe SNHL patients with various degrees of SNHL, and (5) 10 patients with ANSD, participated in the study. Monosyllabic words were processed through a 16-channel "auditory chimera" in which E from a monosyllabic word of one tone was paired with FS from the same monosyllable of other tones. On average, 92.0%, 67.4%, 58.1%, 37.5%, and 17.1% of the tone responses were consistent with FS cues, while 5.8%, 23.7%, 31.1%, 45.2%, 42.7% of the tone responses were consistent with E cues for the 5 groups of subjects mentioned above. Therefore, as the hearing loss becomes more severe, the ability of SNHL patients to use FS for tone recognition becomes more deteriorated. The ability of ANSD subjects to use FS is even poorer than patients with severe SNHL even though their pure-tone thresholds were only moderately elevated in the low and mid frequencies.

3:40

4pPP6. Lexical tone development in children with cochlear implants. Li Xu (Ohio University, Athens, OH 45701, xul@ohio.edu)

Most of the languages in the world are tonal. In a tonal language, voice pitch variation (i.e., tone) at the syllable level is a segmental structure that conveys lexical meaning of words. Multichannel cochlear implants (CIs) have shown great success in providing profoundly-deafened individuals with satisfactory speech perception in quiet. However, contemporary speech-processing strategies used in CIs do not explicitly code pitch information. This presentation will be focused on (1) acoustic cues for recognition of lexical tones, primarily the Mandarin Chinese tones, and the relative contributions of various cues to tone recognition, (2) results of tone recognition experiments in implant recipients in relation to their differences in demographics, devices, strategies, and psychoacoustic abilities, (3) relationship between music pitch perception and lexical tone recognition, and (4) results on tone production and vocal singing in prelingually-deafened, native tonal-language speaking children with CIs. It is concluded that there are marked deficits in tone development in tonal-language-speaking children with CIs. Results also indicate that early implantation and experience with using the device help improve tone development in prelingually-deafened tonal-language-speaking children. [Work supported by the NIH/NIDCD.]

4:00–4:20 Break

4:20

4pPP7. Importance of temporal periodicity on Mandarin tone perception of cochlear implant recipients. Meng Yuan, Jin Sun, Youyuan Chen, and Haihong Feng (Shanghai Acoustics Laboratory, Chinese Academy of Sciences, No. 456, Shanghai Xu Hui Qu Xiao Mu Qiao Rd., Shanghai, China, yuanmeng61@gmail.com)

Cochlear implants (CIs) were designed primarily to enhance the speech perception of western persons with bilateral sensorineural deafness. The CI technology used in the current commercial devices, in terms of both hardware and software, didn't focus on the delivery of tone or pitch related information. It's known that lexical tone is important to Mandarin-speaking people. To better encode pitch-related information, the current study will investigate the effect of temporal envelope periodicity to Mandarin tone perception of CI recipients. Temporal periodicity from each frequency band is manipulated corresponding to the fundamental frequency (F0) of the speech signal. Psychophysical experiment was carried out to evaluate the Mandarin tone perception performance of CI recipients in quiet and noisy conditions. Test materials were monosyllabic and disyllabic Chinese words with four different tones. Experimental results showed that F0-related coding on temporal periodicity can improve the Mandarin tone perception ability of CI recipients. It is indicated that new speech processing strategy should be developed for better tone perception of Chinese CI recipients. This work is supported by National Natural Science Foundation of China (11104316), and Shanghai Natural Science Foundation (11ZR1446000).

4:40

4pPP8. Combined electric and acoustic stimulation to improve tonal language processing in cochlear implant users. Fan-Gang Zeng and Hsin-I Yang (University of California, Irvine, CA 92697-5320, fzeng@uci.edu)

A unique characteristic of tonal language processing is the use of multiple acoustic cues that are distributed in both time and frequency domains to achieve robust recognition of tones. The temporal cues can be represented by simple acoustic differences in duration or amplitude contours that are correlated to tonal patterns. The spectral cues can be represented directly by fundamental frequency or indirectly by its harmonics. Because contemporary cochlear implants process only the less salient temporal cues, their users have generally poor tonal processing abilities. The present study takes advantage of residual low-frequency acoustic hearing (e.g., < 500 Hz) that is often present in the cochlear implant users. Although this low-frequency hearing contributes little to speech intelligibility directly, it is sufficient to convey tonal information via fundamental frequency. The present study measures word recognition in noise in a group of Mandarin-speaking cochlear implant users who have significant residual low-frequency acoustic hearing. Preliminary data showed that combined electric and acoustic stimulation significantly improved word recognition in noise. The mechanism for this improvement will be discussed in terms of independent or synergetic contributions from the acoustic and electric cues.

Contributed Papers

5:00

4pPP9. Cochlear implant users' melody recognition with pitch and loudness cues. Xin Luo, Megan Masterson, and Ching-Chih Wu (Purdue University, luo5@purdue.edu)

Normal-hearing listeners can recognize familiar melodies from loudness changes, although more poorly than from pitch changes. Two hypotheses were tested here: (1) cochlear implant (CI) users can also use loudness changes to perceive melodic contours, and (2) their melody recognition can be enhanced with consistent pitch and loudness changes. In Experiment 1, melodic contours were created by changing the F0s of harmonic complex tones, by changing the intensities of broadband noise bursts, or both. In Experiment 2, familiar melodies were recorded by a pianist. The note intensities were kept as recorded, equalized at 70 dB SPL, or changed from 45 to 75 dB SPL to match the relative pitch changes in semitones. Loudness melodies were also generated with intensity-changing noise bursts. The results showed that CI users' melodic contour identification was better with both pitch and loudness changes than with either alone. Specifically, adding loudness changes significantly improved identification of melodic contours with 1-semitone intervals. CI users had similar familiar melody recognition with pitch or loudness changes alone. For all but one subject with the lowest performance, familiar melodies were not better recognized when pitch and loudness changed together, showing a lack of integration between the two cues.

5:20

4pPP10. Effects of two acoustic continua on the within-category perceptual structure of tones. Junru Wu (Phonetics Laboratory, Leiden University Centre for Linguistics, the Netherlands, Lipsiusgebouw, Cleveringaplaats 1, 2311 BD Leiden, Room Number 104, j.wu@hum.leidenuniv.nl)

The present study investigated effects of two acoustic continua on the within-category perceptual structure of Putonghua Tone 2 and Tone 3. These two tones were simulated with tokens varying along two acoustic continua about F0 contour: the timing of F0 turning point and falling of F0. Three

different syllable durations were tested with voice quality under control. Multi-dimensional scaling analyses were applied to investigate relative influence of phonetic identification and category goodness on the perceptual dissimilarity of synthesized tonal tokens. The result revealed that Tone 3 has later F0 turning point and greater F0 falling than Tone 2, which confirms former findings. The new finding is that perceptual representation of these two tones categories is different in their internal structures. Best tokens disperse within categories and are usually not unique. Perceptual space involving Tone 2 tokens shrink but that involving Tone 3 doesn't. Goodness rating contributes significantly to the dissimilarity scaling across Tone 2 tokens but not Tone 3 tokens.

5:40

4pPP11. How is a five-level-tone contrast possible? Jianjing Kuang (UCLA Campbell Hall 3125, kuangjj@gmail.com)

Black Miao is the most famous five-level-tone language (reported in Kwan1966), which has been a challenge for tonal theories; however, the phonetic realizations of these five level tones have not been studied. This study thus examines the same Black Miao dialect as the one Fang-Kuei Li recorded. We conducted both production and perception experiments in the field to understand how this five-level-tone contrast is possible. Simultaneous EGG and acoustic recordings were made in the production experiment; the perception experiment consisted of an identification task and a discrimination task. Preliminary results show that these five level tones do not merely contrast in F0; non-modal phonations play an important role in the tonal contrasts. Vocal fry and tense voice help distinguish 11 and 55 from other tones, while breathy phonation distinguishes 33 from 22 and 44. Therefore, the tonal contrasts are optimal in dispersion in a 2-D (F0 and phonation) tone space.

THURSDAY AFTERNOON, 17 MAY 2012 — S425, 2:00 P.M. TO 7:00 P.M.

Session 4pSP

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

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Contributed Papers

2:00

4pSP1. Multiple maskers for speech masking in open-plan offices. Yue Wang (Institute of Acoustics, Chinese Academy of Sciences, enereswang@gmail.com), Horst Albert Drotleff (Fraunhofer Institute of Building Physics), and Ping Li (Institute of Acoustics, Chinese Academy of Sciences)

Sound masking is an effective method to reduce speech intelligibility for reaching acceptable speech privacy in open-plan offices. Speech-like masker is considered as a good choice to reduce speech intelligibility. Using one masker in masking system is the common way so far. In this paper, a new method was proposed, using randomly integrated speech frames (SF) of the speaker as masker. Because SF masker was based on speech of the same speaker, it is a good speech-like masker. We further studied the method of using multiple SF maskers. The performance of this new masking method was evaluated by objective measurement (STOI). The speech masked by one SF masker and multiple SF maskers were compared. Results showed that speech with SF masker could reduce speech intelligibility. Speech with multiple SF maskers led to poorer speech intelligibility than speech with only one SF masker. Masking speech by multiple SF maskers is an effective way to reduce speech intelligibility in open-plan offices. (Acknowledgement: This research was supported by the joint training PhD program of Chinese Academy of Sciences-Fraunhofer, and Fraunhofer Institute of Building Physics in Stuttgart, Germany.)

2:20

4pSP2. Acoustic privacy area generation based on simple summation of numerous loudspeaker signals. Takuma Okamoto, Yukio Iwaya, and Yoiti Suzuki (Tohoku University / 2-1-1 Katahira, Aoba-ku, Sendai, 980-8577, Japan, okamoto@ais.riec.tohoku.ac.jp)

We propose a new speech privacy technique based on simple summation of numerous signals using N-channel loudspeakers. A speech signal mixed with high-level white or pink noise and with a delay set appropriately for each channel is reproduced by each loudspeaker to be synchronized at a specified sweet spot. At the sweet spot, SNR increases proportionally to the square root of N. A speech signal at and around the sweet spot is enhanced significantly, making it easily intelligible. Computer simulations were conducted assuming a linear array, a circular array, and a surrounding 157-loudspeaker array to estimate SNR and the Speech Intelligibility Index (SII). Results show that SNR and intelligibility in terms of SII effectively increase only at and near the sweet spot when using a surrounding 157-loudspeaker array. The SII at the sweet spot increases to 0.46, although those at areas distant from the sweet spot are nearly 0.

2:40

4pSP3. Real-time sound source localization based on multi-resolution scanning in frequency domain. Kohei Hayashida, Masanori Morise, Takanobu Nishiura, and Yoichi Yamashita (Ritsumeikan University, 1-1-1 Noji-higashi, Kusatsu, Shiga, Japan, cm012063@ed.ritsumei.ac.jp)

Robust speech recognition is necessary for realizing useful speech interfaces. A microphone array is an effective item at capturing distant-talking speech with high-quality in noisy environments. It captures the target speech by localizing a talker and steering the directivity. These processing is usually executed in each frequency. For realizing useful speech interface, these processing must be finished in real-time. In the research into sound source localization, various methods have already been developed, and these methods localize a sound source based on acoustic space scanning by fixed resolution in each frequency. Therefore, calculation time is increased, and the real-time processing is difficult in higher spatial resolution. To overcome this problem, we proposed the localization method based on multi-resolution scanning in frequency domain. The lower frequency band has lower spatial resolution, and the higher frequency band has higher spatial resolution. The proposed method localizes the sound source by lower spatial resolution in lower frequency band and higher spatial resolution in higher frequency band. Therefore, the proposed method can reduce the calculation time without degrading the localization accuracy. The experimental results in noisy environment indicated that the proposed method could reduce the calculation time and could achieve the real-time sound source localization.

3:00

4pSP4. An identification of speaker-dependence in reverberant-robust speech recognition. Takahiro Fukumori, Masanori Morise, Takanobu Nishiura, and Yoichi Yamashita (Ritsumeikan University, 1-1-1, Nojihigashi, Kusatsu, Shiga 525-8577, Japan, cm013061@ed.ritsumei.ac.jp)

In recent years, a hands-free speech device has been developed with improving speech-recognition techniques. There is, however, a problem that the reverberant speech degrades the recognition performance in the field of distant-talking speech recognition. It is possibly addressed by taking preventive measures against the degradation of recognition performance with the reverberant criteria to estimate the recognition performance. We have already proposed the method to estimate recognition performance with ISO3382 acoustic parameter based on an impulse response. In this method, the recognition performance was estimated without speech features. Identification of the speaker with robust or weak features against reverberation makes it possible to adapt acoustic model for each speaker toward improving the recognition performance. In this research, we designed the speaker-dependence criteria in reverberant speech recognition. We first investigated existence of the speaker with robust or weak features against reverberation in various reverberant environments. After that, we compared clean and reverberant speech data in terms of speech features such as MFCC, delta MFCC, delta power, and utterance speed to evaluate the effects of reverberation on speech recognition. An experimental result showed the utterance speed was one of the effective candidates for the identification of speaker-dependence in reverberant-robust speech recognition.

3:20

4pSP5. Joint Doppler and time delay estimation by compressed sampling. Xuan Li (Institute of Acoustics, Chinese Academy of Sciences, 100190, lixuan.ioa@gmail.com), Xiaochuan Ma, Shefeng Yan, and Chaohuan Hou

Joint Doppler shift and time delay estimation is an important topic in radar, sonar and communication applications. Least square (LS) is a classical and effective method for solving the problem. However, the performance degrades severely in the scenario of low ratio of signal-noise (SNR), due to the instability of matrix inversing. In this paper, a high-resolution method is proposed basing on the compressive sampling theory. The 2-dimension channel response can be sparsely recovered, and high-resolution Doppler shift-time delay estimation can be described with an underdetermined equation solving problem. Three categories of algorithms, including diagonal loading least squares, ℓ_1 Regularization, and Greedy Pursuit, are adopt to solve the problem and show outstanding resolving capabilities. The three categories of algorithms are analyzed and compared in different conditions. For the dictionary appears not to have unit norm columns, Greedy pursuit is not good as ℓ_1 Regularization in general, and simulation results demonstrate it.

3:40

4pSP6. An efficient post-processing filter for multi-channel acoustic echo cancellation system. Henglizi Zhang, Kai Chen, and Jing Lu (Key Laboratory of Modern Acoustics (MoE), Institute of Acoustics, Nanjing University, Nanjing 210093, China, zhanghenglizi@gmail.com)

The acoustic echo usually cannot be totally removed by the adaptive filter due to the insufficient filter taps, nonlinearity of the echo path transfer function, and the background noise. A post-processing filter, which aims to suppress the residual echo, thus plays an important role in the acoustic echo cancellation system. In this paper, an efficient post filter is proposed based on the coherence between the error signal and the echo estimation of the adaptive filter. The proposed method can be seamlessly combined with the multi-delay frequency-domain adaptive filter. Furthermore, it can be easily extended to multi-channel processing. The instrumental evaluation and listening test both demonstrate the superiority of the proposed method.

4:00–4:20 Break

4:20

4pSP7. A study of beamforming in FDTD method to synthesize directional sound for sound field reproduction. Kota Nakano, Masanori Morise, Takanobu Nishiura, and Yoichi Yamashita (Ritsumeikan University, 1-1-1, Noji-higashi, Kusatsu City, Shiga prefecture, 525-8577, Japan, cm010064@ed.ritsumei.ac.jp)

Sound field reproduction is useful to represent sound contents with high-realistic sensation. We have proposed a sound field reproduction system to control the perceived angles of virtual sound sources. The system synthesizes the propagated sounds by using sound field simulation and multi-directionally represents them with head-enclosed loudspeakers. The multi-directional sound representation requires the sound field simulation to synthesize directional sounds. We therefore focus on beamforming by array processing in finite-difference time-domain (FDTD) method. FDTD is an effective method to simulate sound field. It discretizes the time-space with meshes and defines the distribution of sound pressure level (SPL) by calculation result of the SPL propagation between the each mesh. We accordingly propose a new approach for sound field simulation to synthesize directional sound. In the proposed approach, the beamforming is applied to the SPL propagation between FDTD meshes. The beamforming with FDTD meshes achieves to control the directional characteristics at each mesh for SPL propagation. With the directional characteristics controlled by beamforming, the directional sound is synthesized. According to the evaluation conducted to verify the performance, it was confirmed that the proposed approach enabled FDTD to control the directional characteristics for directional sound synthesis.

4:40

4pSP8. Persymmetric adaptive detection of distributed targets in homogeneous environment. Chengpeng Hao, Xiaochuan Ma (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, haochengp@mail.ioa.ac.cn), Shefeng Yan (Institute of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China; State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China), Qi Xu, and Chaohuan Hou (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China)

In this paper, we deal with the problem of adaptive detection of distributed targets in Gaussian disturbance with unknown but persymmetric structured covariance matrix. A homogeneous environment is considered at the design stage, and a receiver based on the generalized likelihood ratio test (GLRT) is derived. Remarkably the new receiver ensures the constant false alarm rate (CFAR) property with respect to the covariance matrix of the disturbance. Finally, a performance assessment, conducted by Monte Carlo simulation, has shown that the new receiver can significantly outperform its unstructured counterpart due to embedded a priori knowledge about the structure of the disturbance covariance matrix, especially in a severely heterogeneous scenario where a very small number of training data is available.

5:00

4pSP9. Design and implementation of an acoustic signal acquisition system based on Wi-Fi. Xin Li (No. 21, Bei-Si-huan-Xi Road, Beijing, China, lixincome@gmail.com)

This paper presents an acoustic signal acquisition system based on Wi-Fi aimed at acquiring real-time acoustic signals and vibration signals, and transferring them to a center server for further processing and analyzing. In this design, the system is a three-layer hierarchy consisting of wireless nodes, access point (AP) managers and a center server. The center server manages all AP managers and receives signals from them. Wireless nodes can search for the desired AP according to signal strength and are managed by the corresponding AP manager. And the robustness of the communication between AP managers and wireless nodes is enhanced by an improved heartbeat mechanism. The system has been applied into an existing acoustic signal acquisition system and desired performances have been achieved.

5:20

4pSP10. Determination of the physical parameters of systems with a time-domain approach. Yum-Ji Chan and Yumin Zhang (Mechanical Engineering Department, The University of Hong Kong, yjchan@hku.hk)

A time-domain method to analyze the effective stiffness, damping and mass of systems, such as a loudspeaker or a porous cavity, is presented. A time-domain approach is taken to show the phase variation of greater than 2π in transfer functions. With the help of the least-mean-square (LMS) algorithm, the impulse response can be sought by feeding random noise into the systems. The impulse response is then used to deduce the effective stiffness, damping and mass of the system. Experimental demonstration with shunt loudspeakers is presented, and the traditional approach of analyzing air cavities filled with porous material is evaluated.

5:40

4pSP11. A new soft decision feedback equalizer in underwater acoustic communications. Xiaoxia Yang, Jun Wang, and Haibin Wang (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, xiaoxiayang1987@gmail.com)

The tap coefficients of the decision-feedback equalizer (DFE) are associated with the multipath spread of the channel. For the underwater acoustic communication channel, the impulse response often covers tens to hundreds of symbols, requiring at least tens of taps in the feedback filter. Once the incorrect decisions are fed back, the error propagation will be severe. In this paper, we propose a new soft decision equalizer. The expected output symbols of the equalizer will be weighted according to their reliability, and summed up, then the result is fed back. This approach can alleviate the effect of those incorrect symbols. For evaluating the performance of the proposed method, a sea test in shallow water was carried out. The experimental results show that its BER decreases by 50% without channel codec, compared with the conventional DFE. Furthermore, the mean square error can also reduce about 2 dB when the error propagation is severe.

6:00

4pSP12. Design of wavelet frequency-division based signal processing algorithm for an implantable middle ear hearing device. Jiabin Tian, Zhushi Rao, and Na Ta (State Key Laboratory of Mechanical System and Vibration, Shanghai Jiao Tong University Dongchuan RD. 800, Shanghai 200240, China, tian201002@sjtu.edu.cn)

A signal processing algorithm was designed based on wavelet frequency division for an implantable middle ear hearing device(IMEHD) to make the piezoelectric actuator's vibration simulate normal middle ear transfer function. Firstly, the input digital signals were divided into seventeen bands based on Bark frequency scale using wavelet transform. Then, the assigned band gains were applied corresponding to middle ear transfer function. Finally, the algorithm was implemented and verified the performance through an experiment. The satisfactory agreements between the output of the piezoelectric stack and normal middle ear transfer function indicate that the designed algorithm is feasible. This work is supported by the National Natural Science Foundation of China (Grant No. 11072145 and No. 81170910) and the Science and Technology Commission of Shanghai Municipality Foundation (No. 08JC1404700). We are also thankful to Dr S. K. Yin, Department of Otolaryngology, Shanghai Sixth People's Hospital, China, for the facilities and encouragement given.

6:20

4pSP13. Study on mechanical feedback of hearing aids using maximum length sequence method. Jie Cui, Yongjia Ge, and Xiaoli Han (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-Huan-Xi Road, Haidian District, Beijing, China, cuij@mail.ioa.ac.cn)

Mechanical feedback is one of reasons causing hearing aids to howl, therefore the mechanical feedback problems of behind the ear (BTE) hearing aids was investigated by the maximum length sequence (MLS). The realtime data transmission was set up between a BTE digital hearing aid and a computer. The feedback path form the input port of receiver to the output port of microphone was measured with different input magnitude of receiver, in order to investigate the effect of mechanical feedback on the whole feedback path. The experience results verified the availability of the method.

6:40

4pSP14. A novel F0 estimator for pitch enhancement in cochlear implant. Haiyang Hu (Graduate University of Chinese Academy of Sciences and Shanghai Acoustics Laboratory, No. 456, Xiaomuqiao Road, Shanghai, huhaiyang87@gmail.com), Meng Yuan (Shanghai Acoustics Laboratory, No. 456, Xiaomuqiao Road, Shanghai), Qinglin Meng (Graduate University of Chinese Academy of Sciences and Shanghai Acoustics Laboratory, No. 456, Xiaomuqiao Road, Shanghai), and Haihong Feng (Shanghai Acoustics Laboratory, No. 456, Xiaomuqiao Road, Shanghai)

Current experimental Cochlear Implant (CI) sound processing strategies like F0mod [Laneau, et.al., 2006], which aim to improve the pitch perception of CI subjects, amplitude-modulate the envelope of each frequency channel by the fundamental frequency (F0) of the speech sound. Due to the hardware limitation of CI devices, typical F0 estimation algorithms are too complex to be implemented. Low-complexity and accurate F0 estimator will be necessary for such manipulation. This study uses the Summary Auto-Correlation Function (SACF) to extract F0. The new pitch estimator was simulated and compared with typical algorithms that estimate F0 before the sub-band filtering. By the use of SACF, the new estimator reduces the computation load significantly and preserves good F0 estimation performance. # This work is supported by National Natural Science Foundation of China (11104316), and Shanghai Natural Science Foundation (11ZR1446000).

Session 4pUWa

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

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Contributed Papers

2:00

4pUWa1. Detection of broadband sound sources using a randomly spaced linear array. Mario Zampolli, Laurent Fillinger, Alan J. Hunter, and Martijn C. Clarijs (TNO Acoustics and Sonar Group, 2597 AK, The Hague, The Netherlands, mario.zampolli@tno.nl)

Arrays are commonly used in relatively narrow band applications. The array design and processing are usually optimized in order to maximize the gain in the look-direction, while minimizing the interference from sources located in other directions (but possibly not far off from the look-direction). This led for instance to the definition of minimum redundancy linear arrays that maximize the resolution for a given number of sensors. This maximization is, however, effective only at the design frequency of the array. For many active applications (such as active sonar or radar), which are narrow band in essence, this limitation is not an issue. However, if the signal to be detected in its entirety has a broadband frequency content, optimizing the performance of the array at a given frequency leads to possibly poor performance at other frequencies. Since optimal performance cannot be achieved at all frequencies simultaneously, one can aim for sub-optimal but useful performance throughout the entire frequency band. One possibility for achieving this is based on the use of a random hydrophone spacing. The effectiveness of the approach is investigated using simulations and is illustrated with experimental data collected using a randomly spaced linear array.

2:20

4pUWa2. Passive localization using flexible hydrophone array. Lijun Chen, Liang An, and Xiang Gao (School of Information Science and Engineering, Southeast University, ljchen@seu.edu.cn)

Localizing a radiated acoustic source in shallow water precisely is a difficult problem. To achieve this goal, a large aperture between hydrophones should be considered. But a large and rigid array is not convenient when used by ship. Here a flexible hydrophone array is designed to solve this problem. The position of each array hydrophone is randomly placed. Spherical interpolation method is used to estimate the position of acoustic source. This method need precise 3-D coordinates of each hydrophone to estimate 3-D coordinates of the source. In this paper a new method is derived to estimate the coordinates of each hydrophone. Computer simulation is conducted to analysis the performance of passive localization algorithm using the flexible array. Factors such as time-delay, hydrophone coordinates, and array shape, which affect the localization performance are discussed. Lake trial results are presented and show that the experiment results agree highly with the simulation results.

2:40

4pUWa3. Investigation of the horizontal directivity of underwater ambient noise with the circular hydrophone array. Shi Yang, Yang Yixin, and Ma Yuanliang (College of Marine, Northwestern Polytechnical University, Xi'an 710072, China, shiyangeagle@gmail.com)

The ambient noise field in shallow water is highly variable in time and space. The horizontal directivity pattern of ambient noise is anisotropic and can be exploited for source localization in sonar signal processing. In the present work, the 12-element circular hydrophone array with radius of 1 meter was designed and used to observe the ambient noise in the shallow water experiment in the Yellow Sea. The MVDR beamformer with diagonal loading was designed to get the horizontal directivity pattern in frequency band of 1kHz to 3kHz. The conventional beamformer was used to analyze the horizontal directivity of ambient noise above 3 kHz. The noise data were sampled at 25kHz, bandpass filtered between 100Hz and 5kHz in the experiment. The horizontal directivity patterns at the experiment location at different frequencies were obtained based on above methods and the noise data. This work was supported by the National Natural Science Foundation of China(10734030)

3:00

4pUWa4. Study of ocean ambient noise characteristics based on vector signal processing of acoustic. Jialiang Li (Chinese Academy of Sciences, No. 8, Shangqing Road, Shibei District, Qingdao 266023, Shandong Province, China, lijiaqingdao001@163.com), Jianheng Lin, and Xuejuan Yi

With the development of technology, vector sensors are more and more applied in underwater acoustics measurements. As one of the signal processing methods, vector signal processing method based on acoustic energy flow is gradually developed. Methods based on acoustic energy flow overcome some inherent shortcomings of traditional pressure signal processing methods and increase processing gain and DOA estimation accuracy. Methods based on acoustic energy flow have been verified to be practically useful by trials. The existing vector sensor frequency measurement model is modified in this paper. The anisotropy and other characteristics of ocean ambient noise are studied in the paper based on vector signal processing of acoustic energy flow, and the simulation results are reasonable.

3:20

4pUWa5. Directional pattern of a cross vector sensor array. Shi-e Yang (Acoustic Science and Technology Laboratory, Harbin Engineering University, Harbin, Heilongjiang Province, China, yangshie@hrbeu.edu.cn)

The directional pattern of array is a kind of spatial filter for signal processing. For the signal coming from different azimuth can be expressed as a periodic function of, it can be expanded in Fourier series. With the help of a cross vector sensor array, more directional patterns such as quadrupoles and octopoles can be obtained. By solving equations, the first few coefficients of Fourier series can be determined. A combine directional pattern of the vector sensor array, which has

a maximum value at the required direction and very small value at other directions, is properly chosen to improve the performance of the final directional pattern. The method of determining virtual directional pattern proposed in this paper greatly reduces calculation quantity rather than full-space scanning.

3:40

4pUWa6. Bispectrum and cross-bispectrum feature extraction based on vector hydrophone. Lanyue Zhang, Yang Wang, and Desen Yang (Harbin Engineering University, 150001, zhanglanyue@hrbeu.edu.cn)

A vector hydrophone co-locating and simultaneously measures pressure and particle, which obtains more entire information of acoustic field than pressure hydrophone, so that to render the more methods of signal processing. Bispectrum is a kind of high order spectrum analysis method, which impose a lot of information that two order statistics can't obtian. In this paper, the algorithms of bispectrum was researched based on vector hydrophone to obtain the feature of acoustic signals. Bispectrum of particle velocity and cross-bispectrum of pressure and particle velocity were analyzed. Different methods were used to compress the information measured of bispectrum and cross-bispectrum to get the signal feature quickly and effectively. The experiment was carried on based on theoretical research, and the bispectrum and cross-bispectrum were used to extract the features of radiation noise of two kinds of acoustic targets, and the pressure and vector features were obtiand. The researched results showed that Gaussian noise and no-correlated noise by bispectrum and cross-bispectrum could be compressed. The more information of sound field of targets was obtained by vector hydrophone and the distinguishing probability could be improved.

4:00–4:20 Break

4:20

4pUWa7. Research on active detection and direction finding method of autonomous underwater vehicle based on vector sensor. Qing Ling (Science and Technology on Complex Ship Systems Simulation Laboratory, Beijing 100000, shengxueli@yahoo.com.cn), Xue-li Sheng (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001), Yan-yi Yuan (Science and Technology on Complex Ship Systems Simulation Laboratory, Beijing 100000), Jia Lu, Chun-Yan Sun, and Ye Bai (Science and Technology on Underwater Acoustic Laboratory, Harbin Engineering University, Harbin 150001)

To resolve the detection and orientation problem of autonomous underwater vehicle which platform is limited, single vector sensor with orientation ability is selected. The method of detection and bearing estimation based on vector sensor and its performance are researched in this paper. Vector sonar synthetical copy correlator and vector sonar synthetical adaptive correlator are put forward here, which use both pressure and velocity information. The composing principle and realizing method of the two processors above are studied and the processing gain, detecting probability and bearing estimation ability of the two are analyzed in acoustic multi-path condition. The theoretical results simulation and experiments show that all methods studied above have better performance than common copy correlation, and in multipath condition, the vector sonar synthetical adaptive correlation does better than vector sonar synthetical copy correlation in the aspect of processing gain and bearing estimation. Keywords: active detection; direction finding; autonomous underwater vehicle; vector sensor; synthetical copy correlator; synthetical adaptive correlator

4:40

4pUWa8. Fast prediction of underwater sound scattering based on Fourier diffraction theorem with modified born approximation. Peizhen Zhang (School of Communication and Information Engineering, Shanghai University, Shanghai 200072, China, and School of Information, Guangdong Ocean University, Zhanjiang 524088, China, zpzhen7242@163.com), Shuozhong Wang (School of Communication and Information Engineering, Shanghai University, Shanghai 200072, China), Runtian Wang (Shanghai Acoustics Laboratory, Chinese Academy of Sciences, Shanghai 200032, China), Yunfei Chen (760 Research Institute, Dalian 116013, China), and Luxian Wang (School of Communication and Information Engineering, Shanghai University, Shanghai 200072, China)

It has been shown in a previous work that the directional pattern of underwater sound scattered from an object insonified by an incident plane wave can be efficiently predicted based on a reversal of diffraction tomography. The method uses the Fourier diffraction theorem with the first order Born approximation. In this paper, modification to the Born approximation is proposed by taking into account the difference in acoustic impedance between the object and water. The impedance difference was ignored in the original derivation of the Fourier diffraction theorem, leading to inaccuracy in the computation. In a two-dimensional case, the proposed modification causes a shift of a circle in the 2D Fourier transform domain, from which spectral samples are taken to give a more accurate generalized projection in the sound field. This leads to improved prediction of acoustic scattering. Extension to 3D is straightforward. The Fourier diffraction theorem with modified Born approximation is applied to produce far-field directional patterns of scattered sounds from several objects of different shapes. Comparison with the original method shows effectiveness of the proposed method. The work was supported by the Natural Science Foundation of China under the Grant No. 61071187.

5:00

4pUWa9. Lake test of 70 kHz correlation velocity log for AUV. Changhong Wang, Long Chen, Wei Qiu, and Yuling Wang (Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, wangch@mail.ioa.ac.cn)

A 70 kHz correlation velocity log (CVL) for AUV has recently been developed. The hardware design and signal processing were discussed. Localized least mean squares (LLMS) algorithm was proposed as the criterion of velocity estimation. The CVL underwent lake trials in Qiandao Lake Sea both on a ship and an AUV in Aug 2011. The CVL was calibrated. Trackline test, varying speed cruise test and drifting test were carried out to examine the performance of the CVL. The uncertainty statistics from the lake trial showed that the velocity accuracy of the CVL was about 0.8%V. This work was supported by Knowledge Innovation Project of Chinese Academy of Sciences.

5:20

4pUWa10. Examination of the complex signals distribution conditions influence. V.A. Akulichev (V.I.Il'ichev Pacific Oceanological Institute, FEB RAS Vladivostok 690041, Russia, akulich@poi.dvo.ru), A.A. Golov, S.I. Kamenev (V.I.Il'ichev Pacific Oceanological Institute, FEB RAS Vladivostok 690041, Russia), and Yu.N. Morgunov

Solving tasks of underwater acoustic communication and navigation for controlling underwater objects much depends on right hydrology and acoustic media condition estimation in operation area. Technically and economically it is worth to deploy on a operation area a stationary source of navigational and communication signals system with the range of functioning equal to operation area maximum size. For navigational system tasks each source in direct period of time beam a unique signal, which is recognized by underwater object and then propagation time and distance is been calculated. In this examination the using of a complex phase manipulated signals with carrier frequency 2000 Hz and 6000 Hz as sounding signals is been tested. These signals were used for transmitting information and navigational data. Beside their beam let to measure and examine waveguide impulse characteristics on the acoustic tracks. In current work the experiment results of informational signal transfer in case of shelf sea is shown.

5:40

4pUWa11. improved methods and results of hrbsss for mapping the deep sea bottom. Xiaodong Liu, Weiqing Zhu, Fangsheng Zhang, Dongsheng Zhang, and Gaofeng Xu (Lab of Ocean Acoustic Technology, Institute of Acoustics, Chinese Academy of Sciences, 100190, liuzhiyu@mail.ioa.ac.cn)

High resolution bathymetric sidescan sonar (HRBSSS) is designed by the Institute of Acoustic, Chinese Academic of Sciences (IACAS). HRBSSS have used the technique of direction of arrival (DOA) estimation. It can work near the sea bottom and mapping the deep sea bottom, and get the high resolution bathymetry and sidescan map, concurrently. HRBSSSs have been mounted on the DTA-6000 deep towed body (maximum work depth 6000m) and the Jiaolong HOV (maximum work depth 7000m). In order to upgrading the performance of the system, a new method for improving the depth accuracy on the nadir of the vehicle and an error correct method for

minimizing the influence of the inconsistency among the channels have been developed by IACAS. With these method, we have got the good survey results on the area of the seamount (>2000m depth) and on the area of the sea plain (>5000m depth) in the Pacific Ocean in 2011. These methods and results will be introduced in this article.

6:00

4pUWa12. Recognition of ship echo signal applying wigner distribution feature. Yuan Peng, Guijuan Li, Xin Wang, and Zhengqing Lin (Dalian Scientific Test and Control Technology Institute, Dalian 116013, China, 03081514@163.com)

Recognition of ship echo signal's difficulty lies in extracting effective feature from target reflection signal. Researches show that target echo signal is double grading with time and frequency, so time-frequency is a effective way in the field of echo signal recognition. Wigner distribution is a real means of time-frequency analysis, but it is limited by its false time-frequency spectrum called interference item. In the paper, on the basis of studying the theory of interference item generation and reduction, Choi-Williams kernel function and self frequency-window methods are applied to implement Wigner's interference reduction. Through simulation test using typical signal such as multicomponent signal, Merits and drawbacks of the two means are also analyzed. Then to merchant ship, reef and reverberation, three kinds of CW echo signal, the interesting time-frequency features are extracted. At last, sample sets of above three kinds of echo signals are divided into training and testing sample sets. The number of training samples to the number of testing samples ratio is 1 to 4. The training samples are regarded as typical sample and input to Fuzzy Adaption Resonance Theory (FART) network to train. According to typical samples, the testing samples are tested by the same network. The results show that self frequency-window is a better means to reduce interference item. But high recognition rate can be achieved if interference items exist.

THURSDAY AFTERNOON, 17 MAY 2012 S428, 2:00 P.M. TO 5:00 P.M.

Session 4pUWb

Underwater Acoustics: Underwater Acoustics Poster Session (Poster Session)

Bong-Chae Kim, Cochair
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Xiaodong Liu, Cochair
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Contributed Papers

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

4pUWb1. Performance simulation and implementation of gold sequences in underwater acoustic CDMA. Yi Tao, Zheguang Zou, and Xiaomei Xu (College of Ocean and Earth, Xiamen University, 361005, China, taoyi@xmu.edu.cn)

Underwater Acoustic Network (UAN) is becoming a hotter research area of underwater acoustic communication recently. Underwater acoustic channels are restricted by narrow bandwidth, long delay and high ambient noise, so Code Division Multiple Access (CDMA) appears to be a promising multiple access technique for UAN. However, due to low underwater data rate, only short pseudo-random sequences are available. Comparing to m sequences, Gold sequences with the same length show the same cross-correlation properties but enable networks to support more users. This paper examines the performance of short Gold sequence in underwater acoustic channel, to determine whether it is a good choice for UAN. Specifically, the paper makes contributions on the following fronts: (1) Based on a Rayleigh fading shallow water acoustic channel model, the performance of bit error rate (BER) of a 63-bit Gold-sequence up to 65 users CDMA is simulated and evaluated in MATLAB platform. (2) Underwater CDMA nodes using Gold-sequence Direct Sequence Spread Spectrum (DSSS) modulation is implemented with National Instruments' new embedded device (cRIO9076). An experiment at sea will be conducted at Xiamen Port to confirm the better performance of short Gold sequences in underwater CDMA.

4pUWb2. Technical scheme and optimization of precision timing and synchronization for underwater systems. Yu Chen (IOA, CAS, chenyu@mail.ioa.ac.cn), Haibin Wang, Xi Chen, and Dejun Wang

Real time base is one of the key features for underwater systems. However, time bias accumulates inevitably in a long-term unattended underwater system, even if high precision oscillators are used. Since it's difficult for underwater equipments to obtain timing information directly from GPS satellites, there's demanding need for effective ways of timing services. In this paper, a technical scheme based on Precision Time Protocol and virtual instruments is given. Timing and synchronization services are provided from remote time server to underwater systems. Technical optimizations are made considering the requirements of practical applications. The optimized solution has been equipped in a practical system. Experiments show it's easy to implement and with high precision, good stability and efficiency.

4pUWb3. Deployment analysis of underwater acoustic wireless sensor networks. Xia Li and Shiliang Fang (School of Information Science and Engineering, Southeast University, Si Pailou 2# Nanjing, zzhlixia@seu.edu.cn)

In this paper, underwater sensor nodes and gateway nodes deployment strategies for two-dimensional communication architecture in Underwater Acoustic Wireless Sensor Networks(UWSNs) are proposed. In the sensor nodes deployment strategy, underwater sensor nodes are deployed in two rows along the coastline, which is of complete coverage and connectivity, localization available and scalable. In the gateway deployment strategy, the gateway deployment is modelled as an optimization problem, by finding the locations of underwater gateway nodes required to achieve a given design objective, which can be minimal expected delay and minimal expected energy consumption. The OPNET network simulator is used to measure the performance of the strategies we studied. Acknowledgment of support: This work was supported by the key laboratory of underwater acoustic signal processing of ministry of education.

4pUWb4. Performance of least mean square equalizer in shallow water acoustic communication channel. Jong Rak Yoon, Kyu-Chil Park, Jihyun Park (Pukyong National University, Department of Information and Communication Engineering, Daeyon-3dong, Namgu, Busan 608-737, Korea, jryoon@pknu.ac.kr), Jungchae Shin, and Seung-Wook Lee (Hanwha Corporation Gumi Plant, 258 Kongdandong, Gumi Kyunbuk 730-030, Korea)

In shallow water, a transmitted signal is severely influenced by sea surface and bottom boundaries. Every signal to receiver except the signals through the water medium experiences a time-variant scattering in the sea surface and grazing-angle-dependent bottom reflection loss in the bottom. Consequently, the performance of underwater acoustic communication systems is degraded, and high-speed digital communication is disrupted by inter-symbol interference (ISI) effect. In this study, least mean square (LMS) algorithm equalizer is adopted to cancel out ISI effect. The equalizer is implemented in underwater acoustic communication system. Experiment is conducted in 26 m depth littoral ocean and the standard Lenna image which consists of 50x50 pixels and 8 bits per pixel is transmitted. It is verified that equalization is an efficient way to achieve high transmission rate in shallow water acoustic communication channel. This work was supported by Research Programs of Hanwha Corporation 2011.

4pUWb5. Performance analysis of anti-multipath fading underwater acoustic communication (AMF-UAC) system in the ocean environmental variability. Jangeun Kim, Taebo Shim, Euicheol Jeong (Soongsil Univ, sentije@ssu.ac.kr), and Youngnam Na (Agency for Defense Development)

Due to surface and bottom space constraints of the underwater acoustic channel, multi-path fading occurs and causes degradation of communication. Multi-path fading depending on the maximum delay time(Tm) and symbol period(Ts) can be divided into two kinds of channels. In this paper, we propose Anti-Multipath Fading Underwater Acoustic Communication(AMF-UAC) System. This system estimates the kind of channel and distinguishes flat fading (Tm<Ts) from frequency selective fading (Tm>Ts) under the ocean environmental variability. After checking the channel status, this system selects a mitigation technique depending on the type of multipath fading. In order to verify AMF-UAC system performance, we test transmission of image using 43.2kbit of gray image through the multi-path fading channel. Test results show that the number of bit errors is reduced from 300 to 10 under flat fading channel and from 20,000 to 90 under frequency selective fading channel when the reference SNR is 14dB.

4pUWb6. A simple distributed networking protocol for underwater acoustic networks. Yongfeng Wang, Yongqing Wu, Chunhua Zhang, and Huizhi Cai (Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, wyf@mail.ioa.ac.cn)

A simple distributed networking protocol for underwater acoustic networks is proposed. Each node in the network learns about the degrees of its separation to a certain destination node by passively analysis the packets transmitted from its immediate neighbors. The address of neighboring node with the least hops to a certain destination node will then be saved in a route table as the relay address for later communication with that destination. The route table is initialized with broadcast address and the relay address to a destination node will be reset to broadcast address if no packets from the destination have been heard through the corresponding neighbor for a given time. By applying such a decay mechanism, the routing information is locally updated to keep up with the topology changes of the network due to mobility, node failure, and channel variations. There is little overhead traffic for networking and the network-wide flooding only happens in the initialization process, the proposed protocol is energy efficient. The details of the protocol and evaluation and simulation results are given in the paper.

4pUWb7. Geoacoustic characteristics of P-wave velocity in Donghae City–Ulleung Island line, East Sea (Sea of Japan). Woo-Hun Ryang (Chonbuk National University, Jeonju Jeonbuk 561-756, Republic of Korea, ryang@jbnu.ac.kr), Seong-Pil Kim (Korea Institute of Geoscience and Mining Resources, KIGAM), and Jin-Hyuk Choi (Agency for Defense Development, ADD)

Donghae City–Ulleung Island Line (DC-UI Line) is a representative line for underwater and geoacoustic modeling in the middle western East Sea (Sea of Japan). In this line, an integrated model of P-wave velocity is proposed for a low-frequency range target (<200 Hz), based on high-resolution seismic profiles (2–7 kHz sonar and air-gun), shallow and deep cores (grab, piston, and Portable Remote Operated Drilling), and outcrop geology (Tertiary rocks and the basement on land). The basement comprises 3 geoacoustic layers of P-wave velocity ranging from 3750 to 5550 m/s. The overlying sediments consist of 7 layers of P-wave velocities ranging from 1500 to 1900 m/s. The bottom model shows that the structure is very irregular and the velocity is also variable with both vertical and lateral extension. In this area, seabed and underwater acousticians should consider that low-frequency acoustic modeling is very range-dependent and a detailed geoacoustic model is necessary for better modeling of acoustic propagation such as long-range surveillance of submarines and monitoring of currents. This research was supported by Basic Science Research Program through the National Research Foundation of Korea (NRF) funded by the Ministry of Education, Science and Technology (2010-0025733) and by the Ministry of Knowledge Economy through the grant of Marine Geology and Geophysical Mapping Project (GP2010-013).

4pUWb8. Hybrid geoacoustic inversion scheme with an equivalent seabed model. Zhenglin Li and Renhe Zhang (State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China, lzhl@mail.ioa.ac.cn)

Acoustic propagation in shallow water is greatly influenced by the properties of the bottom. The purpose of geoacoustic inversion is estimation of ocean bottom acoustic parameters such as sediment sound-speeds, densities, and attenuations from measured acoustic fields. It is a helpful supplement to direct measurements. Especially, geoacoustic inversion could give low frequency attenuation, which cannot be measured by coring the sediment. Therefore, it has been paid much attention in recent years. A hybrid geoacoustic inversion scheme, which combines several inversion methods together to invert for the bottom parameters, has been proposed based on the fact that the bottom acoustic parameters have different sensitivities to the different physical parameters of acoustic field. This inversion scheme could avoid the problem of the multiple solutions, which are often companied with some geoacoustic inversion methods. The validity of the inversion scheme is verified in a series of sea experiments at different sites. The inverted bottom parameters could be used to forecast the sound propagations and distinguish the atlas marked bottom type quite well. The mapping relation between the sediment types and the acoustic parameters are also given. [Work supported by the National Natural Science Foundation of China under Grant No. 10974218 and Grant No. 10734100]

4pUWb9. Robust wideband adaptive beamforming using waveguide invariant focusing method. Biao Jiang (Hangzhou Applied Acoustics Research Institute, 96 Huaxing Road, Hangzhou, 310012, China, jiangbiao@sina.cn)

The waveguide invariant describing the dispersive propagation in underwater environment can provide useful information for signal processing. In this paper, waveguide invariant focusing is exploited to preprocess the received horizontal array signal, such that the moving target is aligned in a single rank-one signal subspace over the bandwidth, results a reduction of the number of the snapshots necessary for the adaptive beamforming, and the diagonal loading is optimized using the robust Capon method to further improve the robustness. Numerical results show that the proposed method can improve the detection performance with limited observation time. Moreover, passive ranging is accomplished when the beamforming output achieves the maximum over a scanned target range limit.

4pUWb10. The relationship between normalmode interference structure and waveguide invariant. Liu Fuchen (Huaxing Road, Hangzhou, China, fuchen-liu@163.com)

In shallow water, waveguide invariant describes the dispersive characteristics of the filed.It relates the normal group slowness and phase slowness of normal modes.In this paper, waveguide invariant changes with normal mode and interference patterns difference are researched. the relationship formula between waveguide invariant and horizontal wavenumber of normal mode is deduced, which is valided by comparision between theorical value with theorical formula with the value from new formula. Correspondingly, the conclusion is drawn that waveguide invariant is decreasing with horizontal wavenumber decreasing, with is also valided by simulation.

4pUWb11. Fluctuation of acoustic signals due to internal waves in the East Sea of Korea. Jooyoung Hahn, Joung-Soo Park, HyoungRok Kim, Woogeun Chon, Haksue Lee, and Young-Nam Na (Agency for Defense Development, hahnjy@add.re.kr)

This study attempts to investigate the fluctuation of underwater acoustic signals due to internal waves (IWs) off the east coast of Donghae, Korea. Sea experiment was performed with thermistor strings, a sound source, and an array of hydrophones. Based on the thermistor string data, the IWs have characteristics of typical periods of 10-20 minutes, amplitudes of 10-20m, and a duration of 1-2 hours. The IWs were analyzed as they moved from offshore to the coast at a speed of 70 cm/sec. Underwater acoustic signals (CW 80 - 800Hz) also show obvious energy fluctuations with the IWs. Through an analysis of these acoustic signals, fluctuations of periods of 15 minutes are located in time domain. As mixed layer depth varies with time, it may cause travel time difference of acoustic signals. This travel time differences causes fluctuation of acoustic signals in range-independent stratified ocean structure. The spectrum characteristics of the acoustic signals show the possibility that acoustic waves may react to the IWs through mechanisms such as mode coupling and travel time fluctuation.

4pUWb12. Experimental study for short-range acoustic field fluctuations and time-space related characteristics in shallow. ZhongCheng Ma and QingGang Cao (Dalian Scientific Test and Control Technology Institute, Dalian 116013, China, ma_zc@yahoo.com.cn)

Short-range sound propagation characteristic in shallow is an important basis for the underwater test parameters settings, and it is also an important basis for the data summarized and analysis. The interference of the interface reflection and the direct wave in water makes shallow water acoustic show complex time-space variability. By a series of experimental design, fluctuations and time-space characteristics variation in short-range sound field were analysis. Statistical characteristics of the sound field fluctuations and variability characteristics were obtained.

4pUWb13. The effect on the propagation in shallow water with a mega-ripple. Sungho Cho, Donhyug Kang, Young-Kuk Lee (Korea Ocean Research and Development Institute, Ansan, Korea, shcho@kordi.re.kr), and Jee Woong Choi (Department of Environmental Marine Sciences, Hanyang University, Ansan, Korea)

It is well known that acoustic propagation in shallow water is greatly influenced by an interaction of reflection, transmission, and scattering with the boundary condition. Especially, acoustic propagation of low frequency is depended on the structures of seabed and sub-botttom layer. In this study, acoustic propagations for low-frequency in shallow water were measured from field experiments under the condition of mega-ripple. The study area formed various mega-ripple caused by strong tidal current. Precisely bathymetry and sub-bottom structure were obtained from multi-beam and sparker system. In the experiments, acoustic source and receiver for propagation measurement were single bulb system and multiple self recording hydrophone system, respectively. From the measurement, we described characteristics of acoustic transmission for low frequency around the mega ripple regions. [Supported by KORDI (Korea Ocean Research & Development Institute)]

4pUWb14. Research on waveform design of sinusoidal frequency-modulated to extract target phase feature. Wu Yongqing (Institute of Acoustics, Chinese Academy of Sciences, No. 21, Bei-Si-huan-Xi Road, Beijing, China, wyq@mail.ioa.ac.cn)

The majority of active sonar systems to detect and classify a target based on the amplitude of the received echo strength or the induced Doppler shift. However, additional classification information is available from the phase shift introduced by some targets as a result of the acoustic boundary conditions. In this paper, waveform design of sinusoidal frequency-modulated (SFM) based on the use of sub-band correlators is presented for measuring the phase shifts associated with certain stationary and moving targets when insonified by broadband transmissions. With the aim of providing improved range resolution, it maintains the amplitude of the transmission constant to maximize energy efficiency and compatibility with existing nonlinear power amplifiers. And the influences on target echo phase measurement are analyzed from reverberation gain achievable with broadband, Doppler sensitive SFM transmissions in littoral waters. At last, field trial results are given for a mine-like classification sonar system and a forward-looking sonar system designed for operation near the sea surface.

4pUWb15. Baseband code estimation for underwater acoustic phase encoding signal. Xiaoyan Wang, Shiliang Fang, and Li Wang (Key Laboratory of Underwater Acoustic Signal Processing of Ministry of Education, Southeast University, No. 2, Sipailou, Nanjing 210096, China, xyanwseu@gmail.com)

A method based on phase jump is proposed in this paper to solve the problem of the baseband code extraction from the underwater acoustic phase encoding signal under non-cooperative conditions. Phase estimation is conducted to the baseband signal, which is demodulated from the received signal according to the estimated carrier frequency. Besides, data smoothing processing is performed twice to reduce the noise effect. Then the baseband code can be estimated by using phase jump of the baseband signal. In practical application, the code sequence is difficult to directly estimate by using the baseband signal waveform due to the distinct amplitude distortion caused by the complexity of underwater acoustic channel. However, the approach based on phase jump is not sensitive to the amplitude fluctuation. So it is especially suitable for the estimation of baseband code form the underwater acoustic phase encoding signal. Computer simulations and experiments in the lake verify the feasibility of this method.

4pUWb16. Sonar detection performance analysis with environmental uncertainty using vertical array data. Xinmin Ren (College of Information Science and Technology, Ocean University of China, 238 Song-Ling Rd, Qingdao 266100, China; The Institute of Acoustics, Chinese Academy of Sciences, 21 West Road of Beisihuan, Beijing 100190, China, qdpeople-2008@163.com), Guofeng Zheng (College of Information Science and Technology, Ocean University of China, 238 Song-Ling Rd, Qingdao 266100, China), Qihu LI, Guiqing Sun, Haining Huang (The Institute of Acoustics, Chinese Academy of Sciences, 21 West Road of Beisihuan, Beijing 100190, China), and Ju Lin (College of Information Science and Technology, Ocean University of China, 238 Song-Ling Rd, Qingdao 266100, China)

Sonar detection performance is related to ocean environmental parameters, such as the source position, the ocean depth, the sound speed profile and geoacoustic parameters, etc. These parameters have strong spatial and temporal variability, which result to environmental uncertainty. The sonar detection system can be limited by the presence of environmental uncertainty. Based on a statistical model of the environmental uncertainty, the optimal Bayesian predictor by L. Sha has been applied in this paper to analyze the effects of environmental uncertainty on detection performance using vertical array data collected in two experiments. The first experiment took place in shallow water off the Italian west coast by the NATO SACLANT Center in 1993(SACLANT Sonar Data). The second experiment took place in shallow water in China in 2008(LOFAR'08 data). Quantitative effects of various uncertain parameters on detection performance have been illustrated to evaluate which one is the most sensitive and which one is insignificant. The present work is supported by the National Defense Fundamental Fund of China (No.613xxxxx).

4pUWb17. Analysis of influence of line array beamforming on the target modulation feature. Xinwei Luo (2# Sipailou, Nanjing 210096, China, luoxinwei@seu.edu.cn)

In the line array sonar, the target signal was obtained after beamforming processing. The beamforming processing could improve the Signal to Noise Ratio (SNR) by using array signal's spatial processing gain. But the distortion would be brought in signal characteristics under non-ideal conditions during line array beamforming in practical application, which increased the difficulty of target characteristics extraction. As an important basis for target classification and recognition, the performance of modulation characteristics detection and extraction is affected by SNR, signal bandwidth, modulation depth and other factors. In this paper, the influence of the time delay error on target characteristics was analyzed based on the conventional beamforming of linear array. And then a method with the combination of theroy and simulation is proposed to analyze the influence of plane wave assumption, azimuth estimation error and disturbance of array shape on modulation characteristics.

4pUWb18. Simulation of time reversed acoustic inversion in shallow water. Bok Kyoung Choi, Byoung-Nam Kim, Bong-Chae Kim, Seom-Kyu Jung, and Donhyug Kang (Korea Ocean Research and Development Institute, Ansan, Korea, bkchoi@kordi.re.kr)

For simulation to time reversed acoustic inversion in any enclosure system as shallow water, we must know the impulse response of the system. Throughout this process we can reproduce an original signal at focal point by convolution of time reversed inversion. To apply this TRA simulation to shallow water environment, after extracting the impulse response about simple condition as shallow water, we analyze the acoustic focusing by TRA inversion. In result, the spatial focusing and time signal patterns can be built by TRA simulation for shallow water condition. [Supported by KORDI (Korea Ocean Research & Development Institute)]

4pUWb19. Spherical Acoustic Lens for 3D Imaging SONAR. Shinta Takano, Teiichiro Ikeda, Kunio Hashiba (Hitachi Ltd., Central Research Laboratory, 1-280 Higashi-koigagubo, Kokubunji-shi, Tokyo 185-8601, Japan, shinta.takano.zo@hitachi.com), Shinsuke Sato, Kentaro Kato, and Mitsuhiko Nanri (Hitachi Ltd., Defense Systems Company, 216 Totsuka-cho, Totsuka-ku, Yokohama 244-8567, Japan)

The states of the structures within a harbor after underwater construction or after a disaster are mainly visually checked and confirmed by divers. A new method that is independent of the water turbidity is desired to replace this direct diver necessity. In this study, we propose a spherical acoustic lens consisting of two or more concentric layers for use in a real-time three-dimensional imaging SONAR, which has many advantages. It does not require electronic circuitry for the beam forming, and it has a wide field of view and can simultaneously collect signals from various directions. The acoustical characteristics of the spherical lens are analyzed under conditions in which the speed of sound of the lens materials by using a finite difference time domain (FDTD) method. The results show that the spherical lens has a narrower receive beam profile and a higher receive gain than conventional compound lenses consisting of three aspheric lens.

4pUWb20. Time-reversal passive localization for underwater radiated noise source. Liang An, Lijun Chen, and Shiliang Fang (Key Laboratory of Underwater Acoustic Signal Processing of Ministry of Education, Southeast University, Nanjing 210096, China, an_liang@seu.edu.cn)

Time reversal (TR) is a technique of passive acoustical sources localization using a time reversal mirror (TRM) and is especially useful in multi-path environments. TR is commonly used to localize noise sources in aero acoustics and in many cases pulsed waveforms are used. A TR passive localization method for long duration waveforms was presented in this paper. The TR is applied to actual radiated noise recorded by only one hydrophone to estimate the source distance and depth. Compared to a standard time reversal approach, the proposed technique is based on the virtual time reversal mirror which can match the acoustic channel automatically and leads to an adaptive spatial focusing and temporal compressing as the conventional TRM. The processing result of simulated data and experiment data show this method could mitigate the multi-path interference on passive localization.

4pUWb21. A delay-sensitive gts allocation scheme in ieee 802.15.4 for wireless audio applications. Ying Wang, Peng Zhang, and Jun Yang (Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, No. 21, North 4th Ring Road West, Haidian District, Beijing 100190, P.R. China, wangying_lab@yahoo.com)

The IEEE 802.15.4 standard is designed for low rate wireless personal area networks, targeting at low cost and low power communications. It also provides a guaranteed time slot (GTS) mechanism to support time sensitive wireless audio applications. The coordinator allocates specific durations within a superframe to guarantee reliability and performance of data deliveries. However the existing first-come-first-served (FCFS) GTS allocation policy can result in the scheduling inflexibility in low-latency data transmission, since it doesn't take into account the traffic specification, delay requirements and the energy resources. In this paper, an adaptive GTS allocation scheme is developed to satisfy the delay constraints of wireless audio applications. Based on recent GTS usage feedback, each device in the WPAN is dynamically allocated a priority number. Then the devices are allocated specific slots according to this priority information. The devices with higher priorities indicate more recent traffic and have higher probabilities to transmit their data in the subsequent superframe. The performance of the scheme is analyzed using Markov chain. The analytical model is validated through simulation results, indicating that the proposed adaptive scheme has better performance for audio transmissions.

4pUWb22. The improved methods and results of HRBSSS for Mapping the Deep Sea Bottom. Xiaodong Liu, Weiqing Zhu, Fangsheng Zhang, Dongsheng Zhang, and Gaofeng Xu (100190, liuxd@mail.ioa.ac.cn)

High resolution bathymetric sidescan sonar (HRBSSS) is designed by the Institute of Acoustic, Chinese Academic of Sciences (IACAS). HRBSSS have used the technique of direction of arrival (DOA) estimation. It can work near the sea bottom and mapping the deep sea bottom, and get the high resolution bathymetry and sidescan map, concurrently. HRBSSSs have been mounted on the DTA-6000 deep towed body (maximum work depth 6000m) and the Jiaolong HOV (maximum work depth 7000m). In order to upgrading the performance of the system, a new method for improving the depth accuracy on the nadir of the vehicle and an error correct method for minimizing the influence of the inconsistency among the channels have been developed by IACAS. With these method, we have got the good survey results on the area of the seamount (>2000m depth) and on the area of the sea plain (>5000m depth) in the Pacific Ocean in 2011. These methods and results will be introduced in this article.

4pUWb23. Measurement of propeller-induced cavitation noise for ship identification. Endang Widjiati, Eko Budi Djatmiko, Wisnu Wardhana, and Wirawan (Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia, ewidjiati@na.its.ac.id)

Acoustic cavitation noise caused by propeller is used in many underwater applications as one of the ships noise signatures. This paper reports characterization process done by measuring the cavitation noise generated by a propeller in a cavitation tunnel. The cavitation tunnel used for the measurement is of the K16B type belonging to the Indonesian Hydrodynamic Laboratory, Surabaya, Indonesia. The experiment is done using a B-series four-blade bronze propeller (diameter 23cm) using a hydrophone placed in the tunnel at window section (60x90)cm^2 of the measurement section (400x85x85)cm^3. Some of the solutions to get accurate and reliable measurement results include calibration measurement and minimizing acoustic noise system and environment. Preliminary process has been done to analyse the characteristics of the measurement results in the time-frequency domain, with the objective being to detect when and which type of cavitation noise occurs in any kind of condition. Measurement outcomes in the form of acoustic data signals are obtained in different conditions by varying the water pressure, flow velocity and propeller speed rotation.

4pUWb24. Underwater signals' characteristics of power spectrum based on data mining. Qingfu Wang (Science and Technology on Sonar Laboratory, Hangzhou Applied Acoustic Research Institute Hangzhou 310012, P.R. China, wangqfu@sohu.com), Shuanping Du, and Huiliang Ge

Power spectrum was always utilized for underwater signal recognition. Distance based data mining was applied to feature extraction of power spectrum. The distances within and between classes were first calculated, then, the weight for each frequency bin was get according to its discrimination, finally, the weighted power spectrum was used to feature extraction and classification. All kinds of signals were generated with underwater acoustic propagation model through wave guide. Experiments and tests were conducted and performance of the method was verified.

4pUWb25. Research on the reverberation measuring technique of underwater directional sound sources. Lee Chi, Shang Dajing, and Zhang Lin (Harbin Engineering University, leechi819@yahoo.com)

The radiated noise of underwater directional sound sources was generally measured in the corresponding ocean environmental condition. Influenced by the ocean environmental noise, the reflection of ocean bottom and ocean surface, it is very difficult to get the radiated sound power of underwater directional sound sources. Reverberation method used in the Architectural acoustics was taken in this paper for measuring the radiated sound power of underwater directional sound sources. It has been used in the radiated sound power measurement of underwater spherical sound sources, but never been used in underwater directional sound sources. The reverberant theoretical formula of directional sound sources was derived by introducing the directivity factor. The radiated sound power of directional sound sources such as dipole and the two same-phase sphere was measured. The measuring and analyzing results show that the reverberant radius of directional sound sources is related to directivity factor. With the directivity factor turns strong, the reverberant radius will become larger but the reverberant controlled area will become smaller, so the measurement range of spatial averaging will become small. The radiated sound power of directional sound sources such as dipole and two same-phase spheres can be measured by large-range of spatial averaging. The measurement techniques can apply to measure the radiated sound power of underwater complex sound sources.