Session 2aAAa

Architectural Acoustics: Acoustics of Concert Halls II

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Contributed Papers

8:00

2aAAa1. Acoustical design consideration for the new IFEZ concert hall. Jin Yong Jeon (Hanyang University, Department of Architectural Engineering, 133-791 Seoul, Republic of Korea, jyjeon@hanyang.ac.kr), Yong Hee Kim (Hanyang University, Department of Architectural Engineering, 133-791 Seoul, Republic of Korea, kimyonghee@gmail.com), Shin-Ichi Sato (Hanyang University, Department of Architectural Engineering, 133-791 Seoul, Republic of Korea, s_sato@mac.com)

The design of a new concert hall for the Incheon Free Economic Zone (IFEZ) Arts Center in Korea is in progress. The hall is supposed to be the home of the Asia Philharmonic Orchestra. It has over 1,700 seats, including 150 choirs. A vineyard seating arrangement will be applied, but the seats around the stage area will be minimized in order to form a solid stage enclosure. Lateral walls are designed for each seating block to increase acoustical intimacy. Average width between lateral walls has been designed to be less than 15 m, and every seat is arranged within 7.5 m to the closest lateral wall. All lateral walls are designed to be inclined to lead first reflections and to improve spatial impression at the audience area. Diffusers are functionally installed to the effective reflecting surfaces from the source on stage. The design considerations are investigated through both computer simulation and scale modelling.

8:20

2aAAa2. Objective and subjective analysis of acoustical response in newly renovated Palais Moncalm, Quebec City, Canada. Jean-Philippe Migneron (Lab. d'acoustique, École d'architecture, 1, Côte de la Fabrique, Vieux Séminaire, Université Laval, Québec, QC G1K 7P4, Canada, jean-philippe.migneron.1@ulaval.ca), Jean-Gabriel Migneron (Lab. d'acoustique, École d'architecture, 1, Côte de la Fabrique, Vieux Séminaire, Université Laval, Québec, QC G1K 7P4, Canada, jgmigneron@hotmail.com), Jean-François Hardy (Lab. d'acoustique, École d'architecture, 1, Côte de la Fabrique, Vieux Séminaire, Université Laval, Québec, QC G1K 7P4, Canada, jean-francois.hardy.1@ulaval.ca)

To celebrate the 75th anniversary of Palais Montcalm, the building has been almost entirely rebuilt to convert it into the House of Music. The main room, named Salle Raoul-Jobin, has been designed specifically for the resident chamber orchestra: Les Violons du Roy. Collaboration between acoustical consultant, Larry S. King, and architect Jacques Plante of MUSE consortium results in a 979 seats concert hall with variable acoustic to accommodate different kinds of musical events. Since reopening in March 2007, the general acoustic has been enthusiastically acclaimed by professionals, by world known musicians, and by the public. The principal objective of this study is to characterize acoustical response of the room with various adjustments of wall curtains and motorized canopy. This important reflective area is software-controlled in a range of 3 meters depending on musical needs. Acoustical objective parameters, such as reverberation time, early decay time, impulse response or C80 factor will be analyzed and compared to musical subjective perception of a few instruments played at multiple positions on stage and for different listening locations.

8:40

2aAAa3. New design tendencies in modern concert hall design. Alban A. Bassuet (Arup Acoustics, 155 avenue of the americas, New York, NY 10013, USA, alban.bassuet@arup.com)

New concert hall design trends are emerging as seen from recently completed halls around the world and recent international architectural competitions. Analyzing the current situation, the paper starts by differentiating the acoustical characters of major traditional concert hall forms such as the large shoe-box, the Vineyard and the surround hall. It discusses in particular the balance of acoustical energy that characterize these different basic hall shapes, the sensation of intimacy in relation to the listener's distance to the performance area and the different early reflection patterns inherent from these hall shapes. As an example of modern concert halls design tendencies, the paper then describes a design option chosen by Arup Acoustics for the Paris Concert Hall Architectural design competition with Zaha Hadid architect. The paper describes the development of the form and shape of the hall in response to the brief, the concept chosen for the design of sending/receiving surfaces to improve the early reflection ″efficiency″ and ″stability″ and the use of an overhead reflector to improve the balance between soloist and orchestra and the balance between the orchestra and the reverberation in the hall.

9:00

2aAAa4. Acoustic design and evaluation of a multi-purpose hall of a new conference centre. Attila Balázs Nagy (Kotschy Bt., Álmos vezér u. 4, 2045 Törökbálint, Hungary, nagyab@hit.bme.hu), Ferenc Tamás (Kotschy Bt., Álmos vezér u. 4, 2045 Törökbálint, Hungary, tamas.ferenc@kotschy.hu), András Kotschy (Kotschy Bt., Álmos vezér u. 4, 2045 Törökbálint, Hungary, bandi@kotschy.hu)

At Forum Acusticum 2005 the building and room acoustic design of a new Conference Centre was presented. Since then the construction work has been finished and the Centre is now open. The Conference Centre itself is a multi-functional building, having three wings of different functions. The Main Hall of the Centre is a room for 750 people which can be extended with adjacent section rooms for 1100 persons, giving a total volume of 14000 m^3. It was designed to host conferences, lectures, and - most of the time - to be used as a concert hall. The extremely different acoustical demands of being a concert hall and a lecture room had been fulfilled with appropriate room acoustic design and with variable acoustics by employing a DCR (Digital Control of Reverberation) system. In this paper we give a report on the achieved acoustical performance of the Main Hall of the Conference Centre. We have performed extensive room acoustic measurements in the Main Hall, the results of which are compared to the designed values and are presented in this paper. The design of the DCR system is discussed in a different paper.

9:20

2aAAa5. Renovation of the concert hall The Doelen: A case study on the impact of a stage canopy on stage and room acoustics. Margriet R. Lautenbach (Peutz BV, PO Box 696, 2700 AR Zoetermeer, Netherlands, m.lautenbach@zoetermeer.peutz.nl), Martijn Vercammen (Peutz, De Grippen 1124, 6605 TA Wijchen, Netherlands, m.vercammen@mook.peutz.nl), Klaus-Hendrik Lorenz-Kierakiewitz (Peutz BV, Kolberger Strasse 19, D-40599 Duesseldorf, Germany, khl@peutz.de)

When opened in 1966, the main concert hall in De Doelen, Rotterdam, the Netherlands, was provided with six canopies above the stage platform. Their function was twofold: to provide a large part of the audience with early reflections; to create good ensemble conditions for the musicians on stage. Despite good reviews after the opening, a few years later the canopies were removed to get rid of unwanted reflections at the recording microphones positions just below the canopy. Since then, a significant percentage of the orchestra is unhappy about the acoustic conditions on stage. During the design process of the renovation, possibilities to re-introduce a stage canopy are investigated. Objective acoustic parameters obtained by carrying out measurements in the hall as well as in a 1:10 scale model and by calculations with a ray-tracing computer model are examined. Parallel, three questionnaire rounds gave an impression about the musicians' subjective judgement about the stage acoustics. In this paper the acoustic differences of three situations (without canopy, with original canopies and with new canopy design) and the search for an optimal balance between improving the stage acoustics without altering room acoustical conditions in the audience are discussed.

Invited Paper

9:40

2aAAa6. Evaluation of virtual acoustic stage support for musical performance. Wieslaw Woszczyk (McGill University, Schulich School of Music, 555 Sherbrooke Street West, Montreal, QC H3A 1E3, Canada, wieslaw@music.mcgill.ca), William L. Martens (McGill University, Schulich School of Music, 555 Sherbrooke Street West, Montreal, QC H3A 1E3, Canada, wlm@music.mcgill.ca)

The effects of performance space acoustics on musical performance can be evaluated most effectively by eliminating the influence of powerful non-auditory factors, such as the visual appearance of the performance space. To allow for such relatively unbiased evaluations, a virtual acoustic stage support system was set up for a live musical performance, and the performer was asked to make blind comparisons between a variety of architectural acoustic simulations. While results support the broad generalization that the preferred reverberation time for acoustical stage support depends upon the piece of music to be performed, it was also clear that preferences strongly depended upon performers' aural familiarity with architectural acoustic spaces in which they had considerable experience in previous performances. After some exposure to descriptive analysis techniques, performers could explain their preferences in terms of the perceptual characteristics that differed between presented aural architectures.

TUESDAY MORNING, 1 JULY 2008 ROOM 253, 8:00 TO 9:00 A.M.

Session 2aAAb

Architectural Acoustics and Noise: Low Frequency Absorption: Mechanisms, Measurement Methods and Application II

Peter D'Antonio, Cochair
RPG Diffusor Systems, Inc., 651-C Commerce Drive, Upper Marlboro, MD 20774, USA

Christian Nocke, Cochair
Akustikbüro Oldenburg, Katharinenstr. 10, Oldenburg, 26121, Germany

Contributed Papers

8:00

2aAAb1. Effects of Low-frequency Absorption on Perceived Tightness of Bass Imagery in Music Reproduction. William L. Martens (McGill University, Schulich School of Music, 555 Sherbrooke Street West, Montreal, QC H3A 1E3, Canada, wlm@music.mcgill.ca)

In order to enable unbiased observation of the effects of low-frequency absorption on auditory imagery associated with multichannel loudspeaker reproduction, binaural recordings were made of surround sound program material that was reproduced over full-range loudspeakers located in a room that was specially constructed to allow for variation in low-frequency acoustical treatment. These recordings were then presented via headphones to allow for double-blind comparison of the variation in auditory imagery associated with selected changes in room acoustics while holding listener and loudspeaker locations constant. Several perceptual attributes were examined, but the listeners were able to make the most clear distinctions between auditory spatial images in terms of the attribute identified as the ″perceived tightness of bass imagery.″ Analysis of the signals presented to the listeners' ears in these binaurally-reproduced multichannel music samples showed that the tightest bass imagery was associated with high values of interaural coherence, with lower values producing more ″muddy″ bass imagery. [Work supported by Canada Foundation for Innovation.]

8:20

2aAAb2. Acoustical tests of custom-made, low-cost bass absorptive treatment for small rooms. Ioana Pieleanu (Acentech Incorporated, 33 Moulton Street, Cambridge, MA 02138, USA, ipieleanu@acentech.com), Jeffrey Fullerton (Acentech Incorporated, 33 Moulton Street, Cambridge, MA 02138, USA, jfullerton@acentech.com), Marc Choiniere (University of Nebraska, 101A Peter Kiewit Institute, Omaha, NE 68182, USA, mchoiniere@gmail.com)

Small music classrooms, practice rooms or small budget recording studios and control rooms are a few of the numerous applications where low frequency absorption is sought and necessary for a well-balanced acoustical spectrum. Often, low frequency absorption is achieved by using pre-engineered products, such as tuned panel resonators. Unfortunately, the cost of these pre-engineered products can be excessive for small budget projects,

or projects that involve a large number of such spaces. For these applications, more affordable, custom-made low frequency absorbers are desired. Several types of custom-made treatments were physically tested, to determine if the degree of low frequency absorption they provide would make them an effective replacement for the pre-engineered counterparts. The test samples included glass fiber panels mounted as corner traps, custom-made plywood panel resonators and others. The testing was conducted in a space comparable in size to a small studio room/control room, or a medium practice room. This presentation describes the test method and measurement results.

8:40

2aAAb3. The importance of bass clarity in pop and rock venues. Niels W. Adelman-Larsen (Flex Acoustics, Diplomvej 377, 2800 Lyngby, Denmark, nwl@flexac.com), Eric R. Thompson (Centre for applied hearing research, Technical University of Denmark, DTU, Bygn. 352, 2800 Lyngby, Denmark, et@oersted.dtu.dk)

High levels of bass sound have been shown to stimulate the part of the brain that controls such basic instincts as sexual desire and hunger [Todd, 2000]. In rock and pop music, the bass frequencies from 40-125 Hz get amplified to very loud levels. Easily half of the electrical power of the PA and monitor system goes to these 1.5 octaves. A recent survey [Adelman-Larsen et al., 2007] showed that the most important subjective parameter for a rock and pop music hall to score a high rating was 'bass clarity' which correlated with a coefficient of 0.74 to 'overall impression'. Informal discussions with audio engineers and bass players give the perspective that artificial reverberation is rarely, if ever, added to bass-frequencies. In fact the ideal hall should be as dry as possible at low-frequencies. In the mid-treble frequency range, sound absorption, and thereby 'clarity', is easily obtained through the presence of the audience that absorbs 4-6 times more mid/high frequency sound energy than bass sound energy. In the low-frequency range 'clarity' is not so easily obtained. This paper discusses the challenge in depth and proposes design solutions.

TUESDAY MORNING, 1 JULY 2008 ROOM 202/203, 8:20 A.M. TO 12:40 P.M.

Session 2aAAc

Architectural Acoustics and Engineering Acoustics: Acoustics and Electroacoustics of Small Rooms

Jiri Tichy, Cochair
Pennsylvania State University, 5552 N. Citation Road, Toledo, OH 43615, USA

Mendel Kleiner, Cochair
Chalmers Rm. Acoust. Group, Dept. of Appl. Acoust., Chalmers Univ. of Technology, Gothenburg, 41296, Sweden

Invited Papers

8:20

2aAAc1. Investigation of Bonello criteria as a practical tool in small room acoustics. Todd Welti (Harman International Industries, Inc., R&D Group, 8500 Balboa Blvd, Northridge, CA 91329, USA, twelti@harman.com)

The Bonello criteria are often used as an objective metric for assessing room dimensions with respect to modal response. In reality, the modal response is not simply a property of the room dimensions, but is quite dependent on source/receiver locations as well. The legitimacy of Bonello's criteria when used in rooms with typical subwoofer and seating configurations has not been well investigated. In this study, a room model is used to calculate room responses for a large number of combinations of room dimensions, subwoofer positions, and seating locations. The Bonello criteria are evaluated for each combination and compared to perceptually relevant criteria such as flatness and seat to seat consistency of the calculated acoustical responses.

8:40

2aAAc2. Region of control for low frequency modal equalisation in small listening rooms. Aki Mäkivirta (Genelec Oy, Olvitie 5, FIN-74100 Iisalmi, Finland, aki.makivirta@genelec.com)

Small-size listening rooms are characterized by sparse modal density with pronounced spectral colouration when the decay times at low frequencies are large. Various active approaches for reducing the modal decay time have been suggested in the literature. A review of the salient principles of these approaches is given. Active control of the modal decay time at the primary listening position by using the primary radiator for control is demonstrated and evaluated for performance. The locality and size of the region of control, and the amount of control achieved, are studied in light of the experiment. Effect of the modal equalizer filter to the perceived response flatness, audibility of low frequency resonances, and the required amount of modal equalization are discussed in light of the experiment.

9:00

2aAAc3. Improving room acoustics at low frequencies with multiple loudspeakers and time based room correction. Sofus Birkedal Nielsen (Aalborg University, Fredrik Bajers Vej 7 B, 9220 Aalborg Ø, Denmark, sbn@es.aau.dk), Adrian Celestinos (Oticon A/S, Kongebakken 9, 2765 Smørum, Denmark, adc@oticon.dk)

Small and medium size rectangular rooms are often used for sound reproduction. These rooms have substantial acoustical problems at low frequencies primarily caused by the reflections from the room boundaries. The spatial variation in sound pressure level (SPL) can be up to 30 dB in a room at low frequencies, and appear not only at modal frequencies. The problem is an acoustical issue in time, and should therefore be analyzed in the time-domain, instead of the traditional steady state frequency domain. The construction of a finite-difference time-domain approximation program (FDTD) has lead to a simple and untraditional solution called CABS (Controlled Acoustical Bass System) that makes use of multiple loudspeakers. With the proper placement of low frequency loudspeakers, CABS can create a plane wave from the front wall which will be absorbed by additional low frequency loudspeakers at the back wall. With the back wall reflection removed a homogeneous sound field will be created in the whole room at low frequencies. Simulations and measurements of normal size listening rooms show that 4 loudspeakers are enough to even the sound field in a room. The CABS system is controlled by a developed DSP system.

9:20

2aAAc4. Applications of a plane wave based room correction system for low frequencies using multiple loudspeakers. Adrian Celestinos (Oticon A/S, Kongebakken 9, 2765 Smørum, Denmark, adc@oticon.dk), Sofus Birkedal Nielsen (Aalborg University, Fredrik Bajers Vej 7 B, 9220 Aalborg Ø, Denmark, sbn@es.aau.dk)

When low frequency sound is radiated inside small listening spaces by loudspeakers, large uniformities occur over the sound field. This is due to the multiple reflection and diffraction of sound on the walls and different objects in the room. A developed system named Controlled Acoustically Bass System (CABS) produces uniform sound field at low frequencies. This is performed by utilizing loudspeakers at the front wall and extra loudspeakers at the opposite wall, processed to remove the rear-wall reflection of a rectangular room. Effectiveness of CABS on different room scenarios has been evaluated by using a computer simulation program based on the Finite Difference Time Domain Method (FDTD). CABS has been simulated in a small car cabin and in an irregular room. Non-ideal placement of loudspeakers in CABS have been evaluated. The influence of utilizing CABS with different types of loudspeakers has been evaluated by simulations and measurements.

9:40

2aAAc5. Active Sound Field Equalization. Akira Omoto (Kyushu University, 4-9-1, Shiobaru, Minamiku, 815-8540 Fukuoka, Japan, omoto@design.kyushu-u.ac.jp), Hisaharu Suzuki (Kyushu University, 4-9-1, Shiobaru, Minamiku, 815-8540 Fukuoka, Japan, hisaha@souldsp.jp), Akihiro Kakiuchi (Kyushu University, 4-9-1, Shiobaru, Minamiku, 815-8540 Fukuoka, Japan, kakirikakki @hotmail.com)

Dominant acoustic modes often result in the inevitable non-uniform distribution of the acoustic quantities such as sound pressure, especially in the small sized enclosure. In our study, active control technique is thus introduced to overcome the adverse effects of such modes. Instead of the squared sound pressure which is normally adopted in active noise control, the quantity to be controlled is selected as the weighted values of acoustic intensities or the acoustic impedances in rectangular directions such as x, y and z measured at plural points. The reasonably selected combinations of weights yield the control outputs which can alter the directions of propagating waves and result in reducing the complicated standing waves. Further, results of numerical simulations suggest that the well balanced arrangement of the primary and the secondary sources can 'rectify' the propagating direction and result in the uniform distributions of sound pressure. The strategies used in this study have possibilities of realizing useful controller for active sound field equalization.

10:00

2aAAc6. Different approaches for efficient finite element modelling of absorbers in small rooms. Marc Aretz (RWTH Aachen University, Institute for Technical Acoustics, Neustraße 50, 52066 Aachen, Germany, Marc.Aretz@akustik.rwth-aachen.de)

The FEM is a powerful tool for the numerical simulation of sound fields in enclosures. It accounts for the modal characteristics of the sound field, which are dominant at frequencies below the Schroeder frequency and it is also possible to model the mutual coupling between airborne and structure borne sound fields, when an appropriate structure model is implemented. When applying the FEM to complex room acoustics applications, like e.g. a control room in a recording studio, it is a challenging task to specify realistic boundary conditions. Different kinds of acoustical absorbers like Helmholtz resonators, plate absorbers and complex layered porous absorbers are found in these environments. While it is possible in principle to use complex and exhaustive models for these acoustical absorbers, it is often computationally much more efficient to use acoustic impedances or two-port network FEM elements to represent the fluid structure interactions. In the course of this study we compare different approaches for the modelling of Helmholtz resonators (without and without flow resistance), since it has proven very costly in terms of computation time to explicitly model every single hole in the resonator boxes. The simulation results are compared with results measured in a model room.

10:20-10:40 Break

10:40

2aAAc7. Direct and modal frequency response analysis of sound fields in small rooms by finite element method. Reiji Tomiku (Faculty of Engineering, Oita University, Dannoharu 700, 870-1192 Oita, Japan, tomiku-reiji@cc.oita-u.ac.jp), Toru Otsuru (Faculty of Engineering, Oita University, Dannoharu 700, 870-1192 Oita, Japan, otsuru@cc.oita-u.ac.jp), Noriko Okamoto (Faculty of Engineering, Oita University, Dannoharu 700, 870-1192 Oita, Japan, nokamoto@cc.oita-u.ac.jp), Yuka Kurogi (Faculty of Engineering, Oita University, Dannoharu 700, 870-1192 Oita, Japan, kurogi@cc.oita-u.ac.jp)

In this paper, sound pressures are computed by two techniques using finite element method. One is a technique by solving the system of linear equations directly (direct analysis) and the other is a technique by modal superposition (modal analysis). To confirm the accuracy of the direct analysis, sound pressures obtained by the technique are compared with those obtained by the modal analysis in a room with the volume of 10 m^3. Then, as in the modal analysis, two methods are employed: one is a simplified method based on a real eigenvalue problem assuming that the damping matrix, [C], has orthogonality; and another is the method based on complex eigenvalue problem. Those obtained by the direct analysis are in good agreement with those obtained by the two kinds of modal analyses regardless of absorption conditions, even if the analysis is carried out at the frequency close to an eigen frequency. Next, diffuseness of sound field below 315 Hz in a room, which is used in the measurement of ISO140-3, is investigated by the direct analysis from the viewpoint of mean sound pressure level measurements.

11:00

2aAAc8. The effect of diffusers on low frequency modes. Jamie A. Angus (University of Salford, 12 Lister Way, YO30 6NL York, UK, j.a.s.angus@salford.ac.uk)

Small listening rooms are becoming prevalent, due to the availability of production facilities on personal computers, and the financial pressures faced by the industry. In these rooms, modal behaviour at low frequencies significantly affects the quality of the reverberant decay, due to their non-diffuse nature compared with other frequencies. Diffusion and absorption can help and may be a useful way of improving the low frequency performance of such rooms. This paper discusses the properties of these rooms at low frequencies, in particular the effect of diffusing boundaries on the modal behaviour of such rooms. The paper will first discuss what is meant by a mode and modal decay. It will then go on to examine the effect of diffusing boundaries on the frequency and density of modes. In particular, it will examine the effect of the scale of the diffuser on its efficacy in this task. For ease of visualisation this will be done using a two dimensional model and, for accuracy a finite element, element simulation. The effect of going to three dimensions on the results will also be discussed. Finally, the effect of absorption based diffusing boundaries will be examined.

11:20

2aAAc9. Power output regularization in the active reproduction of sound fields in rooms. Nick Stefanakis (National Technical University of Athens, School of Electrical and Computer Engineering, Heroon Polytechniou 9, 157 73 Athens, Greece, nstefan@mobile.ntua.gr), Sotiris Dalianis (National Technical University of Athens, School of Electrical and Computer Engineering, Heroon Polytechniou 9, 157 73 Athens, Greece, dalias@central.ntua.gr), Tilemachos Karatzas (National Technical University of Athens, School of Electrical and Computer Engineering, Heroon Polytechniou 9, 157 73 Athens, Greece, tilkar6@hotmail.com), George Cambourakis (National Technical University of Athens, School of Electrical and Computer Engineering, Heroon Polytechniou 9, 157 73 Athens, Greece, gcamb@cs.ntua.gr)

In this paper we address the problem of using a multi-channel active control system in order to reproduce a harmonic sound field in a large part of the volume of a reverberant room. The problems associated with the calculation of the inverse system matrix are confronted by introducing a term that is proportional to the sound power-output of the system in the cost function that is obtained by the multiple point method. Simulation results show that this technique results to a better conditioning of the system matrix at low frequencies, comparing to other traditional regularization techniques. Moreover, it is shown that this method can be employed to increase the spatial robustness of the control sensor array inside the listening room.

11:40

2aAAc10. Low-frequency response in active acoustic practice rooms. Ronald Freiheit (Wenger Corporation, 555 Park Drive, Owatonna, MN 55060, USA, ron.freiheit@wengercorp.com)

Integrating active (virtual) acoustics into relatively small practice rooms to create a sense of envelopment is critical to the satisfaction of the musicians using these spaces for learning. The number of speakers and their locations play an important role in minimizing the ability to localize to the sources supporting this sense of envelopment. Equally important is the frequency response required for the speakers used in these applications, to more accurately simulate the acoustics of a performance environment. Excessive high frequencies increase the ability to localize and decrease the sensation of a larger space, since the air in larger spaces naturally absorbs many high frequencies. Another challenge is creating the sense of envelopment for instruments in the bass region (125Hz octave band and below) by providing enough low-frequency energy response. Work will be presented on low-frequency response desired to satisfy a sampling

of solo cello players using active acoustic practice rooms. The optimal sources for these low frequencies - single or multiple - will be discussed. Updated information will also be provided on the sound field coverage in active rooms with extended low-frequency response.

Contributed Papers

12:00

2aAAc11. Active playback of acoustic quadraphonic sound events. Domenico Stanzial (Italian National Research Council, FSSG-CNR Lab Acustica, c/o Fondazione Cini, Isola di San Giorgio Maggiore, 30124 Venezia, Italy, domenico.stanzial@cini.ve.cnr.it), Giorgio Sacchi (Physics Department - University of Ferrara, V. Saragat, 44100 Ferrara, Italy, giorgio.sacchi@student.unife.it), Giuliano Schiffrer (Physics Department - University of Ferrara, V. Saragat, 44100 Ferrara, Italy, schgln@unife.it)

The reproduction in a given confined space - such as a cinema hall or a smaller room - of a sound event previously recorded in a completely different acoustical environment is an interesting and still open acoustical problem. A new method for hi-fi audio playback based on the general solution of the acoustic inverse problem is here pourposed. A feed-forward control based on overdetermination of conditions at active contours - i.e. loudspeakers - in order to obtain an optimal stable solution via least square approach is here proposed. This is easily possible even for complex configurations thanks to acoustic quadraphony, the application of sound intensimetry to audio technology developed in the last years within the IST-2-511316-IP European project denominated IP-RACINE. After a short explanation of the model theory, the experimental application to the simplest case of 1-D confined field is presented and some obtained results are shown.

12:20

2aAAc12. Acoustics in a small control room. Sergio Beristain (Mexican Institute of Acoustics, P.O. Box 12-1022, Narvarte, 03001 Mexico, D.F., Mexico, sberista@hotmail.com)

Acoustics in small rooms is allways a problem, particularly at the low frequency range, because of their size and the fact that there is very limited space to install all the acoustic materials needed in order to obtain the desired acoustic response, but when an electroacoustic system has to be employed as it is the case in a small Control Room for a small recording studio, matters become even worse. First of all, the space is further reduced in order to accomodate the required sound system, and with the presence of hard to control low resonant frecuencies, which can be excited at any moment during any recording session, it can be produced highly different sound presure levels at those frequencies, changing the timbre of sounds. Some measurement results are presented.

TUESDAY MORNING, 1 JULY 2008 ROOM 253, 9:20 A.M. TO 1:00 P.M.

Session 2aAAd

Architectural Acoustics and Noise: Acoustics and Privacy in Healthcare Facilities I: Emerging Policy Around the World

David M. Sykes, Cochair
ANSI S12 Workgroup 44 and the Joint ASA/INCE/NCAC Subcommittee on Healthcare Acoustics & Speech Privacy, 23 Buckingham Street, Cambridge, MA 02138, USA

Kerstin Persson Waye, Cochair
Dept. of Environ. Medicine, The Sahlgrenska Acad. of Gothenburg Univ., Box 414, Gothenburg, 405 30, Sweden

Invited Papers

9:20

2aAAd1. Waves of change: global policies & their impacts on the acoustics profession. David M. Sykes (ANSI S12 Workgroup 44 and the Joint ASA/INCE/NCAC Subcommittee on Healthcare Acoustics & Speech Privacy, 23 Buckingham Street, Cambridge, MA 02138, USA, david.sykes@remington-partners.com)

In a decade, five waves of change swept across the globe stimulating new interest in acoustics. This session examines their effects on the EU, North America and Japan. The Internet caused the first wave-the EU's privacy Directive 9546EC drove nations everywhere to develop privacy laws, many covering ″Speech Privacy.″ A second wave, in 2000 when the ″tech bubble″ burst, produced laws to improve financial accountability and forcing organizations to find ways to shield their leaders. A third wave arose in 2001 with the increase in terrorism, producing a surge in security laws that challenge the right to privacy enshrined in the 1948 Declaration of Human Rights. The fourth wave is demographic: racing to build healthcare facilities for ″boomers,″ this giant industry is now wrestling with a ″noise epidemic″ resulting from decades of neglecting the health effects of noise. The fifth wave came from the ″green″ movement-the concept of ″indoor environmental quality″ has renewed interest in noise as a pollutant. How countries balance such concerns as state security, citizen privacy and human health will be decided by courts but these five waves have catalyzed demand for understanding, insight, expertise, standards, codes, manpower and solutions from the acoustical profession.

9:40

2aAAd2. Developing acoustical policies around in EU countries. Katrin Bergmark (Saint-Gobain Ecophon AB, Box 500, 26061 Hyllinge, Sweden, katrin.bergmark@ecophon.se), Marc R. Janssen (Saint-Gobain Ecophon AB, Box 500, 26061 Hyllinge, Sweden, marc.janssen@ecophon.se)

According to research, noise levels in hospitals worldwide have increased. At the same time, the awareness of the negative effects of noise on patients and healthcare staff has grown. To manage or eliminate noise in hospitals, standards and policies are a great help. An overview of international standards will be presented, showing various parameters and differences in values and angles of approach, e.g based on activity or room type. To further optimize the content, and stimulate the use of the standards, some countries have even defined healthcare specific standards. These can support the planning and the evaluation of the sound environment in those premises. New insights from research and case studies, as well as emerging laws, provide opportunities or even force us to extend the existing content of standards to meet future demands. Suggestions for that include extension of the amount of parameters, such as decrease of sound pressure level (ΔSPL) and parameters addressing privacy (PI, AC). Finally, revision of standards provides opportunities for end users to increase their understanding and interest in acoustic and to supply possibilities to define the desired quality of the sound environment.

10:00

2aAAd3. Effects of healthcare acoustics on medical outcomes. Roger Ulrich (Texas A&M University, Department of Architecture, 3137 TAMU, College Station, TX 77843, USA, rulrich@archmail.tamu.edu)

Although considerable research has examined detrimental effects of noise on patient sleep quality in healthcare buildings, few studies have investigated the extent to which noise may worsen other types of patient clinical outcomes. Studies are also scarce concerning the effects of noise and poor acoustics on healthcare staff. The presentation describes a prospective controlled study conducted with colleagues in Sweden that examined the impact of higher versus lower noise levels, and longer in contrast to shorter reverberation times, on several patient and staff outcomes in a hospital coronary critical care unit (CCU). Acoustics were altered during the study period by changing the ceiling tiles throughout the CCU from sound-reflecting tiles to sound-absorbing tiles of similar appearance. Regarding patients, an improved acoustics environment significantly reduced physiological stress, increased satisfaction with quality of care, improved sleep quality, and lessened incidence of costly re-hospitalizations following discharge. Better acoustics also improved speech intelligibility, and healthcare staff experienced reduced work demands and less pressure and strain. The discussion concludes by outlining research directions, including the need for more outcomes studies to enable development of a strong business case for better healthcare acoustics.

Contributed Papers

10:20

2aAAd4. Influence of intenesive coronary care acoustics on the quality of care and phyiological state of patients. Inger Hagerman (Dept of Cardiology, Karolinska University Hospital, M52, Huddinge, 141 86 Stockholm, Sweden, inger.hagerman@karolinska.se), Gundars Rasmanis (Dept of Cardiology, Karolinska University Hospital, M52, Huddinge, 141 86 Stockholm, Sweden, gundars.rasmanis@karolinska.se), Vanja Blomkvist (Dept of Cardiology, Karolinska University Hospital, M52, Huddinge, 141 86 Stockholm, Sweden, vanja.blomkvist@pubcare.uu.se), Roger Ulrich (Texas A&M University, Department of Architecture, 3137 TAMU, College Station, TX 77843, USA, rulrich@archmail.tamu.edu), Töres Theorell (Dept of Cardiology, Karolinska University Hospital, M52, Huddinge, 141 86 Stockholm, Sweden, tores.theorell@stressforskning.su.se)

Background unstable coronary disease is a stressful situation and environmental influences may increase stress mechanisms important for the cardiovascular status. Aim to evaluate acoustic influence on patients with coronary artery disease and to test if the effects of poor sound absorption on work environment affects quality of care and medical status of patients. Methods 94 patients admitted to the intensive coronary heart unit for chest pain evaluation participated. Patient groups were recruited during bad and good acoustic conditions respectively. Blood pressure, pulse amplitude, heart rate and heart rate variability were monitored. Patients were asked to fill in a questionnaire about the quality of the care and a follow up of re-hospitalisation and mortality was made at 1 and 3 months. Results good and bad acoustics differed significantly with respect to pulse amplitude in acute myocardial infarction and unstable angina pectoris groups, with lower values at night during the good acoustics period. Re-hospitalisations were higher for the bad acoustics group. During the good acoustics period patients considered the staff attitude much better than during the bad acoustics period. Conclusion, bad acoustics environment during acute illness may have important physiological effects of importance for rehabilitation.

10:40

2aAAd5. The future of UK hospital design. Adrian Popplewell (Arup Acoustics, St Giles Hall, Pound Hill, CB3 0AE Cambridge, UK, adrian.popplewell@arup.com)

The importance of acoustic conditions within hospitals is highlighted by recent research indicating that patient recovery times are significantly influenced by the noise levels within wards. Added to this are concerns about speech privacy and confidentiality, patient and staff comfort, communication between patient and doctor, and the effects of vibration on sensitive medical equipment, all of which makes the acoustic design a key parameter in any hospital development. This paper will discuss the practical issues associated with the implementation of the current national guidance on hospital acoustics (Health Technical Memorandum 2045) and the new, currently unpublished replacement, Health Technical Memorandum 08-01. The experiences of the author with several large Private Finance Initiative projects, designed to the current standard, have illustrated the practical difficulties of fully complying with all its requirements and the design solutions which can be adopted. Speech privacy and confidentiality has been a critical issue in developing the new design standard. With Arup Acoustics as key contributors to this document, and acoustic designers of one of the first major projects in the UK to comply with this standard, the author is in a prime position to be able to assess the practical and technical implications of the future guidance.

11:00-11:20 Break

11:20

2aAAd6. Development of 'sound and vibration design guidelines for health care facilities'. Kurt Rockstroh (Steffian Bradley Architects, 100 Summer Street, Boston, MA 02110, USA, kurtr@steffian.com)

"The Guidelines for Design & Construction of Healthcare Facilities" is utilized by the majority of the United States and its federal agencies involved in funding health care projects. As part of the development of the 2010 Edition of "The Guidelines" a joint subcommittee of ASA, INCE, NCAC and the Health Guidelines Revision Committee was formed to develop sound and vibration guidelines. This presentation will explain the collaborative process used to develop, publicize, review and formally vote acceptance of the sound and vibration guidelines. The presentation will also outline the major components of the new guidelines.

11:40

2aAAd7. Acoustics in green buildings: refining the concept of environmentally quality while improving occupant health and productivity synergistically. Sholem Prasow (Teknion Furniture Systems, 1150 Flint Rd., Thornhill, ON L3T 4 M9, Canada, sholem .prasow@teknion.com)

Environmental Quality has always been a keystone of the green building movement. The concept has evolved from "Indoor Air Quality", for example, in the first version of LEED seven years ago to a much broader and enriched concept - a concept that fully embraces Acoustics - today. This paper explores the evolution of Environmental Quality in both general and health care environments, and demonstrates the emerging role of Acoustics as an equal partner with Air Quality and Lighting as determinants of both health and productivity in the built environment. Specific references will be made to LEED rating systems and to emerging acoustical design requirements within LEED.

12:00

2aAAd8. Getting noise and speech privacy issues heard in design of healthcare facilities. Anjali Joseph (The Center for Health Design, 1850 Gateway Blvd., Suite 1083, Concord, CA 94520, USA, ajoseph@healthdesign.org)

Hospitals are extremely noisy, and noise levels in most hospitals far exceed recommended guidelines. The high ambient noise levels, as well as peak noise levels in hospitals, have serious impacts on patient and staff outcomes ranging from sleep loss and elevated blood pressure among patients to emotional exhaustion and burnout among staff. Poorly designed acoustical environments can pose a serious threat to patient confidentiality if private conversations between patients and staff or between staff members can be overheard by unintended listeners. At the same time, a poor acoustical environment impedes effective communication between patients and staff and between staff members by rendering speech and auditory signals less intelligible or detectable. This has serious implications for patient safety. A well-designed acoustical environment is critical in addressing these problems related to noise and communication of information. The purpose of this presentation is to examine how different aspects of sound - noise, speech privacy, speech intelligibility, and music - impact patient and staff outcomes in healthcare settings and the specific environmental design strategies that can be used to improve the acoustical environment of healthcare settings.

12:20

2aAAd9. Noise confounds in functional MRI research and potential solutions. Marc J. Kaufman (McLean Hospital, 115 Mill St., Brain Imaging Center, Belmont, MA 02478, USA, kaufman@mclean.harvard.edu), Blaise D. Frederick (McLean Hospital, 115 Mill St., Brain Imaging Center, Belmont, MA 02478, USA, BBFrederick@mclean.harvard.edu), Eric E. Ungar (Acentech Inc, 33 Moulton St., Cambridge, MA 02138, USA, eungar@ACENTECH.com), Jonathan D. Kemp (Acentech Inc, 33 Moulton St., Cambridge, MA 02138, USA, jdkemp@acentech.com), David M. Sykes (ANSI S12 Workgroup 44 and the Joint ASA/INCE/NCAC Subcommittee on Healthcare Acoustics & Speech Privacy, 23 Buckingham Street, Cambridge, MA 02138, USA, david.sykes@remingtonpartners.com)

High field functional MRI (fMRI) is becoming a neuroscience research technique of choice because it is noninvasive and can reveal brain circuitry regulating sensory, motor, and cognitive functions. Unfortunately, rapid scan fMRI results in high noise levels (100-140 dB) that can alter auditory, visual, and pain system function, and also can induce stress, which itself modulates brain responses to various stimuli. These effects can confound fMRI data interpretation. A number of solutions for this problem have been proposed including modifying MRI scanner hardware to reduce noise output, an expensive proposition limited to willing manufacturers, and modifying fMRI pulse sequences to reduce noise output, which is effective within certain limitations. Another approach is to develop acoustic noise isolating equipment that separates subjects from noise. This passive approach confers maximum flexibility because it is both hardware- and fMRI scan sequence-independent. We present initial data documenting efficacy of first generation acoustic noise isolating equipment for animal fMRI studies. As nearly 25% of patients referred for clinical MRIs refuse scans because they are stressful, acoustic noise isolating equipment scaled for clinical MRI scanner use may decrease stress and increase patient compliance, thereby decreasing morbidity and mortality, and improve MRI center workflow.

12:40

2aAAd10. Experimental study on applicability of sound masking system in medical examination room. Kanako Ueno (Institute of Industrial Science, University of Tokyo, Komaba 4-6-1, Meguro-ku, 153-8505 Tokyo, Japan, ueno@iis.u-tokyo.ac.jp), Hyojin Lee (Institute of Industrial Science, University of Tokyo, Komaba 4-6-1, Meguro-ku, 153-8505 Tokyo, Japan, leehj@iis.u-tokyo.ac.jp), Shinichi Sakamoto (Institute of Industrial Science, University of Tokyo, Komaba 4-6-1, Meguro-ku, 153-8505 Tokyo, Japan, sakamo@iis.u-tokyo.ac.jp), Atsuko Ito (Center for Advanced Sound Technologies, Yamaha, 203 Matsunokijima, 4380192 Iwata, Shizuoka, Japan, atsuko1_ito@gmx.yamaha.com), Mai Fujiwara (Center for Advanced Sound Technologies, Yamaha, 203 Matsunokijima, 4380192 Iwata, Shizuoka, Japan, mai_fujiwara@gmx.yamaha.com), Yasushi Shimizu (Center for Advanced Sound Technologies, Yamaha, 203 Matsunokijima, 4380192 Iwata, Shizuoka, Japan, yasushi_shimizu@gmx.yamaha.com)

Recently, speech privacy to avoid oral information leakage in healthcare facilities has become an important issue. This study investigated effectiveness of sound masking system in regard to masking efficiency, annoyance and its influence on speech conversation for medical examination rooms in an experimental approach. Considering actual application, two adjacent medical examination rooms partitioned by a low sound insulation wall in a typical healthcare facility were selected as an experimental field and sound masking system was temporally installed. In the rooms, acoustic environment was measured and reproduced in an anechoic room with a 3-D sound field simulation system using a 6-ch sound recording/reproduction technique. In the simulated acoustic condition, subjective tests were designed to quantify the masking efficiency and annoyance caused by the masking sound. The annoyance test was conducted in listening condition (with high attention to the sound) and in talking condition (with low attention). As a result, mixed maskers composed by water stream, synthesized speech signals, and steady state noise showed high performance in both aspect of masking efficiency and annoyance.

TUESDAY MORNING, 1 JULY 2008 ROOM 252B, 10:20 A.M. TO 1:00 P.M.

Session 2aAAe

Architectural Acoustics, Musical Acoustics, Physical Acoustics, and Noise: Acoustics of Opera Houses I

Robin Glosemeyer Petrone, Cochair
918 16th St., Apt 3, Santa Monica, CA 90403, USA

Roberto Pompoli, Cochair
Engineering Dept. - Univ. of Ferrara, Via Saragat, 1, Ferrara, 44100, Italy

Invited Papers

10:20

2aAAe1. Generous opera house acoustics for lyric and symphonic performances. Daniel E. Commins (Commins Acoustics Workshop, 15 rue Laurence Savart, 75020 Paris, France, d.commins@comminsacoustics.com)

It is believed that a short acoustic response is needed in an opera house to guarantee good intelligibility of lyrics. Some of the best opera houses have a relatively long reverberation time associated to good clarity. They are more suitable to symphonic concerts than ″damped″ rooms. The recently opened 1350-seat Grand Théâtre de Provence in Aix-en-Provence, France, is an example of an opera house with a long response and good clarity. An adequate orchestra shell and proper permanent acoustical treatment of the stage tower lead to suitable acoustical parameters in the symphony configuration. The measurements show that in an opera house of this capacity, it is possible to create conditions compatible with opera, symphony, chamber music and recitals. The rich opera configuration sound generates mostly positive reactions. Most concert goers consider that this room is now the reference for the region. One must note however that this approach is risky since the values that are considered ideal for opera must be stretched far. It may lead to excessive reverberation and even to unwanted reflections. This target requires genuine investigations, from early concept to final design, using the most advanced prediction and modeling techniques. The design of larger rooms under the same principle requires even more care.

10:40

2aAAe2. Opera House Reverberation Times - Design Considerations. Eckhard Kahle (Kahle Acoustics, 188 avenue Molière, 1050 Brussels, Belgium, kahle@kahle.be), Thomas Wulfrank (Kahle Acoustics, 188 avenue Molière, 1050 Brussels, Belgium, twulfrank@kahle.be), Yann Jurkiewicz (Kahle Acoustics, 188 avenue Molière, 1050 Brussels, Belgium, yjurkiewicz@kahle.be), Brian F. Katz (LIMSI-CNRS, B.P. 133, 91403 Orsay, France, brian.katz@limsi.fr)

The intimacy of many historic European Opera Houses, especially of the traditional Italian style, is highly cherished and many of these halls are considered to be among the best halls acoustically. From an acoustical point of view the generally small dimensions - often combined with a moderate seat count - provide excellent source presence and clarity. On the other hand, the corresponding small volume leads to short reverberation times, and in recent decades higher reverberation times have been preferred and asked for by clients and audiences in many countries. Ideas will be presented on how this apparent dilemma between the preference for small dimensions (for intimacy, source presence and definition) and increased volume (in order to create longer reverberation times) can be addressed.

11:00

2aAAe3. The ″Teatro di San Carlo″ in Naples and its smaller clone ″Teatro Verdi″ in Salerno. Raffaele Dragonetti (DETEC, University of Naples Federico II, Piazzale Tecchio 80, 80125 Naples, Italy, dragonet@unina.it), Carmine Ianniello (DETEC, University of Naples Federico II, Piazzale Tecchio 80, 80125 Naples, Italy, ianniell@unina.it), Francesco Mercogliano (DETEC, University of Naples Federico II, Piazzale Tecchio 80, 80125 Naples, Italy, francesco.mercogliano@unina.it), Rosario A. Romano (DETEC, University of Naples Federico II, Piazzale Tecchio 80, 80125 Naples, Italy, rosroman@unina.it)

The Teatro di San Carlo in Naples (Italy) is a well preserved baroque-type theatre. It was opened in 1737 and flourished up to a destructive fire. Nine months after the disastrous event the San Carlo was rebuilt completely as it was and reopened in 1817. It is still praised for its sound quality for opera performance. The Teatro Verdi was conceived by the Municipality of Salerno (Italy) in 1843. However, the ″querelles″ of local parties delayed much its construction so that it could be opened only in 1872. The architects who designed the Teatro Verdi were influenced very much by the successful Teatro di San Carlo and tried to copy the older and not-far-one in a smaller scale. The Verdi has undergone few minor changes during its life and is used also for opera shows. The main halls of the two are in a scale about 1:5. This paper reports a comparison of their acoustical features in terms of objective room-acoustics parameters accepted for opera-house sound quality. Acoustic measurements were performed with the same instrumentation set. Similarities and differences are discussed.

11:20

2aAAe4. Teatro 'La Fenice', Venice --- The secrets of the acoustical reconstruction of the destroyed theater according to historical and modern requirements. Jürgen Reinhold (Müller BBM GmbH, Robert Koch Strasse 11, D 82152 Planegg/München, Germany, Juergen.Reinhold@MuellerBBM.de)

The reconstruction of this artistically and architecturally very important opera house that was completely destroyed by a fire and which was famous throughout the world not least for its excellent acoustical qualities, was a brilliant challenge also for an acoustician. The historical planning requirements - complete reconstruction ″as it was; where it was″ - the latest findings in room acoustics as well as a huge number of ″modern″ requirements are to be brought in line. The theatre hall as heart of the Teatro ″La Fenice″ was reconstructed in its classical horseshoe shape with five tiers in pure wooden construction. Modern standard specifications had to be met for ventilation and air conditioning, the installation of a modern stage machinery as well as an improved safety and fire prevention concept. Furthermore the space allocation plan had to be extended by the integration of new rehearsal rooms and technical equipment rooms. From the point of view of building and room acoustics this was a very complex task - if in addition the cramped conditions of the Teatro ″La Fenice″ in Venice are taken into consideration. The lecture provides an insight into the building and room acoustical planning, its realization and the achieved acoustical results.

11:40

2aAAe5. The acoustics of the Beijing National Grand Theatre of China. Isabelle Schmich (CSTB, 24, Rue Joseph Fourier, 38400 Saint Martin D'Hères, France, isabelle.schmich@cstb.fr), Paul Chervin (CSTB, 24, Rue Joseph Fourier, 38400 Saint Martin D'Hères, France, paul.chervin@cstb.fr), Zhu Xiangdong (School of Architecture, Tsinghua University, Room 104, Centre Main Building, 100084 Beijing, China, zxd@abcd.edu.cn), Yan Xiang (School of Architecture, Tsinghua University, Room 104, Centre Main Building, 100084 Beijing, China, yx@abcd.edu.cn), L Guo-Qi (The National Grand Theatre, 2, Chang an sast street, 100031 Beijing, China, liguoqi@fujisound.com)

The National Grand Theatre of China opened in December 2007 in Beijing. It is a complex of three performance halls: the Opera (2400 seats), the Concert Hall (2000 seats) and the Theatre (1100 seats). These three halls are covered by a super-ellipsoidal shell made out of titanium and glass. The history of the acoustic design evolution from the winning of the competition in 1999 by the Architect Paul Andreu to the final construction will be retraced including the preliminary design, the acoustic simulations and auralisations. The acoustic design of each hall will be presented with its specifications and detailed particularities. Final acoustic measurements have been done in the three empty halls and in the occupied concert hall. The results and acoustic criteria will be discussed. In addition, a subjective evaluation has been done with psychoacoustic questionnaires and the results will be presented and analysed.

Contributed Papers

12:00

2aAAe6. Acoustical design and scale model test for the opera house of Zhongshan city. Shuo Xian Wu (State Key Laboratory of Subtropical Building Science, South China University of Technology, 381 Wushan Street, 510640 Guangzhou, China, arshxwu@scut.edu.cn), Yue Zhe Zhao (State Key Laboratory of Subtropical Building Science, South China University of Technology, 381 Wushan Street, 510640 Guangzhou, China, arzhyzh@scut.edu.cn)

The opera house of Zhongshan City in Guangdong province of China is the performing art center of the city. It was completed in Oct. 2005. It has 1400 seats including 833 stall seats and another 484 seats on two levels of balcony. The volume of the auditorium is 12000m3 and the V/N is 9.1m3 per seat. The main purpose of the building is for the performances of opera and ballet. Some music and conference events are also to be held there. Therefore, the occupied reverberation time at medium frequencies is set to be 1.6s and its background noise level has to meet NR-20. During the acoustical design, a 3D computer simulation model established with ODEON software and a 1:20 scale model was made to analyze the sound fields of the auditorium with and without a music shell on the stage. After the completion of the building, an acoustics test was taken. Several performances show that its acoustics reaches a quite high level and has won high praises.

12:20

2aAAe7. Subjective Assessments of Acoustical Environments for Un-assisted Traditional Peking Opera Performances. Wei-Hwa Chiang (National Taiwan University of Science and Technology, 43, sec. 4 Keelung Rd, Taiwan, 106 Taipei, Taiwan, edchiang1224@hotmail.com), Wei Lin (National Taiwan University of Science and Technology, 43, sec. 4 Keelung Rd, Taiwan, 106 Taipei, Taiwan, D9313001@mail.ntust.edu.tw), Ya-Jhen Yu (National Taiwan University of Science and Technology, 43, sec. 4 Keelung Rd, Taiwan, 106 Taipei, Taiwan, ianmelody@hotmail.com)

The prevalence of Western performing arts has impacted on how traditional Peking opera is performed. It is, however, necessary to preserve and pass on traditional Peking opera for it integrates the essence of Chinese

opera. The research has been conducted regarding subjective assessment of the acoustical environment for traditional Peking opera. Recording of dry sound sources was also performed. Sound strength and listening direction were found to be the principle factors that determined evaluation for overall impression while reverberation time is less important. The 6000 m3 can be used as the upper limit for room volume when a thrust stage is used.

12:40

2aAAe8. Considerations about the acoustical properties of Teatro Nuovo in Spoleto after the restauration works. Alessandro Cocchi (University, DIENCA Dept. Facoltà di Ingegneria, Viale Risorgimento 2, 40136 Bologna, Italy, alessandro.cocchi@mail.ing.unibo.it), Marco Cesare Consumi (University, DIENCA Dept. Facoltà di Ingegneria, Viale Risorgimento 2, 40136 Bologna, Italy, marcocesare.consumi@mail.ing.unibo.it),Ryota Shimokura(AIST, 1-8-31 Midorigaoka, Ikeda, 563-8577 Osaka, Japan, ryota.shimokura@aist.go.jp)

In 2003 the municipality of Spoleto decided to stop the performances within the Teatro Nuovo, among the others also of ″Two World Annual Festival″, as it was necessary to modify some material, repair the floor, refurbish pictures and so on: the first author was charged with all the acoustical aspects and he decided to perform extensive acoustical measurements. During this measurement campaign, a flutter echo was detected in the stalls, so it became necessary to detect the origin of this problem, then to find some modification able to remove this problem: we have already presented our studies on this subject, from which a new design of the orchestra pit was derived. For fire safety reasons it was necessary also to remove some elastic panels from the boxes, and an acoustically equivalent technical solution was adopted. Even the stalls floor was removed and a new solution was adopted incorporating the heating plant and an acoustical solution. In this paper we will present the results of measurements performed after the opening of the theatre (a typical Italia Opera House of the end of the Eighteen Century) and some consideration about the acoustical results so obtained.

TUESDAY MORNING, 1 JULY 2008 ROOM 342B, 8:00 A.M. TO 1:20 P.M.

Session 2aAB

Animal Bioacoustics and ECUA: Animal Bioacoustic Censusing I

Marie A. Roch, Cochair
San Diego State University, Dept. of Computer Science, San Diego, CA 92182-7720, USA

Cédric Gervaise, Cochair
E3I2 - EA3876, 2 rue François Verny, Brest Cedex, 29806, France

Invited Papers

8:00

2aAB1. Passive acoustic detection of grouper sound production. David Mann (University of South Florida, College of Marine Science, 140 7th Ave. S., St. Petersburg, FL 33701, USA, dmann@marine.usf.edu), James Locascio (University of South Florida, College of Marine Science, 140 7th Ave. S., St. Petersburg, FL 33701, USA, locascio@marine.usf.edu), Michelle Schärer (University of Puerto Rico-Mayaguez, Department of Marine Sciences, Isla Magueyes, La Parguera, 00667 Lajas, Puerto Rico, m_scharer@hotmail.com), Chris Koenig (Florida State University, Coastal and Marine Laboratory, St. Teresa Beach, FL 32358, USA, koenig@bio.fsu.edu), Michael Nemeth (University of Puerto Rico-Mayaguez, Department of Marine Sciences, Isla Magueyes, La Parguera, 00667 Lajas, Puerto Rico, michaelnemeth@hotmail.com), Misty Nelson (University of South Florida, College of Marine Science, 140 7th Ave. S., St. Petersburg, FL 33701, USA, mnelson@marine.usf.edu), Felicia Coleman (Florida State University, Coastal and Marine Laboratory, St. Teresa Beach, FL 32358, USA, coleman@bio.fsu.edu), Richard Appeldoorn (University of Puerto Rico-Mayaguez, Department of Marine Sciences, Isla Magueyes, La Parguera, 00667 Lajas, Puerto Rico, rappeldo@uprm.edu)

Passive acoustic recordings were used to study the behavior of red hind (*Epinephelus guttatus*) at spawning aggregation sites off of Puerto Rico and Mona Island, and goliath grouper (*Epinephelus itajara*) and red grouper (*Epinephelus morio*) on the West Florida Shelf. The sounds produced by each species were unique, low-frequency pulsed sounds and associated with reproductive-related behavior. Male red hind produced sounds composed of a series of pulses that graded into a tonal-like sound, mostly during territorial patrolling. Long-term acoustic recorder (LARS) data from the west coast of Puerto Rico and Mona Island showed similar diel periodicities of sound production of red hind, but had different monthly peaks in sound production. Goliath grouper sounds consisted of low-frequency individual pulses (50-100 Hz) that are consistent with sounds produced by a large fish, and showed lunar periodicity in sound production. Peaks in sound production occurred nightly after midnight, but sounds were also produced throughout the day. Sound production and spawning by red grouper was recorded using a remotely operated vehicle (ROV) at depths of 80-100 m. Passive acoustic techniques can provide synoptic, long-term time series of sound production associated with reproductive activities of soniferous species at widely spaced sites.

8:20

2aAB2. The value of acoustic technologies for monitoring bird migration. Andrew Farnsworth (Cornell Laboratory of Ornithology, 420 E 54th St, 38J, New York, NY 10021, USA, af27@cornell.edu)

Many North American species of birds give distinctive flight calls during nocturnal migration. Monitoring these calls can be a powerful method for studying the timing and magnitude of migration, as well as for confirming the presence of individual species and

potentially for quantifying passage rates. Recent technological advances in acoustic monitoring, such as increased processor speeds of computers, automated detection software, increased data storage capacities, and a comprehensive identification guide, permit recording of the vocalizations of passing migrants over entire nights for entire migration seasons, thus yielding data on species composition, migration timing and routing, and the magnitude of migration traffic. I will discuss several recent studies that have used acoustic technologies to monitor nocturnal migration, presenting data on species composition and relative abundance. Additionally, I will present some recent approaches to addressing quantification of passing migrants by their calls. I will also discuss wider applications of this technology beyond the realm of North American migrant species.

Contributed Papers

8:40

2aAB3. Automatic detection of short time periodic bird calls in realistic monitoring scenarios. Daniel Wolff (Department of Computer Science III, University of Bonn, Roemerstr. 164, 53117 Bonn, Germany, wolffd@bonn.edu), Klaus H. Tauchert (Humboldt-University Berlin, Institute of Biology, Invalidenstr. 43, 10115 Berlin, Germany, klaus.tauchert@gmx.de), Karl H. Frommolt (Humboldt-University Berlin, Museum für Naturkunde, Tierstimmenarchiv, Invalidenstr. 43, 10115 Berlin, Germany, karl-heinz.frommolt@rz.hu-berlin.de), Rolf Bardeli (Department of Computer Science III, University of Bonn, Roemerstr. 164, 53117 Bonn, Germany, bardeli@iai.uni-bonn.de), Frank Kurth (FGAN-FKIE, Abteilung KOM, Neuenahrer Str. 20, 53343 Wachtberg-Werthhoven, Germany, kurth@fgan.de), Michael Clausen (Department of Computer Science III, University of Bonn, Roemerstr. 164, 53117 Bonn, Germany, clausen@iai.uni-bonn.de)

In this contribution, we propose a method for the automatic detection and localisation of bird calls featuring simple as well as highly repetitive structures. Reporting from a research project focused at computer aided acoustical monitoring, a detector for Locustella luscinoides (Savi's Warbler) is presented, performing reliably even on highly distorted recordings. In cooperation with the Animal Sound Archive at the Humboldt University, Berlin, this detector was incorporated into a multimodal censusing method combining GPS-data and automatically annotated audio recordings to perform line mapping. An evaluation of the proposed techniques for unsupervised monitoring purposes was also performed on 20 hours of multichannel recordings from Lake Parstein, a cane brake area in Germany. Finally, exemplifying the discriminative potential of the underlying features, we will point out the application to other members of the Warbler family as well as the detection of cricket and toad sounds.

9:00

2aAB4. Animal censusing using seismic cues: techniques used for African elephants. Jason D. Wood (Beam Reach Marine Science and Sustainability School, 7044 17th Ave NE, Seattle, WA 98115, USA, jason@beamreach.org), Caitlin E. O'Connell-Rodwell (Stanford University, Department of Otolaryngology, Head and Neck Surgery, Stanford, 94305, USA, ceoconnell@stanford.edu), Simon L. Klemperer (Stanford University, Department of Geophysics, Stanford, 94305, USA, sklemp@stanford.edu)

Counting populations of animals has proven to be difficult and inexact for species that are difficult to detect visually. A growing number of researchers have successfully turned to detecting animals by their acoustic signals in order to count their numbers. To predict the number of animals present a regression line is generated from the relationship between such measures as calling rate and the number of animals documented visually during that recording by the researcher. Depending on the communication system of the species in question, a significant amount of the variation in calling rate can be attributed to other factors such as group behavior rather than the number of individuals present. For species where these other sources of variation are high it may be more appropriate to use acoustic cues to detect and count the number of animals. This paper will present techniques used to detect and estimate elephant numbers from geophone recordings of their footfalls; cues that are a byproduct of their locomotion and are thus not as dependent on group behavior.

Invited Papers

9:20

2aAB5. Species identification and measurement of activity in odontocete species of Palmyra Atoll by acoustic monitoring. Simone Baumann (Scripps Institution of Oceanography- UCSD, 9500 Gilman Dr. #0205, La Jolla, CA 92093, USA, sbaumann@ucsd.edu), John A. Hildebrand (Scripps Institution of Oceanography- UCSD, 9500 Gilman Dr. #0205, La Jolla, CA 92093, USA, jhildebrand@ucsd.edu), Sean M. Wiggins (Scripps Institution of Oceanography- UCSD, 9500 Gilman Dr. #0205, La Jolla, CA 92093, USA, swiggins@ucsd.edu), Hans-Ulrich Schnitzler (Eberhard-Karls-Universität Tübingen, Zool. Institut, Abt. Tierphysiologie, Auf der Morgenstelle 28, 72076 Tübingen, Germany, hans-ulrich.schnitzler@uni-tuebingen.de)

Acoustic monitoring has been used to study odontocete presence at Palmyra Atoll, a remote island in the Northern Line Islands chain. Long-term recordings of high-frequency, broadband acoustic data have become possible with recent technological advances. A High-frequency Autonomous Recording Package (HARP) has been developed which samples at 200 kHz with a duty cycle of 1/4 for up to seven months. This instrument has recorded since October 2006 at Palmyra Atoll. Visual and acoustic surveys were conducted around Palmyra Atoll using a four-element towed hydrophone array sampling real-time at 200 kHz to obtain species-specific acoustic data. These data are used as reference for automatic detection algorithms applied on the long-term recordings. To date, acoustically and visually detected odontocetes include bottlenose dolphins (Tursiops truncatus), spinner dolphins (Stenella longirostris), melon-headed whales (Peponocephala electra) and beaked whales of the genus Mesoplodon. The long-term HARP data reveal acoustic activity primarily at night time and predominantely odontocete clicks. Both the beaked as well as the melon-headed whales are present year round and show a distinct daily acoustic activity cycle.

9:40

2aAB6. Distribution patterns of delphinids in the California Current Ecosystem observed through acoustic monitoring of species-specific echolocation clicks. Erin M. Oleson (Scripps Institution of Oceanography- UCSD, 9500 Gilman Dr. #0205, La Jolla, CA 92093, USA, eoleson@ucsd.edu), Melissa S. Soldevilla (Scripps Institution of Oceanography- UCSD, 9500 Gilman Dr. #0205, La Jolla, CA 92093, USA, msoldevilla@ucsd.edu), John Calambokidis (Cascadia Research Collective, 218 1/2 W. 4th Ave., Olympia, WA 98501, USA, calambokidis@cascadiaresearch.org), Curtis Collins (Naval Postgraduate School, 833 Dyer Road, Rm 328,

Monterey, CA 93943, USA, Collins@nps.edu),Sean M. Wiggins (Scripps Institution of Oceanography- UCSD, 9500 Gilman Dr. #0205, La Jolla, CA 92093, USA, swiggins@ucsd.edu), John A. Hildebrand (Scripps Institution of Oceanography- UCSD, 9500 Gilman Dr. #0205, La Jolla, CA 92093, USA, jhildebrand@ucsd.edu)

Visual surveys along the U.S. west coast conducted in the 1980s-90s suggested that Pacific white-sided and Risso's dolphins likely undergo annual movement between California during winter-spring, and Oregon-Washington in the summer-fall. Using high-frequency autonomous acoustic recordings within the southern California, central California, and Washington regions of the California Current System, we evaluated the seasonal occurrence of these dolphin species through detection of their echolocation clicks. The clicks of Pacific white-sided and Risso's dolphins are characterized by a unique combination of local frequency peaks for each species, providing a statistically robust means of identification from autonomous acoustic records. Based on these unique spectral characters, we extracted periods of clicking by each species from concurrently sampled coastal and shelf/slope locations off southern California and Washington, and a single offshore site off central California. The number of hours that each species was heard per day was quantified and compared among seasons and regions. Both species were heard year-round at shelf/slope sites off southern California and Washington; however their patterns of occurrence at coastal sites differed somewhat. Our results indicate year-round presence of Pacific white-sided and Risso's dolphins in some regions, suggesting that the seasonal distribution of these species may have recently changed.

10:00-10:20 Break

Contributed Papers

10:20

2aAB7. Radial distance sampling with passive acoustics: The prospect of estimating absolute densities of cetaceans from static acoustic datalogger data. Jakob Tougaard (National Environmental Research Institute, University of Aarhus, Frederiksborgvej399, DK-4000 Roskilde, Denmark, jat@dmu.dk)

Recording animal vocalisations with a static acoustic datalogger can be viewed as a form of point transect sampling and results analysed within the framework of distance sampling theory. The key element is the radial detection function, which specifies the probability of detecting a vocalisation as a function of distance from the datalogger. This function can be modelled theoretically or preferably determined experimentally and from it the efficient radial detection distance can be determined. Radial detection functions were determined for two different harbour porpoise dataloggers (T-PODs, version 1 and version 3) by means of concurrent visual tracking of porpoises. 52% and 82% of the porpoises within 100 m from the T-PODs were detected by the V1 and V3 T-POD, respectively and 11% and 30%, respectively were detected between 100 and 200 m from the T-PODs. Effective detection radius (EDR) was 97 m and 150 m for the two T-PODs, respectively. Porpoises echolocate almost continuously. If it is assumed that silent periods rarely exceeds 1 minute in duration an average detection rate on the version 3 T-POD for the study site of 2.7 detection positive minutes per hour can be converted into a density estimate of 0.69 porpoises/km^2.

10:40

2aAB8. Computer-aided detection of non-stereotyped bowhead whale calls in the presence of seismic airgun signals. Aaron Thode (Marine Physical Laboratory, Scripps Institution of Oceanography, 9500 Gilman Dr, MC 0238, La Jolla, CA 92093-0238, USA, athode@ucsd.edu), Delphine Mathias (Marine Physical Laboratory, Scripps Institution of Oceanography, 9500 Gilman Dr, MC 0238, La Jolla, CA 92093-0238, USA, delphine.mathias@gmail.com), Miles McLennan (Greeneridge Sciences, Inc., 4512 Via Huerto, Santa Barbara, CA 93110, USA, billm @greeneridge.com), Charles R. Greene (Greeneridge Sciences, Inc., 4512 Via Huerto, Santa Barbara, CA 93110, USA, cgreene@greeneridge.com)

In 2007 thirty-five autonomous recording packages were deployed over a 150 mile swath in the Beaufort Sea to monitor the annual migration of the bowhead whale (Balaena mysticetus) population during seismic exploration activities. Over 1350 days worth of acoustic data were gathered, generating a need for computer-aided assistance in detecting calls. Bowhead whales produce over ten types of frequency modulated calls, and the frequency range, duration, and fine structure of individual calls vary considerably even within each call type, creating difficulties when using simple matched-filtering or spectrogram correlation. Numerous airgun signals also display frequency-modulated structure, complicating the challenge. In this presentation a three-stage process is presented for detecting arbitrary bowhead whale calls in the presence of seismic airgun signals, implemented in JAVA and MATLAB, with components extracted from the industry-supported PAMGUARD software package. The first stage runs several ″energy-based″ detectors simultaneously across multiple frequency bands to capture events, and a second stage analyzes the timing of these detections to remove regular periodic sequences, such as those expected from airguns. The final stage attempts to trace contours on spectrograms. Both genetic algorithms and direct optimization are used to optimize the program's 21 input parameters. [Work supported by Shell Exploration and Production Company].

Invited Papers

11:00

2aAB9. Comparison of feature extraction methods for the identification of odontocete species based upon echolocation clicks. Marie A. Roch (San Diego State University, 5500 Campanile Dr, Dept of Computer Science, San Diego, CA 92182-7720, USA, marie.roch@sdsu.edu), Holger Klinck (Alfred Wegener Institute, P.O. Box 120161, 27515 Bremerhaven, Germany, holger.klinck@awi.de), David K. Mellinger (Oregon State Univ. and NOAA, 2030 SE Marine Science Dr., Newport, OR 97365, USA, David.Mellinger@oregonstate.edu), Melissa S. Soldevilla (Scripps Institution of Oceanography- UCSD, 9500 Gilman Dr. #0205, La Jolla, CA 92093, USA, msoldevilla@ucsd.edu), John A. Hildebrand (Scripps Institution of Oceanography- UCSD, 9500 Gilman Dr. #0205, La Jolla, CA 92093, USA, jhildebrand@ucsd.edu)

Recent work by several groups has shown that odontocete echolocation clicks contain information that can be used to detect or identify specific species. In this study, we compare the relative performance of cepstral and wavelet features on various Pacific Ocean species of odontocetes. Comparison of features within different systems is often complicated by the large number of variables unrelated to feature extraction that change between systems. By experimenting within a proven state of the art classification framework, it is possible to make meaningful comparisons of feature extraction performance with respect to common machine learning algorithms such as neural networks, support vector machines, and Gaussian mixture models.

11:20

2aAB10. Long-term Passive Acoustic Monitoring of Delphinids in the Southern California Bight. Melissa S. Soldevilla (Scripps Institution of Oceanography- UCSD, 9500 Gilman Dr. #0205, La Jolla, CA 92093, USA, msoldevilla@ucsd.edu), John A. Hildebrand (Scripps Institution of Oceanography- UCSD, 9500 Gilman Dr. #0205, La Jolla, CA 92093, USA, jhildebrand@ucsd.edu), Sean M. Wiggins (Scripps Institution of Oceanography- UCSD, 9500 Gilman Dr. #0205, La Jolla, CA 92093, USA, swiggins@ucsd.edu), Marie A. Roch (San Diego State University, 5500 Campanile Dr, Dept of Computer Science, San Diego, CA 92182-7720, USA, marie.roch@sdsu.edu)

Spectral characteristics of clicks are described for five species of delphinids in the Southern California Bight (long-beaked common dolphins, Delphinus capensis, short-beaked common dolphins, Delphinus delphis, Risso's dolphins, Grampus griseus, Pacific white-sided dolphins, Lagenorhynchus obliquidens, and bottlenose dolphins, Tursiops truncatus). Recent technological advancements allow long-term, broadband (100 kHz bandwidth), passive acoustic monitoring from autonomous sea-floor mounted instruments. The ability to record higher frequencies allows study of a broader range of odontocete vocalizations including echolocation clicks. To determine whether delphinid species could be identified by their clicks, concurrent ship-based visual and acoustic surveys were conducted. We find that clicks from Pacific white-sided and Risso's dolphins contain spectral peaks and notches that are unique and consistent for each species. These spectral patterns are also apparent in long-term autonomous acoustic recordings throughout the Southern California Bight. Utilizing this spectral classification method, we examine of diel, seasonal, and habitat use patterns of acoustically active Risso's and Pacific white-sided dolphins. The ability to monitor animals through the night provides insight into distinct diel patterns of acoustic activity for both species while the high temporal resolution acoustic data can be used to relate oceanographic time series to dolphin activity.

Contributed Papers

11:40

2aAB11. Sound production and spawning by black brum (Pogonias cromis) in southwest Florida. James Locascio (University of South Florida, College of Marine Science, 140 7th Ave. S., St. Petersburg, FL 33701, USA, locascio@marine.usf.edu), Ernst Peebles (University of South Florida, College of Marine Science, 140 7th Ave. S., St. Petersburg, FL 33701, USA, epeebles@marine.usf.edu), David Mann (University of South Florida, College of Marine Science, 140 7th Ave. S., St. Petersburg, FL 33701, USA, dmann@marine.usf.edu)

We used the Long Term Acoustic Recording System (LARS) to investigate patterns of sound production in spawning aggregations of black drum (Pogonias cromis) during 2004 - 2006 in southwest Florida. Our purposes were to document trends in black drum sound production on daily and seasonal time scales and to investigate the temporal and quantitative relationships between egg production and sound production. Sound production was strongly diel, beginning near dusk and lasting for several hours. Sound production occurred from October through April and peaked in February-March, consistent with prior descriptions of the spawning season for this species based on the gonado-somatic index (GSI). To investigate the relationship between sound production and egg production, surface plankton tows were conducted hourly between 1600 - 0400 on two consecutive nights while continuous underwater acoustic recordings were made. This was done five times between January and April, 2006. Neither the timing nor quantity of sound production was correlated with egg production on a nightly basis. These results indicate that patterns in sound production are not useful for predicting patterns in egg production by black drum on a daily scale but do provide accurate characterization of spawning behavior on a seasonal basis.

12:00

2aAB12. Nearest-neighbor techniques for automated monitoring of nocturnal flight calls. Harold Figueroa (Cornell University, Bioacoustics Research Program, 159 Sapsucker Woods Road, Ithaca, NY 14850, USA, harold.figueroa@gmail.com), Andrew Farnsworth (Cornell Laboratory of Ornithology, 420 E 54th St, 38J, New York, NY 10021, USA, af27@cornell.edu)

Flight-calls are short vocalizations used primarily during nocturnal flight . Their observation provides a means for studying the timing, location, and composition of nocturnal migrations. As part of a three-year study the Cornell Lab of Ornithology is using autonomous recorders to sample flight-calls of nocturnal migrants in the Northeastern US. The resulting tens of thousands of hours of recording, make software-assisted detection and classification essential. Automatic processing and human evaluation have yielded a considerable collection of flight-calls, 5-1000 examples for ~100 species. The many-class classification problem, along with the availability of many examples from most of the classes, and established (condensation and editing) and recent (metric-trees) techniques used in prototype-based classification nearest-neighbor techniques, have led us to develop nearest-neighbor based techniques and software to assist in the analysis of this data. We will present classification results on two examples, a set of thrushes (genera Catharus and Hylocichla, family Turdidae) consisting of six species and wood-warblers (family Parulidae) consisting of 48 species. The thrush flight-calls are visually and aurally distinctive, usually 100-400 ms in duration and occupy and the 2-5 kHz band. Wood-warbler flight-calls are typically between 20-100 ms in duration and occupy the 5-10 kHz, and are difficult for many experienced observers to distinguish.

Invited Paper

12:20

2aAB13. Sperm whale monitoring with a deep acoustic platform: Results from NEMO ONDE experiment and way ahead. Gianni Pavan (CIBRA - Università di Pavia, Via Taramelli 24, 27100 Pavia, Italy, gianni.pavan@unipv.it)

Within the INFN NEMO Project on the underwater detection of high energy neutrino, a deep hydrophone station, named ONDE (Ocean Noise Detection Experiment), has been deployed on the seafloor 21 km offshore Catania (Sicily, Italy), at 2000 m depth. Acoustic data recorded in 2005 and 2006 provided long term information on the underwater noise and a huge amount of sperm whales' detections that indicate a presence of the species higher than previously believed. Only few sightings are available for the previous years and scarce literature is available for the area. With ONDE, in year 2005 sperm whales were detected in 117 of the 231 recorded days and in 31 of the 83 days in year 2006. Clicks were the most common vocalizations recorded. Chirrups and codas (dominated by the 3+1 pattern) were present frequently, but creaks, possibly indicating feeding actions, were seldom heard. The whales were often detected, solitary or in groups, for time segments of only a few hours. This, with the low rate of creaks, may mean they were just in transit. Based on these results, a new project named (Listening Into the Deep Ocean) has been set with INGV to create a Mediterranean wide network by upgrading existing underwater seismic detectors with broadband acoustic sensors.

Contributed Paper

12:40

2aAB14. Detection and classification of call types in the vocalizations of north-east pacific blue whales. Jack McLaughlin (University of Washington, 1013 NE 40th St, Seattle, WA 98105-6698, USA, jackm@apl.washington.edu), Nicolas Josso (GIPSA-lab, dep. DIS, 961, rue de la Houille Blanche, 38402 St Martin d'Hères, France, nicolas.josso@gispa-lab.inpg.fr), Cornel Ioana (GIPSA-lab, dep. DIS, 961, rue de la Houille Blanche, 38402 St Martin d'Hères, France, cornel.ioana @gipsa-lab.inpg.fr)

Characterization of marine mammal vocalizations is an essential part of any program of marine mammal monitoring as well as being of great help for furthering understanding of subjects such as underwater communication, sonar, etc. The vocalizations of the North-East Pacific (NEPAC) blue whales are known to be made of at least three different call types: the A call, the B call and the C call. This study aims at the development of a wholly automatic process of detection and classification for the two most common call types of the NEPAC population which are the A call and the B call. We created one template for the A call and one for the B call in order to extract features with matched filtering operations. We show that a simple Gaussian Mixture Model classifier can be used to accurately track and identify the call types in 24-hour long records. The proposed methodology is applied to real data sets recorded by seismic sensors.

Invited Paper

1:00

2aAB15. Experience with VoxNet: a rapidly-deployable acoustic monitoring system for bio-acoustic studies. Lewis Girod (MIT/CSAIL, 32G-918, 32 Vassar St, Cambridge, MA 02139, USA, girod@nms.csail.mit.edu), Michael Allen (Cogent Computing ARC, Coventry University, Coventry, UK, allenm@lecs.cs.ucla.edu), Travis Collier (UCLA Dept. of Biology, 3563 Boelter Hall, Los Angeles, CA 90095, USA, travc@taylor0.biology.ucla.edu), Daniel T. Blumstein (UCLA Dept. of Biology, 3563 Boelter Hall, Los Angeles, CA 90095, USA, marmots@ucla.edu), Deborah Estrin (UCLA Dept. of Biology, 3563 Boelter Hall, Los Angeles, CA 90095, USA, destrin@cs.ucla.edu), Charles Taylor (UCLA Dept. of Biology, 3563 Boelter Hall, Los Angeles, CA 90095, USA, taylor @biology.ucla.edu)

Terrestrial bioacoustic census is a difficult problem because of propagation characteristics, obstructions, the diversity of bioacoustic sources, and the impact of noise. To address this problem we have developed VoxNet, a complete hardware and software platform for distributed acoustic monitoring applications. Each VoxNet node is a portable, self-contained processor with a small four-channel acoustic array. Using a distributed set of VoxNet nodes, a forested habitat can be monitored and the behavior of animals can be recorded and analyzed acoustically. In this work we present our experiences applying VoxNet to bioacoustic census. This work is based on data collected using the system during a deployment in Chiapas, Mexico at the Chajul Biological Field Station. The Chajul station is located in a region of dense rain forest and is home to Mexico's most diverse ecosystem. Using VoxNet in this harsh environment, we obtained census estimates based on observation of bird calls.

Session 2aAOa

Acoustical Oceanography and ECUA: Marine Ecosystem Acoustics II

Kenneth G. Foote, Cochair
Woods Hole Oceanographic Institution, Woods Hole, MA 02543, USA

Olav R. Godoe, Cochair
Institute of Marine Research, PO Box 1870, Bergen, 5817, Norway

Invited Papers

8:00

2aAOa1. Spawning Behaviour and Spatial Distribution of Atlantic Herring on Georges Bank Revealed by Ocean Acoustics Waveguide Remote Sensing. Purnima Ratilal (Northeastern University, 302 Stearns Center, Rm 311, 360 Huntington Ave, Boston, MA 02115, USA, purnima@ece.neu.edu), Zheng Gong (Northeastern University, 302 Stearns Center, Rm 311, 360 Huntington Ave, Boston, MA 02115, USA, zgong@ece.neu.edu), Daniel Cocuzzo (Northeastern University, 302 Stearns Center, Rm 311, 360 Huntington Ave, Boston, MA 02115, USA, dcocuzzo@ece.neu.edu), Mark Andrews (Northeastern University, 302 Stearns Center, Rm 311, 360 Huntington Ave, Boston, MA 02115, USA, Andrews.mar@neu.edu), Srinivasan Jagannathan (Massachusetts Institute of Technology, Room 5-435, 77 Massachusetts Avenue, Cambridge, MA 02139, USA, jsrini@mit.edu), Ioannis Bertsatos (Massachusetts Institute of Technology, Room 5-435, 77 Massachusetts Avenue, Cambridge, MA 02139, USA, ibertsat@mit.edu), Tianrun Chen (Massachusetts Institute of Technology, Room 5-212, 77 Massachusetts Avenue, Cambridge, MA 02139, USA, trchen@mit.edu), Hector Pena (Institute of Marine Research, PO Box 1870, 5817 Bergen, Norway, hector.pena@imr.no), Ruben Patel (Institute of Marine Research, PO Box 1870, 5817 Bergen, Norway, ruben.patel@imr.no), Olav R. Godo (Institute of Marine Research, PO Box 1870, 5817 Bergen, Norway, olav.rune.godoe@imr.no), J. Michael Jech (NOAA/Northeast Fisheries Science Center, 166 Water Street, Woods Hole, MA 02543, USA, michael.jech@noaa.gov), Thomas C. Weber (University of New Hampshire, Ctr. for Coastal and Ocean Mapping, 24 Colovos Road, Durham, NH 03824, USA, weber@ccom.unh.edu), Nicholas Makris (Massachusetts Institute of Technology, Room 5-212, 77 Massachusetts Avenue, Cambridge, MA 02139, USA, makris@mit.edu)

An ocean acoustics waveguide remote sensing (OAWRS) system was deployed in the Gulf of Maine, near Georges Bank to image Atlantic herring and other fish populations from Sep-Oct 2006. OAWRS provides spatially unaliased imaging of herring over wide areas, spannning over 100 km diameter. Migration and spawning behaviour of Atlantic herring was observed using OAWRS over several diurnal periods, including massive movements on and off the bank to spawn. Measurements made simultaneously with a conventional fish-finding echosounder (CFFS) and a multibeam sonar provide the depth distribution and local 3D morphology respectively of the herring schools in the water column. Concurrent trawl surveys provide identification of the fish species. Measurements mades by OAWRS and CFFS systems are highly correlated. Examples will be provided of the co-registration between the two systems over a one week period. Calibration of the OAWRS system using CFFS estimates of fish population densities along with a full-field scattering model that takes into account both coherent and incoherent scattering from a fish group is discussed. The fish swimbladder is modelled as a spheroidal bubble. Resonance scattering behaviour of herring is observed in the OAWRS system with significant changes in scattering amplitude over the 300 Hz to 1.5 kHz frequency range of the OAWRS system.

8:20

2aAOa2. Passive acoustic mapping of marine biological choruses. Gerald D'Spain (Marine Physical Lab, Scripps Institution of Oceanography, 291 Rosecrans St., San Diego, CA 92106, USA, gld@mpl.ucsd.edu), Heidi Batchelor (Marine Physical Lab, Scripps Institution of Oceanography, 291 Rosecrans St., San Diego, CA 92106, USA, heidi@mpl.ucsd.edu), Catherine Berchok (Marine Physical Lab, Scripps Institution of Oceanography, 291 Rosecrans St., San Diego, CA 92106, USA, cberchok@ucsd.edu)

Since underwater sound is a critical aspect of the ocean environment for marine animals, passive acoustic recordings provide important information on marine habitats. Conversely, because biological sounds can be a dominant component of the ocean sound field, knowledge of their characteristics is important for assessing sonar system performance. This presentation summarizes the properties of three biological choruses observed in experiments off the southern California coast. Measurements were made with large aperture, well-filled hydrophone arrays, yielding highly resolved estimates of the chorus directionality. The first chorus generated energy in the 200-800 Hz band, occurred predominantly at night in spring and summer, and displayed an unusual spatiotemporal pattern where the region of chorusing periodically propagated upcoast over 25 km of coastline at nearly 1.5 km/sec. The other two choruses occurred in the 1-10 kHz frequency band and come from slowly-moving, or fixed, sources at, or near, the ocean bottom. One chorus was present only on two consecutive nights during a summer experiment, whereas the second occurred at all times throughout the 10-day duration of a late winter experiment. Numerical models predict some of the characteristics of the choruses. [Work supported by the Office of Naval Research].

8:40

2aAOa3. Understanding potential effects of using active sonar to study marine ecosystems. Mardi C. Hastings (Penn State University, Applied Research Laboratory, 801 North Quincy Street, Suite 120, Arlington, VA 22203-1708, USA, mch26@psu.edu)

Biological organisms can suffer various kinds of effects when exposed to high intensity sound. Active sonar systems used to investigate marine ecosystems typically have source levels exceeding 220 dB re: 1 μPa at 1 m, which are high enough to have an effect on organisms in relatively close proximity. Although these systems operate at frequencies from the low kilohertz to megahertz ranges that are not usually associated with potentially harmful effects in the marine environment, fundamental interactions with sound occur at various biological scales that strongly depend on the size of the organism with respect to acoustic wavelengths. These effects range from subtle changes in behavior to various types of trauma that can result in temporary or permanent hearing loss, or in hemorrhage or even mortality. Examples from the literature will be used to illustrate the basic relationships between biological effects of sound and the size and structure of marine organisms, as well as potential concerns associated with using active sonar to study the marine environment.

9:00

2aAOa4. How toothed whales echolocate to find and capture prey in the deep ocean. Peter Tyack (Woods Hole Oceanographic Institution, Applied Ocean Physics & Engineering Dept., Woods Hole, MA 02543, USA, ptyack@whoi.edu), Mark Johnson (Woods Hole Oceanographic Institution, Applied Ocean Physics & Engineering Dept., Woods Hole, MA 02543, USA, majohnson@whoi.edu), Peter T. Madsen (University of Aarhus, Biological Sciences, Zoophysiology, C. F. Møllers Allé, Building 1131, DK-8000 Aarhus, Denmark, peter.madsen@biology.au.dk), Walter M. Zimmer (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, walter@nurc.nato.int), Natacha A. Soto (University of La Laguna, BIOECOMAC Department of Animal Biology, 38207 La Laguna, Spain, naguilar@ull.es)

Sperm and beaked whales dive to feed on squid and deepwater fish. We have attached sound-and-orientation recording tags to study how these whales use echolocation to forage at depth. Tagged whales are usually silent when starting a dive, but start producing echolocation clicks at a few hundred meters depth, shallower than the depth at which they feed, suggesting that descending whales scan the deep layers where they will feed. Once sperm or beaked whales encounter prey, they switch from regular search clicks to a buzz of rapid clicks. Tags on beaked whales not only record outgoing clicks, but also echoes from prey at ranges out to 10-20 m. Beaked whales produce clicks every 0.2-0.4 sec when searching. Echoes from several targets are often detected after each beaked whale click. Beaked whales will pass by many targets before selecting one. Whales may switch from the search clicks to a buzz as they close within a body length of the prey. Sperm and beaked whales have an angular acceleration at the end of the buzz, which probably indicates turning to catch the prey.

9:20

2aAOa5. Hypotheses regarding exploitation of bubble acoustics by cetaceans. Timothy G. Leighton (Institute of Sound and Vibration, Univ. of Southampton, University Road, Highfield, SO17 1BJ Southampton, UK, T.G.Leighton@soton.ac.uk), Paul R. White (Institute of Sound and Vibration, Univ. of Southampton, University Road, Highfield, SO17 1BJ Southampton, UK, prw@isvr.soton.ac.uk), Daniel C. Finfer (Institute of Sound and Vibration, Univ. of Southampton, University Road, Highfield, SO17 1BJ Southampton, UK, d.c.finfer@soton.ac.uk)

Bubbles are the most acoustically active naturally occurring entities in the ocean, and cetaceans are the most intelligent. Having evolved over tens of millions of years to cope with the underwater acoustic environment, cetaceans may have developed extraordinary techniques from which we could learn. This paper outlines some of the possible interactions, ranging from the exploitation of acoustics by humpback whales (Megaptera novaeangliae) in bubble nets to trap prey, to techniques by which coastal dolphins (e.g. of the genus Cephalorhynchus) could successfully echolocate in bubbly water (a hypothesis which has led to the development of a man-made sonar which can penetrate bubble clouds, and a range of possibilities for homeland security).

9:40

2aAOa6. Marine ecosystem acoustics: contributions of sonar technology. Kenneth G. Foote (Woods Hole Oceanographic Institution, Woods Hole, MA 02543, USA, kfoote@whoi.edu)

Sonar is considered in the widest sense as an acoustic system consisting of a transmitter and receiver of underwater sound. It may be monostatic, bistatic, or multi-static; narrowband or broadband; with one or more fixed, rotating, or electronically formed beams. Transducers may be mounted on hulls of vessels and autonomous underwater vehicles, fixed structures, or seafloor; towed; or suspended. Spatial scales may be sampled from millimeters to thousands of kilometers; corresponding temporal scales vary from a fraction of a second, as in target-tracking, to seasons, as in observing animal migration. Several sonar types are described. Contributions of sonar technology to ecosystem studies, both actual and potential, are noted. These include, for example, fine-scale surveying of pelagic and semi-demersal fish by scientific echo sounder; mapping of the three-dimensional structure of fish aggregations and bathymetry by multibeam sonar; quantification of benthic egg beds of the squid Loligo opalescens by sidescan sonar; and large-scale surveying of swimbladdered fish by parametric sonar. Each of these sonar systems has potential for detection, quantification, and classification of marine organisms, as well as for characterization of the habitat, hence yielding the kind of information that is required for ecosystem assessment and management.

10:00

2aAOa7. Use of broadband active acoustics to study marine organisms. Timothy K. Stanton (Woods Hole Oceanographic Institution, Applied Ocean Physics & Engineering Department, 98 Water Street, MS #11, Woods Hole, MA 02543, USA, tstanton@whoi.edu), Dezhang Chu (NOAA/NMFS/NWFSC/FRAMD, Montlake Blvd., E. Seattle, WA 98112, USA, dchu@whoi.edu)

Broadband signals inherently have more information than narrowband signals. In essence, they have more ″channels″ of information due to the wider range of frequencies spanned. It is therefore advantageous to use broadband signals in active acoustic studies of marine organisms and to apply techniques that best exploit the broadband nature of the signals. In this presentation, a brief background on the use of narrowband systems for studying marine organisms by these authors and other investigators is given. Limitations to those studies are outlined, with arguments made for the use of broadband acoustics. The past twenty years of studies centered at the Woods Hole Oceanographic Institution are subsequently reviewed in which broadband signals are used both in the laboratory and field to study a variety of organisms-- swimbladder-bearing fish and three major anatomical categories of zooplankton. The analyses are divided broadly into two major categories-- time and frequency domain-- for various types of analyses, including pulse compression processing and spectral analysis. Results are first shown based on studies, one organism at a time, in the laboratory. Those laboratory approaches are then applied to ocean studies of fish and large zooplankton using a new towed instrument spanning the frequency range 1.7 kHz - 100 kHz.

TUESDAY MORNING, 1 JULY 2008 ROOM 342A, 10:40 A.M. TO 6:40 P.M.

Session 2aAOb

Acoustical Oceanography, Underwater Acoustics, Signal Processing in Acoustics, and ECUA: Geoacoustic Characterization of the Ocean Bottom and Geoacoustic Inversion I

Peter Gerstoft, Cochair
Marine Physical Laboratory, Scripps Institute of Oceanography, 8602 La Jolla Shores Drive, La Jolla, CA 92093-0238, USA

Dag Tollefsen, Cochair
Norwegian Defence Research Establishment (FFI), Box 115, Horten, 3191, Norway

Invited Papers

10:40

2aAOb1. Characterization of seabed geo-acoustic structure for a shallow water environment. David Knobles (Applied Research Laboratories, UT at Austin, P. O. Box 8029, Austin, TX 78713, USA, knobles@arlut.utexas.edu)

Full-field inversion methods over large frequency bands and spatial scales attempt to exploit the sensitivity of the frequency and spatial dependence of the acoustic field of the properties to the seabed. A comprehensive set of broadband impulsive and narrowband tonal acoustic recordings in the 5-3000 Hz band from two L-arrays separated by 20 km were made in a shallow water area on the New Jersey continental shelf during the Shallow Water 2006 (SW06) experiment. In addition, surface ship of opportunity passages on both arrays were recorded along with noise during the passage of tropical storm Ernesto. Reverberation and transmission loss data in the 25-9200 Hz band from explosive sources were also collected at a nearby site prior to the SW06 experiment. Numerous full-field geoacoustic inversion methods and analyses of all these acoustic data with supporting range and azimuth-dependent geo-physical measurements are employed to characterize the seabed properties. For example, the analyses of these data permit inferences about the frequency-dependence of the seabed sound speed and attenuation to be made. The overlapping data types act to reduce the intrinsic ambiguities associated with inversion, quantify the spatial variability, and serve as independent validation.

11:00

2aAOb2. Effect of ocean sound speed uncertainty on matched-field geoacoustic inversion. Chen-Fen Huang (Department of Marine Environmental Informatics/National Taiwan Ocean University, 2 Pei-Ning Road, 202 Keelung, Taiwan, chenfen@mail.ntou.edu.tw)

The effect of ocean sound speed uncertainty on matched field geoacoustic inversion is investigated using data from the SW06 experiment along a nearly range-independent bathymetric track. Significant sound speed differences were observed at the source and receiving array and several environmental parameterizations were investigated for the inversion including representing the ocean sound speed at both source and receivers with empirical orthogonal function (EOF) coefficients. A GA-based global optimization method was applied to the candidate environmental models. Then, a Bayesian inversion technique was used to quantify uncertainty in the environmental parameters for the best environmental model, which included an EOF description of the ocean sound speed.

11:20

2aAOb3. Quantifying the uncertainty of geoacoustic parameter estimates in a dynamic environment using oceanographic data. Ross Chapman (University of Victoria, 3800 Finnerty Rd, Vcitoria, BC V8W 3P6, Canada, chapman@uvic.ca), Yong-Min Jiang (University of Victoria, 3800 Finnerty Rd, Vcitoria, BC V8W 3P6, Canada, minj@uvic.ca)

This paper presents results of Bayesian matched field geoacoustic inversion in a temporally and spatially varying shallow water environment. The acoustic data were collected by a vertical line array at ranges of 1, 3 and 5 km in the SW06 (Shallow Water 06) experiment carried out in August 2006. The sound speed profiles measured at the array and at the source positions during the experiment showed significant variation in the thermocline in the middle part of the water column. The resulting uncertainty in the sound speed profile in the water impacts the performance of matched field inversion methods in estimation of geoacoustic model parameters. In previous work, an effective sound speed profile was estimated in the inversion, based on empirical orthogonal functions (EOFs) derived from a limited number of sound speed profiles measured at the time of the experiment. This paper explores the effect of using a larger number of profiles measured over several days in the SW06 experiment to include a greater diversity of ocean conditions in generating the EOFs. The inversion results for the two approaches are compared in terms of marginal probability distributions of the estimated geoacoustic parameters. (Work supported by Office of Naval Research).

11:40

2aAOb4. Short range geoacoustic inversion with a vertical line array. Yong-Min Jiang (University of Victoria, 3800 Finnerty Rd, Vcitoria, BC V8W 3P6, Canada, minj@uvic.ca), Ross Chapman (University of Victoria, 3800 Finnerty Rd, Vcitoria, BC V8W 3P6, Canada, chapman@uvic.ca), Peter Gerstoft (Marine Physical Laboratory, Scripps Institute of Oceanography, 8602 La Jolla Shores Drive, La Jolla, CA 92093-0238, USA, gerstoft@ucsd.edu)

During the SW06 (Shallow Water 06) experiments, short range geoacoustic inversion experiments were carried out at a source-receiver range of 230 m to determine the sea bed properties at fine spatial scales. Acoustic signals were collected at 16 hydrophones on a vertical line array from source depths of 15-65 m in 10 m intervals. For this experimental geometry, the angular coverage for reflectivity versus grazing angle is 7-25°. This paper combines the information extracted from low-frequency (100-900 Hz) and mid-frequency (1500-4500 Hz) LFM signals transmitted from the same source position to estimate multi-layered geoacoustic models. The model parameterization was based on the number of resolved sub-bottom reflections in the data. An inversion approach using adaptive simplex differential evolution was applied to the data to invert for the sub-bottom sound speeds and layer thicknesses. The temporal variation of the water column sound speed profile was included in the inversion. The estimated sound speeds are consistent with the range-averaged profile that was estimated previously from longer range data for the same radial path from the array. (Work supported by Office of Naval Research).

12:00

2aAOb5. Does a depth variable sound-speed profile matter for SW06 geoacoustic inversion? Alexandra Tolstoy (ATolstoy Sciences, 1538 Hampton Hill Circle, McLean, VA 22101, USA, atolstoy@ieee.org)

This work will discuss the effects of depth variability for a sound-speed profile with regard to goeacoustic inversion on some simulated SW06 ″data″. First, broadband (400-800 Hz) simulated ″data″ are generated using RAM_PE (by Collins) in the time domain for some short range (less than 1 km) SW06 scenarios. Next, the first 4 boundary reflection arrivals (from surface, bottom, surface-bottom, bottom-surface) are studied as a function of source range and depth, phone depth, and sound-speed variability. Finally, we examine the effects of sound-speed depth variability on geoacoustic inversion via MFP. We note that knowledge of array phone locations has a large impact on inversion capabilities.

12:20

2aAOb6. Three dimensional geoacoustic inversion on the New Jersey shelf. Megan S. Ballard (Penn. State University, P.O. Box 30, State College, PA 16804, USA, msd200@psu.edu), K.m. Becker (Penn. State University, P.O. Box 30, State College, PA 16804, USA, kmbecker@psu.edu)

Perturbative inversion, based on a linearized relationship between sound speed in the sediment and modal eigenvalues, is applied to data from the Shallow Water Experiment 2006. Data were collected by towing a low-frequency sound source out and back along radials, spanning a 90 degree angular sector, from a common receiver location. Range-dependent estimates of horizontal wave numbers are obtained along each of the radials using high-resolution signal processing techniques capable of detecting and localizing changes in sub-bottom properties, and that are particularly sensitive to changes in layer structure. Wave number estimates at each range are used in a linearized inversion algorithm to estimate local sediment properties. Locations of the R-reflector and other layering information are used as a priori information in the inversion algorithm. The additional information both constrains the solution of an otherwise ill-posed problem and emphasizes the layered structure of the sediment. These methods have been shown to yield accurate estimates of the sound speed profile deep into the sediment using very few perturbations to the forward model. Combining the local inversion results, a three-dimensional map of sediment sound speed structure is obtained for a 25 km2 region of the seafloor. [Supported by NDSEG and ONR]

12:40

2aAOb7. Experimental verification of range-dependent inverse method for geoacoustic parameters from modal dispersion data. Subramaniam Rajan (Scientific Solutions, Inc, 99 Perimeter Road, Nashua, NH 03063, USA, srajan@scisol.com), K.m. Becker (Penn. State University, P.O. Box 30, State College, PA 16804, USA, kmbecker@psu.edu)

The procedure for determining the sediment compressional wave speed profile from modal dispersion data in a range-independent case is well documented in the literature. In this paper, the procedure developed for the range-independent case is extended to cover the range-dependent case. In this procedure the region under investigation is divided into discrete range intervals and the sediment characteristics at each range interval estimated assuming that the sediment properties of each range interval are range independent. The approach has been demonstrated to yield range-dependent sediment properties when applied to synthetic data generated for an experiment consisting of multiple source/receiver combinations. During the Shallow Water Experiment 2006, broadband data transmitted by a single source were measured on multiple receivers. These data are used here to estimate range-dependent properties of the sediment. The results obtained in this manner are compared with inversion results obtained for the same region by other investigators who employed different approaches for estimating the geoacoustic properties. [Work supported by ONR, Ocean Acoustics].

1:00-2:00 Lunch Break

Invited Paper

2:00

2aAOb8. Modal inversion using the acoustic field emitted by a moving source and measured on a vertical line array of hydrophones. K.m. Becker (Penn. State University, P.O. Box 30, State College, PA 16804, USA, kmbecker@psu.edu), Megan S. Ballard (Penn. State University, P.O. Box 30, State College, PA 16804, USA, msd200@psu.edu)

As part of the Shallow Water Experiment 2006 (SW06), a low-frequency continuous wave acoustic source was towed out and back along radials from a fixed vertical line array of hydrophones. The resulting modal pressure field, measured on synthetic range apertures created by the relative source motion, is used to derive input data to inversion algorithms for estimating bottom properties. The inversion algorithms considered are based on well established relationship between modal eigenvalues and geoacoustic properties of the sediment. In this talk, linear inversion methods based on modal input data will be reviewed for their application to the SW06 data. The first step in these methods is the accurate estimation of modal eigenvalues from the pressure field data. For a moving source, theory predicts a Doppler shift along with frequency spreading proportional to both the transmitted frequency and tow speed. Propagating modes and their corresponding modal eigenvalues are predicted to correspond with the Doppler shifted frequencies. An emphasis of this work is on Doppler shift as an observable in the data and properly accounting for it in inversion. Sediment sound speeds estimated from data are consistent with sandy sediments found in the experimental area. [Work supported by ONR and NDSEG]

Contributed Paper

2:20

2aAOb9. Geoacoustic inversion using combustive sound source signals. Gopu R. Potty (University of Rhode Island, Department of Ocean Engineering, Narragansett Bay Campus, Narragansett, RI 02882, USA, gopu@uri.edu), James H. Miller (University of Rhode Island, Department of Ocean Engineering, Narragansett Bay Campus, Narragansett, RI 02882, USA, miller@egr.uri.edu), Preston S. Wilson (The University of Texas, Mechanical Engineering Department, Austin, TX 78712, USA, pswilson@mail.utexas.edu), James F. Lynch (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, jlynch@whoi.edu), Arthur Newhall (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, anewhall@whoi.edu)

In the summer of 2006, Combustive Sound Sources (CSS) were deployed in shallow waters off New Jersey during the Shallow Water experiment (SW 06). The depth of these sources were 26 m in water depths of the order of 100 m. CSS are low frequency broadband sound sources and suitable for modal dispersion based inversion. CSS data collected on various receivers (Single Hydrophone Receiving Units (SHRU), Vertical Line Array, etc.) will be used for the inversions. The inversions are based on an iterative scheme using a new time-frequency analysis technique (dispersion based short-time Fourier transform) in which each time-frequency tiling is adaptively rotated in the time-frequency plane, depending on the local wave dispersion. Data from various receivers will be used to investigate the spatial variation in compressional wave speed. Compressional wave attenuation will also be estimated using modal amplitude ratios. Inversion results will be validated by comparing the data with model predictions. [Work supported by the Office of Naval Research]

Invited Paper

2:40

2aAOb10. Tomographic and bottom geoacoustic inversions using Genetic Algorithms and a statistical characterization of the acoustic signal. Michael Taroudakis (University of Crete & FORTH/IACM, Vassilika Vouton, P.O.Box 1385, 711 10 Heraklion, Greece, taroud@iacm.forth.gr), Costas Smaragdakis (University of Crete, Dept of Mathematics, Knossou Ave, 71409 Heraklion, Greece, kesmarag@gmail.com)

The paper deals with the problem of estimation of the parameters of the water column and/or the sea bed, using measurements of the acoustic field due to a known source at a certain point in the water column. Following the work by Taroudakis et al. (JASA 119, 1396-1405 (2006)) the acoustic signal is characterized using the statistics of the wavelet sub-band coefficients, which obey a certain statistical law, described by an Alpha-Stable distribution. Thus, the signal observables are the set of the parameters of the appropriate distributions at the various levels of the signal decomposition. In this work an inversion procedure based on a Genetic Algorithm and the statistical characterization of the acoustic signal, is described. The quality of a certain population of candidate model parameters (properties of the water columns and/or the sea-bed) is evaluated using the Kullback-Leibler divergence (KLD) of the wavelet sub-band coefficient distributions, between the measured and simulated acoustic signals. Following an appropriate population regeneration procedure, the final population is described by an a-posteriori statistical distribution of the model parameters, indicating among the others their sensitivity in the inversion procedure. Very good inversion results have been observed in simulated shallow water environments.

Contributed Paper

3:00

2aAOb11. Dispersion curve estimation with partcile filters for geoacoustic inversion. Ivan Zorych (Department of Mathematical Sciences, New Jersey Institute of Technology, 323 M L King Blvd, Newark, NJ 07102, USA, zorych@gmail.com), Zoi-Heleni Michalopoulou (Department of Mathematical Sciences, New Jersey Institute of Technology, 323 M L King Blvd, Newark, NJ 07102, USA, michalop@njit.edu)

In this work we extend our particle filtering method for dispersion curve extraction from spectrograms of acoustic signals that have propagated in underwater environments, the goal being to obtain accurate representation of modal dispersion for geoacoustic inversion. The approach combines particle filtering with modeling of sound propagation in ocean environments to track dispersion curves of multimodal signals in noisy environments. In addition to providing connected modal ″trajectories″ that facilitate the computation of maximum a posteriori estimates of modal arrival times, the method provides posterior probability distributions for arrival times, quantifying errors that are then propagated into the geoacoustic inversion process. The method, thus, allows the calculation of posterior probability distributions of geoacoustic parameters. We present results from both synthetic and real data from the Gulf of Mexico experiment. [Work supported by ONR.]

Invited Paper

3:20

2aAOb12. Geoacoustic characterization of the seafloor from a subbottom profiler applied to the BASE'07 experiment. Gwladys Theuillon (SHOM, 13 rue du Chatellier, CS 92803, 29228 Brest cedex 2, France, gwladys.theuillon@shom.fr), Yann Stephan (SHOM, 13 rue du Chatellier, CS 92803, 29228 Brest cedex 2, France, yann.stephan@shom.fr)

The most recent subbottom profilers present good performances in term of signal to noise ratio, resolution and penetration. These devices can thus be used to infer quantitatively the geoacoustic parameters of the seafloor. We have previously developed inversion methods which aim to estimate absorption, reflection, penetration, impedance contrast and micro-roughness in sediments from the SBP 120 subbottom profiler, manufactured by Kongsberg. These methods have been tested against real data and geoacoustic parameters derived from the SBP 120 are fully coherent with in situ measurements, which tends to confirm the possibility of seafloor characterization with subbottom profilers. In this work, the inversion methods are applied to a set of data acquired on the Malta Plateau during BASE'07 experiment. Geoacoustic parameters results are presented and discussed. They are in good agreement with the a priori knowledge of sediment properties in the area.

3:40-5:20 Posters

Lecture sessions will recess for presentation of poster papers on various topics in acoustics. See poster sessions for topics and abstracts.

Invited Paper

5:20

2aAOb13. Bayesian model parameterization selection for seabed reflection-coefficient inversion. Jan Dettmer (University of Victoria, School of Earth and Ocean Sciences, Victoria, BC V8W 3P6, Canada, jand@uvic.ca), Stan E. Dosso (University of Victoria, School of Earth and Ocean Sciences, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca), Charles W. Holland (Penn State University / Applied Research Lab, Applied Science Bldg., State College, PA 16804, USA, cwh10@psu.edu)

This paper considers Bayesian inversion of seabed reflection-coefficient data for multi-layer geoacoustic models; in particular, the important issue of determining appropriate model parameterizations (e.g., number of layers, sound speed and density variations within layers). A poor model parameterization can lead to unreasonable inversion results. In particular, under-parameterization can introduce theory error in the inversion, causing biased results. However, more complex models always fit the data better. Therefore, parameterization quality cannot be quantified only in terms of data misfit, and Occam's razor is commonly applied to prefer simple parameterizations. In a Bayesian framework, Occam's razor is inherently included through Bayesian evidence. Bayesian inversion can be associated with two levels of inference. The first level assumes a specific model parameterization, and quantifies the data information content. This paper focuses on the second level: using Bayesian evidence to compare different model parameterizations. To this end, Gibbs sampling is applied, including full error covariance estimation, to sample the posterior probability density (PPD) for various parameterizations. Bayesian evidence is then computed from the PPD samples by numerical integration (reverse importance sampling).

Contributed Papers

5:40

2aAOb14. Combined inversion of mid-frequency propagation and reverberation sonar data. Peter Louring Nielsen (NURC, Viale S. Bartolomeo 400, 19126 La Spezia, Italy, nielsen@nurc.nato.int), Christopher H. Harrison (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19138 La Spezia, Italy, harrison@nurc.nato.int)

Sonar performance predictions in shallow-water regions are strongly dependent on a good knowledge of the geoacoustic and scattering properties of the seabed. The bottom properties are probably the most difficult parameters to determine, but inversion of measured acoustic data to infer the geoacoustic and scattering properties is a feasible technique. One of these techniques relies on inverting: (1) propagation data to estimate local bottom properties and (2) long-range reverberation data providing effective bottom properties over larger areas. A matched-field inversion approach is applied to propagation and reverberation data received on a towed horizontal array during the BASE'04 experiment conducted on the Malta Plateau, Mediterranean Sea. A total of 30 transmissions along a 10-km track were used to capture eventual range-dependent bottom properties. The inversion algorithm is composed of efficient prediction tools which can provide environmental parameter estimates within tactical time frames for in-situ sonar performance predictions. The experimental set-up mimics a mid-frequency active sonar system using only the suite of sensors available on the vessel towing the sonar system; also known as the through-the-sensor technique. The impact on sonar performance utilizing the environmental characterization approach is shown for various experimental scenarios and seasons. [Research sponsored by NURC and the BOUNDARY Partners].

6:00

2aAOb15. Reverberation inversion for seabed properties. Stan E. Dosso (University of Victoria, School of Earth and Ocean Sciences, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca), Peter Louring Nielsen (NURC, Viale S. Bartolomeo 400, 19126 La Spezia, Italy, nielsen@nurc.nato.int), Christopher H. Harrison (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19138 La Spezia, Italy, harrison @nurc.nato.int)

This paper applies nonlinear inversion to reverberation and/or propagation data recorded on a towed horizontal array for geoacoustic and scattering properties of the seabed. A Bayesian inversion approach is applied to estimate properties of the posterior probability density, such as the maximum a posteriori (most-probable) model, 1-D and 2-D marginal probability distributions, and covariance/correlation matrices, providing optimal parameter estimates and quantifying parameter uncertainties and inter-relationships. Of particular interest is quantifying the information content of different combinations of data (e.g., reverberation or propagation data; joint inversion of reverberation and propagation data) to resolve seabed parameters, and understanding strong parameter inter-relationships (correlations) which, in some cases, act as the limiting factor in resolving geoacoustic/scattering parameters. The inversion is also applied to shallow-water reverberation and propagation data recorded in Mediterranean Sea.

6:20

2aAOb16. Local bottom characterization using an autonomous underwater vehicle. Peter Louring Nielsen (NURC, Viale S. Bartolomeo 400, 19126 La Spezia, Italy, nielsen@nurc.nato.int), Charles W. Holland (Penn State University/Applied Research Lab, Applied Science Bldg., State College, PA 16804, USA, cwh10@psu.edu)

In the past decade the usage of autonomous underwater vehicles (AUV) to sample properties of the underwater environment has increased. The advantages of using platforms are their autonomy and that operations can be performed covertly. In 2007, the CLUTTER'07 experiment was conducted on the Malta Plateau with the main aim of characterizing the underwater environment. An AUV was deployed at a particular site on the Plateau to demonstrate the feasibility to infer bottom properties using such an autonomous platform. The AUV was equipped with 2 sound sources covering a frequency band 800-3500 Hz, and the transmitted signals were acquired on a bottom moored vertical array. The mission of the AUV was to perform a linear track of ~1000 m passing the vertical array as close as possible while transmitting every second from the sound sources. This experimental configuration is similar to the move-out or wide-angle reflection measurements. The received signals are inverted for geoacoustic properties using both matched-field techniques and processed for direct bottom reflection properties. The results obtained are compared to independent findings from different experiments using various types of equipment at different seasons in the same area. [Research sponsored by NURC, ONR OA321 and the CLUTTER Partners]

TUESDAY MORNING, 1 JULY 2008 ROOM 352B, 8:00 A.M. TO 2:40 P.M.

Session 2aBB

Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Ultrasound Contrast Agents for Imaging I

Charles C. Church, Cochair
University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677, USA

Ayache Bouakaz, Cochair
INSERM U930, 2, Bvd Tonnelle, Tours Cedex 9, 37044, France

Invited Paper

8:00

2aBB1. Ultrasound and microbubble interaction. Nico De Jong (Erasmus MC, Dr Molewaterplein 50 room Ee2302, 3015GE Rotterdam, Netherlands, n.dejong@erasmusmc.nl)

Ultrasound contrast agent (UCA) bubble response to ultrasound (US) is still the subject of several studies, since it determines the power of medical diagnostic imaging modalities. Generally, bubbles are studied using optical or acoustical methods, both having their specific advantages, in an environment that mimic the clinical practice as good as possible (small-diameter capillaries, red blood cells, blood viscosity). Acoustical observations could provide a better insight in the acoustical scattering of bubbles, which determines the final efficacy of contrast-enhanced diagnostic US imaging, and are better suitable to determine the properties of populations of bubbles. Optical observations allow for easier characterization of fewer bubbles and a more detailed characterisation in case where bubbles vibrate non spherically, which is expected for bubbles nearby a wall or red blood cells. In this presentation, we show optical recordings of the various bubble responses with the ultrahigh speed Brandaris-128 camera, and discuss the clinical implications of our findings.

8:20

2aBB2. Nonlinear pulsing schemes for the detection of ultrasound contrast agents. Michalakis A. Averkiou (University of Cyprus/Dept. of Mech. Engineering, 75 Kallipoleos Str., 1678 Nicosia, Cyprus, maverk@ucy.ac.cy), Christophoros Mannaris (University of Cyprus/Dept. of Mech. Engineering, 75 Kallipoleos Str., 1678 Nicosia, Cyprus, mannaris@ucy.ac.cy), Matthew Bruce (SuperSonic Imagine, Les Jardins de la Duranne - Bât. E, 510, rue René Descartes, F - 13857 Aix-en-Provence, France, matt.f.bruce@gmail.com), Jeffry Powers (Philips Medical Systems, 22100 Bothell Everett Hwy, Bothell, WA 98021, USA, jeff.powers@philips.com)

Ultrasound contrast agents are used in cardiology for the assessment of myocardial perfusion and in radiology for the detection and characterization of tumors. One widely used approach of imaging contrast agents is to use a low Mechanical Index (*MI*) nonlinear imaging technique to avoid bubble destruction and image both the macro- and micro-circulation in real-time. Various pulsing schemes are employed for the detection of nonlinear echoes from contrast microbubbles. The objective of this paper is to evaluate the various pulsing schemes for low *MI* imaging of contrast microbubbles and better understand their similarities and differences. The pulsing schemes considered are pulse inversion, power modulation, and their combination. Emphasis is placed on identifying whether nonlinearity due to propagation in tissue may be discriminated from nonlinearity due to scattering from bubbles. Bubble destruction (use of high *MI*) and tissue motion were not considered in this work. The evaluation of the different pulsing schemes was performed with numerical simulations from well established theoretical models and experimental data from microbubbles in tissue phantoms and human tissues.

8:40

2aBB3. Contrast agent response to chirp reversal. Anthony Novell (INSERM U930, CHRU Bretonneau, 37044 Tours Cedex 9, France, anthony.novell@etu.univ-tours.fr), Sander Van Der Meer (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, s.m.vandermeer@utwente.nl), Michel Versluis (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, m.versluis@utwente.nl), Nico De Jong (Erasmus MC, Dr Molewaterplein 50 room Ee2302, 3015GE Rotterdam, Netherlands, n.dejong@erasmusmc.nl), Ayache Bouakaz (INSERM U930, 2, Bvd Tonnelle, 37044 Tours Cedex 9, France, bouakaz@med.univ-tours.fr)

We investigate an excitation approach for contrast agents based on chirps. This technique, named chirp reversal, consists in transmitting an up sweep frequency chirp (UPF) followed by a down sweep frequency chirp (DNF). Simulations using a modified Rayleigh-Plesset equation were carried out. Chirps with center frequencies from 1.4 MHz to 2 MHz, pressures from 50 kPa to 200 kPa and frequency bandwidths from 30% to 65% were considered. High speed optical observations and acoustical measurements were performed using individual contrast bubbles of radii from 1 μm to 5 μm and a diluted solution of contrast agent respectively. Simulations showed differences between bubbles' oscillations following UPF and DNF chirps in terms of amplitude and duration. Maximal differences occurred for bubbles that were around 80% and 140% of the resonance size. Bubbles at resonance or far away from resonance provided identical responses to UPF and DNF chirps. Larger bandwidths and higher acoustic pressures accentuate further the difference between the UPF and DNF responses. These findings were confirmed through optical data and acoustical measurements. The results reveal the potential of chirp reversal for contrast agent detection.

9:00

2aBB4. Dual-frequency insonation of single microbubbles. Marcia Emmer (Erasmus MC, Ee2302, P.O. Box 2040, 3000 CA Rotterdam, Netherlands, m.emmer@erasmusmc.nl), Hendrik J. Vos (Biomedical Engineering, Erasmus MC, P.O. Box 2040, 3000 CA Rotterdam, Netherlands, H.J.Vos@ErasmusMC.nl),Nico De Jong(Erasmus MC, Dr Molewaterplein 50 room Ee2302, 3015GE Rotterdam, Netherlands, n.dejong@erasmusmc.nl)

Radial modulation imaging is a new medical imaging technique based on dual-frequency insonation of ultrasound contrast agents. The difference in echo between a high frequency 'imaging' pulse transmitted at either the compression or rarefaction phase of a low frequency 'modulating' pulse is detected by regular correlation techniques. Little is however known about the contrast agent microbubble dynamics in a dual-frequency ultrasound field, which were investigated in this study. Using a high-speed camera system, the radial excursions of single phospholipid-coated microbubbles were recorded. The microbubbles were simultaneously insonified with 2.5 cycles pulse at 0.5 MHz and 30 kPa and a 32 cycles pulse at 3.75 MHz and 80 kPa. The microbubbles studied had diameters ranging from 1.1 - 5.2 μm. Microbubbles smaller than 1.4 μm frequently showed shrinkage. Microbubbles larger than 2.6 μm showed low ($<$ 8 dB) or no amplitude modulation of the high frequency radial excursion. Microbubbles with diameters between 1.4 and 2.6 μm showed high amplitude modulation (up to 25 dB) and strong compression-only oscillation, which both may be explained by nonlinear shell properties. The observed behaviour is beneficial for the detection of contrast agents.

9:20

2aBB5. Sub-harmonic response from polymer-shelled contrast agents with a 40-MHz excitation. Jeffrey A. Ketterling (Frederic L. Lizzi Center for Biomedical Engineering, Riverside Research Institute, 156 William St., New York, NY 10038, USA, ketterling@rrinyc.org), Jonathan Mamou (Riverside Research Institute, 156 William St., 9th Floor, New York, NY 10038, USA, mamou@rrinyc.org)

There is a growing interest in using acoustic contrast agents with high-frequency ultrasound ($>$ 15 MHz) in order to better visualize microcirculation. Experiments were performed with polycaprolactone-shelled agents (POINT Biomedical, San Carlos, CA) having mean diameters of 0.56, 1.1, and 3.4 μm. The agents were heavily diluted in filtered water and injected through a 200 μm channel into the focal zone of a 40-MHz transducer that had a focal length of 12 mm and an outer diameter of 6 mm. Backscatter signals from single agents were digitized using tone bursts of 5 to 20 cycles at peak-negative pressures of 0.6 to 6.3 MPa. 1000 valid single-bubble backscatter events at each exposure condition were digitized and then analyzed for 20-MHz subharmonic content. The data showed that the subharmonic response was initiated between 5 and 10 cycles and the likelihood of a subharmonic event increased as the number of cycles increased. A subharmonic backscatter response was most likely at 3.9 MPa for the 3.4 μm agent and 1.7 MPa for the 0.56 and 1.1 μm agents. The increased pressure for subharmonic activity for larger agent was consistent with its larger size.

9:40

2aBB6. Combined optical and acoustical characterization of single ultrasound contrast agent microbubbles. Jeroen Sijl (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, j.sijl@utwente.nl), Timo Rozendal (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, t.rozendal@student.utwente.nl), Hendrik J. Vos (Biomedical Engineering, Erasmus MC, P.O. Box 2040, 3000 CA Rotterdam, Netherlands, H.J.Vos@ErasmusMC.nl), Benjamin Dollet (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, benjamin.dollet@univ-rennes1.fr), Nico De Jong (Erasmus MC, Dr Molewaterplein 50 room Ee2302, 3015GE Rotterdam, Netherlands, n.dejong@erasmusmc.nl), Detlef Lohse (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, d.lohse@utwente.nl), Michel Versluis (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, m.versluis@utwente.nl)

Contrast enhancement in medical ultrasound imaging is provided by the non-linear characteristics of coated microbubbles used as Ultrasound Contrast Agents (UCA). Optical time-resolved observations of the UCA mi-

crobubble dynamics have revealed new non-linear bubble behavior such as ″compression only″ and ″thresholding″ behavior. Up to now, the contributions of such behavior to the non-linear acoustic response of UCA microbubbles is not known. Theoretically, the sound emission of an oscillating microbubble is derived from the unsteady Bernoulli equation and mass conservation assuming incompressible and irrotational flow. An experimental validation of this relation between the radial dynamics and the sound emission of a microbubble is not straightforward. We present a combined optical and acoustical setup to characterize individual BR-14 (Bracco Research S.A., Geneva, Switzerland) UCA microbubbles. The bubbles were isolated in a capillary fiber by an active flow control. During insonation the radial response of the single microbubble was recorded with the Brandaris ultra high-speed camera while the resulting acoustic response was measured with an accurately calibrated sensitive transducer. The sound emission calculated from the measured radius-time curves gives excellent quantitative agreement with the directly measured sound emission for both the linearly oscillating microbubbles and bubbles displaying ″compression-only″ and ″thresholding″ behavior, which indeed resulted in a strong non-linear sound emission.

Invited Paper

10:00

2aBB7. Secondary Bjerknes forces between ultrasound contrast agent microbubbles. Michel Versluis (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, m.versluis@utwente.nl), Valeria Garbin (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, v.garbin@tnw.utwente.nl), Benjamin Dollet (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, benjamin.dollet@univ-rennes1.fr), Leen Van Wijngaarden (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, l.vanwijngaarden@tnw.utwente.nl), Nico De Jong (Erasmus MC, Dr Molewaterplein 50 room Ee2302, 3015GE Rotterdam, Netherlands, n.dejong@erasmusmc.nl), Detlef Lohse (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, d.lohse@utwente.nl)

Acoustic radiation forces arise on UCA microbubbles from pressure gradients in the ultrasound pressure field. The ultrasound wave emitted by an oscillating UCA microbubble and experienced by a neighboring bubble results in the so-called secondary Bjerknes force. Typically, only the (time) averaged value of the secondary Bjerknes force is estimated from experiments. Here, the ability to resolve in time the radial dynamics of UCAs and the evolution of their relative distances, allows us to obtain a quantification of the instantaneous secondary Bjerknes force. Two bubbles were selected and confined within two separate optical traps and positioned away from the sample chamber wall prior to insonation. The radius-time curves and distance-time curves were then extracted from high-speed optical recordings. The camera fully resolved the alternating attractive-repulsive features of the secondary Bjerknes force in time. The effect of the time averaged secondary Bjerknes force results in a net attraction of the two bubbles, with a typical observed displacement of 1-2 μm, with an instantaneous peak value of the secondary Bjerknes force of up to 10^{-6} N. We predict the mutual interaction of the two coated bubbles in their translation with an accuracy better than 10%.

10:20-10:40 Break

Contributed Paper

10:40

2aBB8. Mathematical models for contrast bubble dynamics. Charles C. Church (University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677, USA, cchurch@olemiss.edu), John S. Allen (University of Hawaii-Manoa, Department of Mechanical Engineering, Holmes Hall Room 302, Honolulu, HI 96822, USA, alleniii@hawaii.edu)

Theoretical investigation of the acoustic responses of albumin-encapsulated microspheres began over fifteen years ago when Albunex, the first agent approved for clinical use in the US, was still in development. Since that time, the number of potential ultrasound contrast agents has grown considerably. Depending on the agent, the shell may comprise a layer of proteins, synthetic polymers, surfactants, or lipids with a thickness from a few nm to 500 mm and surrounding air, sulfur hexafluoride, perfluorocarbon, or other gas. A thorough understanding of the interaction between ultrasound pulses and contrast microbubbles is essential for the successful clinical application of a particular agent. In this talk, the behavior of various contrast agents will be discussed, and appropriate models for each will be described. The basis for each of these theories is a free bubble model ″supplemented″ by the effect of the encapsulating shell. The differences among these models lie primarily in their treatment of the encapsulating layer and, to some extent, the surrounding medium. Comparisons among models will include predictions of radial responses, thresholds for bubble destruction, and clinically significant acoustic properties (resonance frequency, scattering strength, nonlinearity, etc.). [Supported by NIH 2 RO1 EB000350-04A2 (CCC) and by NIH 2 G12RR003061-21 (JSA).]

Invited Papers

11:00

2aBB9. Ultrasound contrast agents: from concept to clinical use. François Tranquart (INSERM U930, 2, Bvd Tonnelle, 37044 Tours Cedex 9, France, tranquart@med.univ-tours.fr), Ayache Bouakaz (INSERM U930, 2, Bvd Tonnelle, 37044 Tours Cedex 9, France, bouakaz@med.univ-tours.fr), Aurore Bleuzen (INSERM U930, 2, Bvd Tonnelle, 37044 Tours Cedex 9, France, bleuzen@chu-tours.fr), Peggy Palanchon (INSERM U930, 2, Bvd Tonnelle, 37044 Tours Cedex 9, France, palanchon@med.univ-tours.fr), Jean-Michel Correas (Hopital Necker, 75015 Paris, France, jean-michel.correas@nck.ap-hop-paris.fr)

Contrast ultrasound imaging techniques are of interest for almost every clinical application in all organs including not only liver and kidney but also superficial organs and brain or lungs. This technique is mainly powerful in terms of lesion detection and characterization especially for liver with diagnostic value greater than 90%. This was the case for all categories of lesions (with values nearly 100% for liver metastases, FNH) but the accuracy was slightly lower for HCC. Specific recommendations were established by EFSUMB for the use of contrast agents in liver, pancreas, kidney, trauma and brain imaging for a proper and safe use to improve diagnosis accuracy. On the other hand, contrast-enhanced ultrasound may be used in evaluating response to therapy. The follow-up of vascularization under specific targeted treatments offers the capacity to early diagnose positive or negative local response for an adaptation of therapeutic way

before or in absence of tumour size changes. This non invasive method could be repeated without adverse events early after starting this therapy in order to depict local response or recurrence for a better adaptation of dose with a strong impact on cost and limitation of adverse events.

11:20

2aBB10. Dynamics of microbubbles targeted to surfaces: numerical and experimental modelling. Richard Manasseh (CSIRO, Fluid Dynamics Group, PO Box 56, Highett, VIC 3190 Melbourne, Australia, Richard.Manasseh@csiro.au), Edward Payne (University of Melbourne, Department of Mechanical & Manufacturing Engineering, Parkville, VIC 3010 Melbourne, Australia, Edward.Payne@csiro.au), Andrew Ooi (University of Melbourne, Department of Mechanical & Manufacturing Engineering, Parkville, VIC 3010 Melbourne, Australia, a.ooi@unimelb.edu.au)

Numerical calculations and illustrative experiments are presented on the volumetric oscillations of microbubbles on and near surfaces. There is a considerable theoretical and experimental literature on the acoustic interactions of bubbles. In the present study, the surface was represented by a mirror-image bubble and the nonlinear frequency response calculated by integrating acoustically coupled sets of Rayleigh-Plesset-like equations. A significant shift was found in the peak nonlinear response frequency of a bubble targeted onto a surface. This effect is increased when other bubbles are nearby on the surface. Owing to the asymmetric influence of the surface, experimental images were dominated by shape-mode instabilities, making optical determination of the peak nonlinear response frequency difficult. Moreover, it was found that even if bubbles are separated by only a small fraction of the sound wavelength, time delays owing to the finite speed of sound have a surprisingly significant influence. Calculations on the symmetric mode of mutual oscillation showed that the introduction of time delays significantly modified harmonics of the spectrum.

11:40

2aBB11. Echogenic liposomes for image-guided drug delivery. Christy K. Holland (University of Cincinnati, Biomedical Engineering, MSB, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, USA, Christy.Holland@uc.edu), David D. McPherson (University of Texas Health Science Center, 6431 Fannin St., MSB 1.252, Houston, TX 77030, USA, David.D.McPherson@uth.tmc.edu)

Echogenic liposomes (ELIP) are under development to enable ultrasound-controlled drug delivery. Ultrasound-triggered release of hydrophilic and lipophilic agents was assessed from circulating ELIP in vitro using color Doppler ultrasound (6 MHz). Calcein, or recombinant tissue plasminogen activator (rt-PA), both hydrophilic agents, or papaverine, a lipophilic agent, were each loaded into ELIP and diluted in 0.5% bovine serum albumin. Ultrasound-triggered release of calcein, rt-PA or papaverine from ELIP was determined relative to detergent and untreated controls. Calcein concentration was measured by fluorimetry and release of rt-PA was assayed with a chromogenic substrate and a spectrophotometric method. Papaverine concentration was quantified by absorbance spectrophotometry and the amount of papaverine associated with P-ELIP was determined using a spin column filtration technique. Dynamic changes in echogenicity were assessed with low output B-mode ultrasound (0.04 MI) as mean digital intensity. Treatment with color Doppler ultrasound resulted in a statistically significant amount of calcein and rt-PA release from liposomes ($p < 0.01$), but did not induce papaverine release ($p > 0.05$). The differential efficiency of ultrasound-mediated pharmaceutical release from ELIP for water-soluble and lipid-soluble compounds suggests that water-soluble drugs are better candidates for the design and development of ultrasound-controlled drug delivery systems.

Contributed Papers

12:00

2aBB12. Ultrasound contrast agents in an in vivo murine melanoma model. François Tranquart (INSERM U930, 2, Bvd Tonnelle, 37044 Tours Cedex 9, France, tranquart@med.univ-tours.fr), Ayache Bouakaz (INSERM U930, 2, Bvd Tonnelle, 37044 Tours Cedex 9, France, bouakaz@med.univ-tours.fr), Sophie Serriere (INSERM U930, 2, Bvd Tonnelle, 37044 Tours Cedex 9, France, serrie_s@med.univ-tours.fr)

The purpose of the study was to test different types of microbubble contrast agents (SonoVue®, Definity® and targeted microbubbles against vascular endothelial growth factor receptors (KDR, antiCD31)) to quantify angiogenesis. After SonoVue® and Definity® injections, a strong, rapid and heterogeneous signal enhancement was detected in all tumors. According to the tumor size, the quantification of the perfused area revealed major inter-individual variations. Three groups of animals bearing tumors, following SonoVue® administration, were arbitrarily constituted to compare various sonographic parameters such as AUC, mTT, etcœ. The tumoral size increase seemed negatively correlated to a decrease in all quantified parameters. Non linear acoustic signal from microbubbles targeted to the molecular site was determined by an ultrasound-based destruction-reperfusion scheme. In tumor-bearing mice, an increase of the retention time (>10 minutes) was revealed following KDR and antiCD31 targeted microbubbles administrations when compared to the retention time of non targeted microbubbles. The amount of remaining targeted bubbles remains always lower than non specific agents. Results showed that quantitative contrast-enhanced ultrasound imaging could be an effective method for monitoring angiogenesis process in mice and so could be used for the follow-up of tumours under specific treatment.

12:20

2aBB13. Ultrasound contrast imaging of angiogenisis in a murine tumor model. Olivier Lucidarme (CHU Pitié-Salpétrière, 47-83 Boulevard de l'Hôpital, 75651 Paris, France, olivier.lucidarme@psl.aphp.fr), Laurent Taillade (Laboratoire d'Imagerie Paramétrique, 15 rue de l'Ecole de Médecine, 75006 Paris, France, laurenttaillade2@yahoo.fr), Aymeric Guibal (Laboratoire d'Imagerie Paramétrique, 15 rue de l'Ecole de Médecine, 75006 Paris, France, guibala@hotmail.com), Sebastien Mule (Lab d'Imagerie Fonctionnelle, 91 Boulevard de l'Hôpital, 75634 Paris, France, Sebastien.Mule@imed.jussieu.fr), Eva Comperat (CHU Pitié-Salpétrière, 47-83 Boulevard de l'Hôpital, 75651 Paris, France, eva.comperat@psl.aphp.fr), Yasmina Badachi (Laboratoire d'Imagerie Paramétrique, 15 rue de l'Ecole de Médecine, 75006 Paris, France, yasmina.badachi@gmail.com), Erwan Jouannot (Laboratoire d'Imagerie Paramétrique, 15 rue de l'Ecole de Médecine, 75006 Paris, France, Erwan.Jouannot@sanofi-aventis.com), Olivier Rixe (CHU Pitié-Salpétrière, 47-83 Boulevard de l'Hôpital, 75651 Paris, France, olivier.rixe@psl.aphp.fr), Lori Bridal (Laboratoire d'Imagerie Paramétrique, 15 rue de l'Ecole de Médecine, 75006 Paris, France, bridal @lip.bhdc.jussieu.fr)

Microvascularization modifications should precede tumor size-changes during anti-angiogenic therapy. We applied contrast functional ultrasound imaging (fUSI) to detect changes in Wilms tumors with anti-angiogenic treatment (Bevacizumab). Human Wilms tumor cells was grafted in left kidney of 32 mice. Once tumors had >5mm diameter, mice received : placebo, N=14; Bevacizumab for 21days, N=11; and Bevacizumab for 10days followed by placebo for 11days, N = 7. On days -1, +1, +9, +14 and +21

with respect to treatment start, fUSI was performed (CPS mode, SonoVue). Linear time intensity curves were obtained from regions in kidney cortex and matched-depth of tumor for first bolus passage and 50s following acoustic destruction of contrast. Excised tumor weight decreased with increased treatment duration: 3.7+/-1.8 g (placebo), 2.3+/-1.9 g (Bevacizumab-10days, placebo-11days), 1.4+/-0.7 g (Bevacizumab-21 days) [$p<0.05$]. Area under the bolus-passage curve (AUC) and the plateau intensity of the destruction-reperfusion were greater from D+9 to D+21 [$p<0.04$] in the placebo than Bevacizumab-21day. For the group treated during the first 10 days, fUSI values were comparable to those of the treated group until D+14, then increased to become slightly superior to those of the placebo group by D+21. Noninvasive fUSI demonstrated revascularization after suspension of anti-VEGF therapy.

12:40-1:40 Lunch Break

Invited Paper

1:40

2aBB14. Violent cavitation from optically configured microbubble pairs. Paul A. Campbell (University of Dundee, Ewing 1-6, Carnegie Physics Laboratory, Main Campus, DD1 4HN Dundee, UK, p.a.campbell@dundee.ac.uk)

The mutual interaction between two cavitating microbubbles was investigated using a novel optical trapping arrangement. This approach facilitated development of arbitrary, stable, and initial spatial configurations for two-bubble systems. Critically, exercising optical control over such a binary system can effectively isolate it from resident bubble populations during insonation. This ensures that any early stage dynamical evolution of the system is dominated by the mutual interaction of the two bubbles in view, rather than any extraneous influence arising from 'cross-talk'. In circumstances where the bubbles are located far from a rigid boundary, we observed, using high speed microphotography at framing rates of over 1MHz, that the action of secondary radiation forces leads to mutual bubble attraction often followed by the development of violent microjetting. Microbubble coalescence, with subsequent fragmentation could be observed, and the generation of short-lived anti-bubbles can often occur also. Finally, when cells are present as confluent monolayers on rigid substrates, the statistics for microbubble outcome and dynamics are modified somewhat. I will relate statistics to the parameter space that we interrogated, which included: inter-bubble spacing; relative bubble radii; and individual bubble stand-off parameters.

Contributed Papers

2:00

2aBB15. Nonlinear propagation in microbubble cloud, does it make the distal myocardium appear brighter or darker? Meng-Xing Tang (Imperial College London, Dept. of Bioengineering Engineering, South Kensington, SW7 2AZ London, UK, mengxing.tang@ic.ac.uk), Robert Eckersley (Imperial College London, Imaging Sciences Department, Hammersmith Campus, Du Cane Road, W12 0HS London, UK, r.eckersley@ic.ac.uk)

A number of questions remain in contrast ultrasound imaging regarding the complex interaction between bubbles, US and tissue. E.g. in contrast echocardiography for perfusion imaging, it is sometimes difficult to interpret the images for the myocardium distal to the probe. Due to US nonlinear propagation through the bubble-filled chamber, the distal part can appear artefactually brighter or darker in e.g. Pulse Inversion images. Although tissue at target is likely to increase in brightness in contrast specific images when nonlinear propagation happens, it is less clear for microbubbles. This work tries to gain better understanding of this by investigating how nonlinear propagation of ultrasound pulses can change the appearance of microbubbles and tissues in Pulse Inversion images by altering their acoustic response. A series of specifically designed simulations and experiments were performed. The results show that nonlinear propagation can have a significant impact on the appearance of tissue containing bubbles, and this varies with insonating frequency and pressure. These results suggest the mechanism involves a balance between a reduction in nonlinear bubble scattering after propagation against an increase in the nonlinear tissue scattering. Consequently the deeper tissues containing bubbles may appear brighter or darker depending on the relative contribution of these two effects.

2:20

2aBB16. Frequency dependence of kidney injury induced by contrast-aided diagnostic ultrasound. Douglas Miller (University of Michigan, Department of Radiology, Ann Arbor, MI 48109-0553, USA, douglm@umich.edu), Chunyan Dou (University of Michigan, Department of Radiology, Ann Arbor, MI 48109-0553, USA, chunyand@umich.edu), Roger C. Wiggins (University of Michigan, Department of Internal Medicine (Nephrology), Ann Arbor, MI 48109, USA, rwiggins@med.umich.edu)

Glomerular capillary hemorrhage (GCH) induced in rat kidneys by contrast aided diagnostic ultrasound was examined as a function of ultrasonic frequency. The right kidneys of anesthetized rats mounted in a water bath were exposed to image pulse sequences at 1 s intervals during intravenous infusion of diluted ultrasound contrast agent. Diagnostic ultrasound (DUS) scanners were utilized for exposure at 1.5, 2.5, 3.2, 5.0 and 7.4 MHz. A laboratory exposure system (LES) was used to simulate DUS exposure at 1.0, 1.5, 2.25, 3.5, 5.0 and 7.5 MHz at relatively high peak rarefactional pressure amplitudes (PRPAs). The exposed kidneys were removed and fixed for histology. GCH was measured by counting glomeruli with blood in the urinary (Bowman's) space on histological sections. The percentage of GCH at a fixed exposure frequency showed a rapid rise with PRPA above an apparent threshold. The threshold values were approximately proportional to the ultrasound frequency, with proportionality constants of 0.5 MPa/MHz for DUS and 0.6 MPa/MHz for LES exposures. The increasing thresholds with increasing frequency limited the GCH effect for contrast aided DUS, and no GCH was seen for DUS at 5.0 or 7.4 MHz for the highest available PRPAs.

Session 2aEAa

Engineering Acoustics, Underwater Acoustics, and ECUA: Sonar Transducer Design and Modeling II

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Invited Papers

8:00

2aEAa1. Acoustic array interactions in the time domain. George W. Benthien (511 Savoy Street, San Diego, CA 92106, USA, benthien@cox.net)

In this paper I discuss the computation of acoustic array interactions in the time domain. Acoustic array interactions in the time domain are expressed in terms of mutual impulse response functions. The mutual impulse response between a pair of array elements is the acoustic force on the first element of the pair due to an acceleration impulse of the second with all other elements held rigid. The mutual response functions are calculated in the frequency domain using the boundary element program CHIEF and then Fourier transformed to the time domain. As the convergence in the frequency domain can be very slow, I introduce modified impulse response functions that converge much faster in the frequency domain and are simply related to the desired impulse response functions. This approach reduces the range of frequencies required in the CHIEF computations and gives better accuracy in the initial time period. Transient results are presented for an array of simple transducer elements. The mutual impulse response functions are combined with the transducer element equations to yield a system of differential-integral equations of delay type. A solution procedure for equations of this type is presented.

8:20

2aEAa2. Exact model of Langevin transducers with internal losses. D. D. Ebenezer (Naval Physical and Oceanographic Laboratory, Thrikkakara, 682021 Kochi, India, tsonpol@vsnl.com), P. A. Nishamol (Naval Physical and Oceanographic Laboratory, Thrikkakara, 682021 Kochi, India, dd_ebenezer@yahoo.com)

An exact method is presented to analyze classical Langevin transducers with internal losses. The transducers consist of an axially polarized piezoelectric cylinder sandwiched between two elastic cylinders. All three cylinders are of the same diameter. Exact solutions to the exact equations of motion of the piezoelectric and elastic cylinders and the Gauss electrostatic condition are used. Complex piezoelectric and elastic coefficients are used to model internal losses. For each cylinder, the first set of solutions contains Bessel functions that form a complete set in the radial direction. The second and third sets contain trigonometric functions that form complete sets in the axial direction. They are used to represent fields that are symmetric and anti-symmetric with respect to the plane midway between the ends of the cylinder, respectively. The interface and boundary conditions are satisfied by using the orthogonal properties of the functions. Transducers with identical elastic cylinders at the ends as well as those with a light head mass and a heavy tail mass are analyzed. Numerical results are presented to illustrate the input electrical admittances of transducers. They are compared with those obtained using ATILA - a finite element package for the analysis of sonar transducers.

8:40

2aEAa3. Numerical acoustic modeling code applied to sonar transducers and arrays: review and perspectives. Gerard Vanderborck (Thales Underwater Systems SAS, Acoustic Department, 525 route des Dolines, BP 157, 06903 Sophia Antipolis Cedex, France, gerard.vanderborck@fr.thalesgroup.com)

In this paper a review based on several examples of Thales Underwater Systems (TUS) design and modeling of transducers will be presented: several examples like wide band free flooded ring flextensional and bender transducers, high frequency transducers with thermal analysis modeling included, will be discussed. An approach of the problem of the mutual impedance computation appearing in array modeling will be also presented. Tus used since several decades FEM - BEM codes to design acoustic transducers and arrays. The perspectives in transducer and array modeling will be indicated like the possibility to take into account the non-linearities in the material, the time dependent problem and the use of new kind of piezoelectric material will be also discussed. Specific aspect of single crystal transducers modeling will be presented. The increase of the computing power will also permit to take into account the complex problem of the interaction between transducers in an array (mutual impedances). We will conclude on complex ultra wide band antenna (based on strong transducers interactions) optimization and the associated modeling architecture issues.

9:00

2aEAa4. Finite element analysis simulations of a piezoelectric cymbal actuator using atila software. Thomas Tremper (Micromechatronics Inc., 200 Innovation Blvd, Suite 155, State College, PA 16803, USA, ttremper@mmech.com), Alfredo Vazquez Carazo (Micromechatronics Inc., 200 Innovation Blvd, Suite 155, State College, PA 16803, USA, avc@mmech.com), Kenji Uchino (Micromechatronics Inc., 200 Innovation Blvd, Suite 155, State College, PA 16803, USA, kenjiuchino@mmech.com)

The piezoelectric ultrasonic cymbal actuator is a well known flextensional type actuator. Because of the actuator's geometry, the small movements of the piezoelectric ceramic are amplified by the flextensional structure. Simulation of this device was performed using Atila finite element software, which is specially suited for piezoelectric devices. This study will focus on some of the necessary parameters to accurately simulate cymbal actuators and other piezoelectric actuators. The simulations discussed will be compared to actual experimental data. Based on the experimental results the simulations will be reexamined and modifications will be made to increase the models accuracy. Employing this iterative process will promote increased accuracy in future simulations, which would therefore decrease development time and increase productivity.

Contributed Papers

9:20

2aEAa5. Underwater tensor sensors based on optical fiber bragg gratings. Francois M. Guillot (Georgia Institute of Technology, Mechanical Engineering, 771 Ferst Drive, Atlanta, GA 30332-0405, USA, francois.guillot@me.gatech.edu), David H. Trivett (Georgia Institute of Technology, Mechanical Engineering, 771 Ferst Drive, Atlanta, GA 30332-0405, USA, david.trivett@me.gatech.edu), Peter H. Rogers (Georgia Institute of Technology, Mechanical Engineering, 771 Ferst Drive, Atlanta, GA 30332-0405, USA, peter.rogers@me.gatech.edu)

This paper deals with the development a new type of low-noise underwater tensor sensor aimed at improving the performance and the design of directional arrays. The transducer can be configured either as a particle velocity sensor (dipole) or as fluid shear sensor (quadrupole). The sensing principle of the device relies on the interference signal from two Bragg gratings written on the same fiber, and illuminated by a tunable, narrowband light source. The gratings are a few centimeters apart, and they each reflect a portion of the incident light. The fiber is epoxied to two spacers separated by a small gap situated between the gratings. This assembly is then adhered to two plates connected by a hinge, which is located below the gap. One plate is held rigidly and the tip of the other (free) plate experiences transverse vibrations, when ensonified. These vibrations produce periodic gap length changes, which modulate the interference signal from the two gratings. The modulation is related to the amplitude of the sound wave and is monitored with a photodetector. A noise analysis will be presented and the performance of prototype sensors will be discussed.

9:40

2aEAa6. Beampattern optimization for a conformal projecting array of transducers with baffle effect and mutual coupling among elements. Yuanliang Ma (Institute of Acoustic Engineering, Northwestern Polytechnical University, 710072 Xi'an, China, ylma@nwpu.edu.cn), Zhengyao He (Institute of Acoustic Engineering, Northwestern Polytechnical University, 710072 Xi'an, China, hezhengyao@163.com)

Beampattern optimization for projecting arrays is much more challenging in comparison with receiving arrays, particularly for conformal projecting arrays. It is because the array elements have to work in resonance state for efficient projecting and the variation of radiation impedance has strong effect on the array performance. The radiation impedance for each element is determined by a lot of factors, i.e, the element position, baffle property, mutual coupling and velocity weighting vector of the array. Thus beampattern optimization for a projecting conformal array must take into account all these factors. One thing makes the topic more difficult that the mutual impedance for each element is changeable along with the weighting vector change. In this paper a global solution approach is formulated and a boundary element model combined with transducer equivalent circuits is developed. The driving voltage weighting vector is deduced through an optimization algorithm. Computer simulation together with experiments are conducted for a 14-element conformal array. The results agree well. The multiple beampatterns formed within a wide observation sector exhibit uniform beamwidth and low sidelobes. The source level for each beam is maximized in constraint of the maximum driving voltage of array elements being constant.

10:00

2aEAa7. Advanced Single Crystal Piezoelectric Transducers for Naval Sonar and Medical Ultrasound Applications. Wesley Hackenberger (TRS Technologies, Inc., 2820 East College Avenue, State College, PA 16801, USA, wes@trstechnologies.com), Jun Luo (TRS Technologies, Inc., 2820 East College Avenue, State College, PA 16801, USA, jun@trstechnologies.com), Kevin Snook (TRS Technologies, Inc., 2820 East College Avenue, State College, PA 16801, USA, kevin@trstechnologies.com)

Single crystal piezoelectrics have received much attention due to their very high piezoelectric and electromechanical coupling coefficients. Naval sonar and medical ultrasound transducers, fabricated from this material, exhibit unprecedented increases in bandwidth, source level, and sensitivity compared to ceramic based devices. In this paper we report many new developments in the growth and manufacture of single crystals. These include increased diameter ($> 3''$) crystal boules, improved compositional uniformity, increased thermal stability, and minimized surface damage. Manufacturing improvements are being applied to a broad range of crystal transducer applications. For navy sonar systems, crystals are resulting in very high bandwidths ($> 100\%$) for transducer sizes that are 1/2 the resonating length and 1/4 the weight of ceramic counterparts with equivalent (or often lower) source level. Manufacturing improvements such as high tensile strength, achieved by precision polishing crystal surfaces, are enabling new transducer operational regimes not possible with ceramic. In particular, single crystal 1-3 composite projectors show promise for achieving high source level in a compact, integrated system designs. Single crystals are also enabling new applications in medical ultrasound including very broadband and extremely high frequency transducers. Further improvements are expected in the near future.

10:20-10:40 Break

10:40

2aEAa8. Design of a phase array ultrasonic sensor using vibration decoupled concept. Chia-Yu Lin (National Taiwan University, Rm.433, Institute of Applied Mechanics, No.1, Sec.4, Roosevelt Rd., 10617 Taipei, Taiwan, cylinx@mems.iam.ntu.edu.tw), Chih-Chiang Cheng (National Taiwan University, Rm.433, Institute of Applied Mechanics, No.1, Sec.4, Roosevelt Rd., 10617 Taipei, Taiwan, cccheng@nbm.ntu.edu.tw), Wen-Jong Wu (Department of Engineering Science and Ocean Engineering, National Taiwan University, Rm.433, Institute of Applied Mechanics, No.1, Sec.4, Roosevelt Rd., 10617 Taipei, Taiwan, wjwu@ntu.edu.tw), Chuin-Shan Chen (Department of Civil Engineering, National Taiwan University, Rm.433, Institute of Applied Mechanics, No.1, Sec.4, Roosevelt Rd., 10617 Taipei, Taiwan, dchen@ntu.edu.tw), Jay Shieh (Department of Material Science and Engineering, National Taiwan University, Rm.433, Institute of Applied Mechanics, No.1, Sec.4, Roosevelt Rd., 10617 Taipei, Taiwan, jayshieh@ntu.edu.tw), Chih-Kung Lee (National Taiwan University, Rm.433, Institute of Applied Mechanics, No.1, Sec.4, Roosevelt Rd., 10617 Taipei, Taiwan, cklee@mems.iam.ntu.edu.tw)

Phase array ultrasonic sensors have been used widely to generate higher directional radiating patterns in which piezoelectric units are distributed

sparsely in space. In this study, we present a novel design of a phase array ultrasonic sensor based on the concept of vibration decoupling. Decoupling is achieved by careful design of source aperture, and such design allows piezoelectric units tightly located in the same structure. The phase array sensor is designed herein as a cylinder with a dumbbell shape groove to decouple vibration, and finite element analysis is used to optimize the design. Two piezoelectric discs are adhered on the bottom plate of the sensor whereby desirable wave generation and detection are controlled adaptively. By electrical steering, the sensor thus operates as a dipole mode. Finally, prototypes of the sensor are made and experiments are conducted to verify simulation results.

11:00

2aEAa9. Nonlinear multifrequency transmitter for seafloor characterization. Lucilla Di Marcoberardino (Institut Jean le Rond D'Alembert, UMR 7190, 2 Place de la Gare de Ceinture, 78210 Saint Cyr L'Ecole, France, lucilladm@hotmail.com), Jacques Marchal (Institut Jean le Rond D'Alembert, UMR 7190, 2 Place de la Gare de Ceinture, 78210 Saint Cyr L'Ecole, France, jmarchal@ccr.jussieu.fr), Pierre Cervenka (Institut Jean le Rond D'Alembert, UMR 7190, 2 Place de la Gare de Ceinture, 78210 Saint Cyr L'Ecole, France, cvk@ccr.jussieu.fr)

In the underwater context, it is known that the frequency diversity provides essential information to derive the nature of the seafloor. This presentation deals with a new concept based on a transmitter that generates simultaneously several harmonic frequencies. Our final objective is to assert the feasibility of a multi-frequency tool whose desirable characteristics could be specified for applications such as detection of sunken oil slicks, sediment characterization, or surveys before cable or pipe laying. The acoustic beams are generated through the harmonic components of a shock wave radiated by an antenna driven at a high level. The source is unique in time and space so that the multi-frequency responses are inherently perfectly matched. A numerical model based on a generalized KZK equation has been developed to estimate the saturated fields. Measurements of the first harmonic fields obtained in our outdoor tank facility are compared with simulations.

11:20

2aEAa10. Modeling the acoustic radiation force in piezoelectrically driven micro fluidic chambers using ATILA. Karl Fisher (Lawrence Livermore National Laboratory, 7000 E Avenue, Brentwood, CA 94513, USA, fisher34@llnl.gov)

A procedure is demonstrated to quantitatively evaluate the acoustic radiation forces in micro fluidic particle manipulation chambers. Typical estimates of the acoustic pressure and the acoustic radiation force are based on an analytical solution for a simple 1-dimensional standing wave pattern. The complexities of a typical micro fluidic channel limit the usefulness of this approach. By leveraging finite element approaches, and a generalized equation for the acoustic radiation force, channel designs can be investigated in two- and three-dimensions. Calculations and experimental observations are in good agreement. This work was performed under the auspices of the U.S. Department of Energy by University of California, Lawrence Livermore National Laboratory under Contract W-7405-Eng-48.

11:40

2aEAa11. Equivalent circuit models derived from finite element models using structural dynamics techniques. Julien Bernard (Thales Underwater Systems, 525 route des Dolines BP 157 Valbonne Parc d'Activités de Sophia Antipolis, 06903 Sophia Antipolis Cedex, France, julien.bernard@fr.thalesgroup.com)

Electroacoustic transducers can be divided into an active part, the driver, and a passive part, the structure. The driver ensures electromechanical transduction, while the structure performs various mechanical and acoustical functions, such as support, shock and pressure protection, and impedance transformation. For design purposes, one often needs an equivalent circuit model which gives a relationship between the acoustic characteristics of the overall device and that of its components. Transducer equivalent circuits are usually either physical or modal. Physical equivalent circuits lend themselves to the treatment of a transducer as an assembly, but in general yield frequency dependant parameters. Modal equivalent circuits are more adequate for resonant transducers, but describe transducers as a whole. This works shows how these two types of equivalent circuits can be obtained from a full finite element model by using common structural dynamics techniques: substructuring and modal expansion. It also shows that a third, hybrid type of equivalent circuit can be obtained by using a component mode synthesis technique derived from the Craig-Bampton method. This hybrid equivalent circuit combines the advantages of physical and modal equivalent circuits, enabling to express transducer modal parameters in terms of driver and structure modal parameters.

12:00

2aEAa12. Acoustic field calculation for a compact barrel-stave flextensional transducer array. Zhengyao He (Institute of Acoustic Engineering, Northwestern Polytechnical University, 710072 Xi'an, China, hezhengyao@163.com), Chao Sun (Institute of Acoustic Engineering, Northwestern Polytechnical University, 710072 Xi'an, China, csun@nwpu.edu.cn)

The boundary element method together with the finite element method is used to calculate the radiated acoustic field of a compact array of barrel-stave flextensional transducers. At first, the surface vibration displacement distribution of one barrel-stave transducer is obtained by the finite element method using the commercial software ANSYS. The calculation results are then imported into the boundary element calculation software SYSNOISE. At the frequency of 1400Hz, the radiated acoustic field and radiation impedance are calculated by the boundary element method for a planar array which is composed of three identical barrel-stave flextensional transducers uniformly distributed on a circle with spacing much less than half wavelength. The calculation results show that the mutual interactions among elements are significant for the compact array. The mutual radiation resistance between two transducers is close to the self-radiation resistance of the transducers. And the transmitting source level of the 3-element array is 8.7dB higher than that of one transducer if the surface vibration velocities of the transducers in the array are the same as that of one transducer. The proposed technique can be used to predict the performance of a transmitting transducer array at the stage of preliminary design.

12:20

2aEAa13. Finite Element Modeling of 2-D Transducer Arrays. Hind Mestouri (ISEN Brest (Institut Supérieur de l'Electronique et du Numérique), 20, rue Cuirassé Bretagne, C.S. 42807, 29228 Brest, France, hind.mestouri@isen.fr), Alain Loussert (ISEN Brest (Institut Supérieur de l'Electronique et du Numérique), 20, rue Cuirassé Bretagne, C.S. 42807, 29228 Brest, France, alain.loussert@isen.fr), Gilles Keryer (ISEN Brest (Institut Supérieur de l'Electronique et du Numérique), 20, rue Cuirassé Bretagne, C.S. 42807, 29228 Brest, France, gilles.keryer@isen.fr)

Active sonar detects objects underwater by sending out sound waves in pulses, scientists could measure the time it takes these pulses to travel through the water, reflect off of an object, and return to the ship. There are a number of factors limiting the performance of low frequency transducer arrays for active sonar systems including the crosstalk and housing interactions which affects the directivity pattern and the sensitivity. Many important issues in transducer arrays design, such as crosstalk, cannot be accurately studied using analytic method due to the complexity of the partial differential equations involved. Finite element method is the only appropriate way to gain more detailed information. In this paper, a 2-D finite element model is constructed to analyze the crosstalk and structure interactions, using ATILA code and GiD graphical interface, of transducer arrays, consists of six active piezoceramic bars, and developed for shallow water use. The directivity pattern strongly depends on the transducer size versus wavelength, each transducer elements interacts directly with its neighbour and indirectly with the other elements, and the radiated acoustic power may considerably vary from one element to another. It is shown that crosstalk and housing interactions can be reduced through the use of different inter-element materials and housing characteristics.

Session 2aEAb

Engineering Acoustics: Acoustic Evaluation I

David Brown, Chair
BTech Acoustics, LLC and Universtity of Massachusetts Dartmouth, ATMC, Electro-Acoustics Research Laboratory and ECE Dept., 151 Martine St, Fall River, MA 02723, USA

Contributed Papers

8:00

2aEAb1. Measurement of Sound Velocity in Water Using Optical Probe and Acoustical Holography. Takeshi Ohbuchi (University of Tsukuba, 1-1-1 Tennoudai, 305-8573 Tsukuba, Japan, ohbuchi@aclab.esys.tsukuba.ac.jp), Koichi Mizutani (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, mizutani@esys.tsukuba.ac.jp), Naoto Wakatsuki (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, wakatuki@iit.tsukuba.ac.jp), Hiroyuki Masuyama (Toba National College of Maritime Technology, 1-1 Ikegami-cho, 517-8501 Toba, Japan, masuyama@toba-cmt.ac.jp)

We propose a method for a determination of a three-dimensional sound velocity using optical computerized tomography (O-CT) and near-field acoustical holography (NAH) for measuring a phase object. Ultrasonic waves affect a phase of the light passing through radiated sound fields. The zeroth order diffraction light including sound pressure information is acquired by an avalanche photodiode. Eighteen projections are acquired by rotational scanning, and each projection along the optical axis is obtained by single linear scanning and electronically quadrature-detected as the complex amplitude. The complex sound fields are reconstructed by O-CT in a region of 40×40 mm^2. Then the sound field on another plane is propagated using NAH from the acquired sound fields, and the sound field on the same plane is reconstructed by O-CT. Comparing the phase of propagated and reconstructed sound fields in wavenumber domain, we can obtain the sound velocity in three dimensional space in a region of $30 \times 30 \times 10$ mm^3. The experimental results are in agreement with the reference value measured by another mean.

8:20

2aEAb2. Influence of a load on the nonlinear behavior of a piezoelectric rod under high sinusoidal voltages. Denis Parenthoine (Lussi, Université F Rabelais de Tours, 40, rue de la Chocolaterie, 41000 Blois, France, parenthoine_denis@yahoo.fr), Lionel Haumesser (Lussi, Université F Rabelais de Tours, 40, rue de la Chocolaterie, 41000 Blois, France, lionel.haumesser@univ-tours.fr), François Vander Meulen (Lussi, Université F Rabelais de Tours, 40, rue de la Chocolaterie, 41000 Blois, France, vandermeulen@univ-tours.fr), Louis-Pascal Tran-Hhuu-Hue (Lussi, Université F Rabelais de Tours, 40, rue de la Chocolaterie, 41000 Blois, France, tran@univ-tours.fr)

At high levels of excitation, appear, in piezoelectric devices, nonlinear phenomena which can adversely affect the quality of the applications to which they are devoted. It is, in particular, true in non-destructive evaluation, where the generation of harmonics in the ultrasonic tranducer can make inacurrate measurements in materials. In a previous work, third-order constants in a piezoceramic rod of PZT-21 under high sinusoidal electric fields have been evaluted from the analysis of second harmonic generation in the mechanical displacement. All these measurements have been performed under condition of complete free stress. In order to allow a better understanding of the nonlinear behavior of the transducer, various loads, are, now, applied at one end of the piezoceramic rod. Velocity masurements performed by a laser probe at the free end of the active element provides then informations about the nonlinear behavior of the system. Influence of load on the linear and nonlinear behavior of the piezoelectric rod is then studied in the case of an aluminium bar, and in the case of more nonlinear materials. Results are compared both experimentally and theoritically in loaded and unloaded configurations.

8:40

2aEAb3. Uncertainty of Acoustical Material Characteristic Measurements. Samir N. Gerges (Federal University of Santa Catarina (UFSC), Campus Universitario - Trindade, 88040-900 Florianopolis, Brazil, samir@emc.ufsc.br), Peter K. Giesbrecht (Federal University of Santa Catarina (UFSC), Campus Universitario - Trindade, 88040-900 Florianopolis, Brazil, samir@emc.ufsc.br)

The experimental determination of the acoustic properties of sound absorption materials depends on numerous factors. The accuracy level in the determination of each property depends, for example, on the measurement apparatus and the type of material. The objective of this work is to present a study about the main sources of uncertainty in the measurement of properties such as sound absorption coefficient, acoustic impedance, flow resistivity and porosity. The uncertainties of the measurement apparatus of each property are quantified. The uncertainty of the materials heterogeneity is also assessed and its individual contribution highlighted. Important aspects of the Standards ASTM C 522 and ISO 10534-2, which present, respectively, recommendations and a procedure for flow resistivity and sound absorption measurements, are discussed. The methodology to calculate the uncertainty of each property is discussed with emphasis on the relevancy of each uncertainty source.

9:00

2aEAb4. Design and implementation of a three-dimensional seven microphone vector intensity probe with low and high frequency compensations. Khalid Miah (University of Texas, P.O. Box 7155, Austin, TX 78713, USA, miah@mail.utexas.edu), Elmer Hixson (Univ. of Texas, ECE Dept., 1 University Station 0803, Austin, TX 78712-0240, USA, ehixson@mail.utexas.edu)

A seven microphone vector intensity probe has been designed and implemented in this report with low and high frequency compensations. Measurements from plane wave tube and anechoic chamber are used for overall system calibration with corrected gain and phase mismatch errors. Measured and calculated intensity level from both plane wave tube and anechoic room settings showed close agreement with each other in comparison to separate low and high frequency components of the array. Particle velocity vectors and pressures determined for calculating intensity were also used to calculate sound power and total energy density components of various sources. This microphone array vector intensity probe is then used to explore various sound source identification and localization problems.

9:20

2aEAb5. Application of contrast optimisation autofocus to flexible ultrasonic arrays for non-destructive testing. Alan J. Hunter (University of Bristol, Queen's Building, University Walk, BS8 1SY Bristol, UK, a.hunter@bristol.ac.uk), Bruce W. Drinkwater (University of Bristol, Queen's Building, University Walk, BS8 1SY Bristol, UK,

b.drinkwater@bristol.ac.uk),Paul D. Wilcox(University of Bristol, Queen's Building, University Walk, BS8 1SY Bristol, UK, p.wilcox@bristol.ac.uk)

Flexible ultrasonic arrays are used for imaging within objects with complicated geometries for non-destructive testing and evaluation (NDT/NDE), e.g., thick-walled pipes, weld caps, etc. We consider the application of autofocus techniques (routinely used in synthetic aperture sonar (SAS)) to this engineering problem. A flexible ultrasonic array is functionally similar to a wide-beam, stripmap SAS with a single transmitter and receiver. Unfortunately, there are few autofocus algorithms available for this configuration. Popular algorithms, such as echo/image correlation, PGA, DPCA, etc., are better suited for use with a narrow-beam or multiple-receiver SAS. However, contrast optimisation is a more general technique that is well suited to the single transmitter/receiver geometry. In this presentation, we describe our implementation of contrast optimisation autofocus and show experimental results using a flexible array prototype.

9:40

2aEAb6. Inspection of complex components with flexible phased array transducer. Olivier Casula (CEA/LIST, bat 611, 91191 Gif sur Yvette, France, olivier.casula@cea.fr), Gwenael Toullelan (CEA/LIST, bat 611, 91191 Gif sur Yvette, France, gwenael.toullelan@cea.fr), Philippe Dumas (ZA, rue des Savourots, 70190 Voray-sur-l'Ognon, France, philippe .dumas@imasonic.com)

Non destructive testing techniques using ultrasonic methods are often carried out in contact. But, the inspection performances are limited to regular surfaces. Thus, surface irregularities lead to thickness variations of the coupling layer that result in beam distortions and losses of sensitivity. In the context, CEA/LIST has designed flexible phased-array techniques for compensating the surface irregularities and fitting the surface. The independent piezoelectric elements composing the radiating surface are mechanically assembled to build an articulated structure. An embedded profilometer measures the local surface distortion allowing to compute the optimized delay laws and to master the characteristics of the focus beam. Those delay laws computed by the UT-acquisition system are applied in real-time to the piezoelectric elements. To evaluate inspection method capabilities, CEA/LIST develops a simulation software for non destructive testing, CIVA, able to simulate realistic configurations in particular with complex 2D and 3D applications. Matrix flexible phased-array probes have been designed and manufactured in collaboration with IMASONIC. This paper sums up examples of inspections in complex geometries where these flexible probes have been successfully used. Moreover, the data are reconstructed thanks to CIVA tools and allow to locate and size the flaw in the part.

10:00-10:20 Break

10:20

2aEAb7. Characterization of laminated glasses by means of an inversion method using Finite Elements. Jorge Frances Monllor (DFISTS. Univ. de Alicante, Carretera de Sant Vicent del Raspeig s/n, 03690 San Vicente del Raspeig, Spain, jfmonllor@ua.es), Jaime Ramis Soriano (DFISTS. Univ. de Alicante, Carretera de Sant Vicent del Raspeig s/n, 03690 San Vicente del Raspeig, Spain, jramis@ua.es), Jesús Alba Fernandez (Escola Politècnica Superior de Gandia, Universitat Politècnica de València, Crtra Natzaret-Oliva s/n, 46730 Gandia, Spain, jesalba@fis.upv .es), Enrique E. Segovia Eulogio (Depto Ing. Construcción. Univ. de Alicante, Carretera de Sant Vicent del Raspeig s/n, 03690 San Vicente del Raspeig, Spain, Enrique.Gonzalo@ua.es), Jenaro Vera Guarinos (DFISTS. Univ. de Alicante, Carretera de Sant Vicent del Raspeig s/n, 03690 San Vicente del Raspeig, Spain, jenaro@dfists.ua.es)

The experimental determination of elastic modules and loss factor in laminated glass components is of great interest in building acoustics since at present there are a lot of building systems using these types of devices. This makes it necessary to predict the transmission losses in a partition. Even though described in regulation, the process is not exempt from serious difficulties. In this work we present a method in order to obtain the parameters mentioned above for a laminated glass composed of a sequence of isotropic layers by solving the model-based inverse problem for frequency admittance experimentally obtained. The parameter that best describes the mechanical constants of material of the layers is obtained by minimizing the discrepancy between the real numerically and numerically predicted waveform. This is done using an iterative optimization. A sensitivity study of the parameters uncertainty is performed in order to establish the feasibility of this technique.

10:40

2aEAb8. Detection of Shallow Underground Buried Object Using Air Vibration Probe. Yuji Sato (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, yuji@aclab.esys.tsukuba.ac.jp), Tomohiro Okamura (Univ. of Tsukuba, 1-1-1 Tennodai, 305-8573 Tsukuba, Japan, mizutani@esys.tsukuba.ac.jp), Koichi Mizutani (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, mizutani@esys.tsukuba.ac.jp), Naoto Wakatsuki (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, wakatuki@iit.tsukuba.ac.jp)

An air vibration probe is a device to measure an acoustic impedance using acoustic delay line oscillation. The frequency is changed when the acoustic impedance is changed. If some objects are varied, the frequency is expected to be changed in the case of the shallow underground detection. The advantage of this device is that it can detect the underground object easily without any contacts or destructions. Two experiments were practiced to study the relation among the frequency, the space between the probe and the ground, and the depth of object. The space was changed with the buried objects in various depths. The frequency became higher when the space became wider. In addition, the frequency was changed more obviously when the space was narrower. The probe scanned the buried objects in the fixed space. The objects were buried in various depths. The frequency became high when the object was buried shallower. In conclusion, the air vibration probe can detect the buried object because the frequency became high when something was buried.

11:00

2aEAb9. Ultrasonic Airflow Meter in Greenhouse Using Acoustic Reflection against Wall. Ikumi Saito (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, saitou@aclab.esys.tsukuba.ac.jp), Naoto Wakatsuki (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, wakatuki@iit.tsukuba.ac.jp), Koichi Mizutani (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, mizutani@esys.tsukuba.ac.jp), Limi Okushima (National Institute for Rural Engineering, 2-1-6 Kannondai, 305-8609 Tsukuba, Japan, limi@affrc.go.jp)

In this paper, we described an ultrasonic airflow meter in a greenhouse using an acoustic reflection against a wall. The ultrasonic airflow meter is available for measuring the spatial mean wind velocity and direction, which consists of two sound probes and the wall of the greenhouse. Use of sound probes has advantage that the airflow accumulated along sound paths from a loudspeaker to a microphone is obtained in contrast to point measurements. The wind velocity and direction are calculated from time of flights (TOFs) of direct and reflected signals. The wind velocity and direction were measured in a greenhouse of 7.2 m $\times$ 29.0 m under factitious winds generated from two large electric fans located at the one side of the greenhouse. In addition, we detected air convection generated from two pairs of fans. These were measured every 20 seconds for 120 minutes. Regarding the measurement; by the proposed airflow meter and a reference; by a conventional one, their mean wind velocities were 0.12 and 0.15 (m/s), respectively. The presented airflow meter using the acoustic reflection against the wall is convenient to measure the wind velocity and direction in large-size facilities such as the greenhouse.

11:20

2aEAb10. An alternative and industrial method using low frequency ultrasound enabling to measure quickly tortuosity and viscous characteristic length. François Fohr (Centre de Transfert de Technologie du Mans, 20, rue Thalès de Milet, 72000 Le Mans, France, ffohr@cttm-lemans.com), Damien Parmentier (Centre de Transfert de Technologie du Mans, 20, rue Thalès de Milet, 72000 Le Mans, France,

dparmentier@cttm-lemans.com),Bernard R. Castagnede(Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, bernard.castagnede@univ-lemans.fr), Michel Henry (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, michel.henry@univ-lemans.fr)

In the past, several different methods have been proposed in order to determine tortuosity and characteristic lengths, using various physical approaches. For instance, the BET (from S. Brunauer, P. Emmett and E. Teller) method used the adsorption of argon molecules within the surface of the pores to measure thermal characteristic length (or the so-called specific surface). Ultrasonic methods, during the two last decades, were efficiently implemented to measure these parameters, using large frequency bandwidth, different gazes including helium, and varying static pressures. The most standard and well-known method relies on phase spectrum measurements performed over short bursts, by plotting the inverse of the squared velocity as a function of the inverse of the square root of the angular frequency. Unfortunately, this method is quite sensitive on noise during the unwrapping procedure. We here propose a much simplier and efficient method, which is very robust with noisy signals, working at one single frequency, in most cases around 40 kHz. Measurements have been done onto a specific bench developped at CTTM, with some calibrated glass beads, enabling to recover the expected values of the physicalparameters. This method is particularly well suited for industrial and ″on line″ applications.

11:40

2aEAb11. A new impedance tube for large frequency band measurement of absorbing materials. Jean Christophe Le Roux (CTTM, 20, rue Thales de Milet, 72000 Le Mans, France, jcleroux@cttm-lemans.com), Jean-Pierre Dalmont (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, Jean-Pierre.Dalmont@univ-lemans.fr), Bruno Gazengel (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, bruno.gazengel@univ-lemans.fr)

The standard two microphones technique does not allow the measurement of absorbing materials characteristics at low frequency. Moreover, to cover a range from 100 to 6000 Hz two experiments have to be done with two different sample diameters. By using a sensor with a known volume velocity source developed by the LAUM together with the CTTM, it is demonstrated that the impedance can be obtained from 10 to 6000 Hz by performing only one measurement with a single material sample. Results showing the behaviour of some materials at low frequency are presented. On the other hand a comparison is done with classical Kundt tube results.

12:00

2aEAb12. An experimental study of Sound Transmission Loss (STL) measurement techniques using an impedance tube. Behrooz Yousefzadeh (Univ. of Tehran, School of Mechanical Engineering, 1439957131 Tehran, Iran, behrooz.j.y@gmail.com), Mohammad Mahjoob (Univ. of Tehran, School of Mechanical Engineering, 1439957131 Tehran, Iran, mmahjoob@ut.ac.ir), Nader Mohammadi (Univ. of Tehran, School of Mechanical Engineering, 1439957131 Tehran, Iran, nmohamady@ut.ac.ir), Ali Shahsavari (Univ. of Tehran, School of Mechanical Engineering, 1439957131 Tehran, Iran, shervin.shahsavari@gmail.com)

A comparison between the two sound transmission loss (STL) measurement techniques using an impedance tube (i.e. two-load method and anechoic termination method) is presented. A modified B&K type 4206 impedance tube has been designed and built. STL tests have been carried out for three homogeneous and isotropic materials with disk-type test samples of identical diameters and different thicknesses. In addition, the results have been compared with those of the classical and more reliable method of two-room. For both methods, the effect of downstream (tube termination) boundary conditions have been completely studied. The two-load method yields results which matches with two-room measurements, especially when the two boundary conditions are considerably different. The anechoic termination method, on the other hand, is significantly dependant on the termination boundary conditions.

TUESDAY MORNING, 1 JULY 2008 AMPHI MAILLOT, 8:00 TO 10:40 A.M.

Session 2aMUa

Musical Acoustics and Physical Acoustics: Brass Instrument Acoustics I

Thomas Moore, Cochair
Rollins College, Department of Physics, Winter Park, FL 32789, USA

Joël Gilbert, Cochair
Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, Le Mans, 72085, France

Invited Paper

8:00

2aMUa1. Does mass matter? Examining a concrete didjeridu. Noam Amir (Tel Aviv University, Dept. of Communications Disorders, Sheba Medical Center, 52621 Tel Hashomer, Israel, noama@post.tau.ac.il)

The influence of wall material on the acoustics of musical wind instruments has been debated widely. While this has been examined for some western instruments, it has been barely touched upon regarding the Australian didjeridu. This is very interesting, considering that didjeridus vary enormously in materials and dimensions. Indeed, musicians and manufacturers alike often have very definite opinions concerning the influence of material type and thickness on instrument quality. As a first step towards examining this issue, we conducted a blind test involving three cylindrical didjeridus of identical internal dimensions: all three shared a basic structure of identical plastic tubing. However, a cement casing was cast on the exterior of one of these, bringing its weight to 17 kilograms. The three

instruments were fixed to a stationary wooden frame, then covered so that only the mouthpieces were visible. 32 players of varying expertise were allowed to play each instrument for as long as they wished, and then asked to judge which instrument was different from the other two. Results indicate that the responses were random, regardless of player level. Further research will be necessary to show whether this holds for noncylindrical bores and different internal finishings.

Contributed Paper

8:20

2aMUa2. Relationships among subjective assessments and measured physical properties in the didjeridu. John Smith (University of New South Wales, Music Acoustics, School of Physics, NSW 2052 Sydney, Australia, john.smith@unsw.edu.au), Guillaume Rey (University of New South Wales, Music Acoustics, School of Physics, NSW 2052 Sydney, Australia, guimms@free.fr), Joe Wolfe (University of New South Wales, Music Acoustics, School of Physics, NSW 2052 Sydney, Australia, J.Wolfe@unsw.edu.au)

Traditional didjeridus are unusual and ancient lip-valve instruments with an irregular bore that is largely constructed by termites eating the interiors of small eucalypt trees. This produces instruments with a broad range of bore geometries with many details not immediately apparent to a player. They are therefore well-suited for examining relationships between subjective assessments by players and their measured physical properties. In this study, seven experienced players assessed 38 didjeridus that spanned a wide range of quality, pitch and geometry. A control group of 11 plastic cylindrical pipes were also studied. Eight subjective parameters (backpressure, clarity, resonance, loudness, overtones, vocals, speed, and overall quality) were assessed and the ranking of each instrument correlated with measurements of their geometry and acoustic input impedance spectrum. A strong result of these experiments was the finding that the ranked quality of a didjeridu correlated negatively with the magnitude of its acoustic input impedance, particularly in the frequency range from 1 to 2 kHz. Maxima in the impedance of the player's vocal tract would have a greater effect on instruments with a low impedance, thus favouring the production of the varying spectral peaks or formants in the sound envelope that characterise this instrument.

Invited Papers

8:40

2aMUa3. A psychoacoustical investigation into the effect of wall material on the sound produced by lip-reed instruments. James W. Whitehouse (Acoustics Research Group, DDEM, MCT Faculty, Open University, Walton Hall, MK7 6AA Milton Keynes, UK, j.w.whitehouse@open.ac.uk), David B. Sharp (Acoustics Research Group, DDEM, MCT Faculty, Open University, Walton Hall, MK7 6AA Milton Keynes, UK, d.sharp@open.ac.uk)

In order to investigate whether a lip-reed instrument's material of manufacture plays a significant role in determining the timbre of the notes it produces, tests have been conducted on five post horns. These post horns have identical geometry but are manufactured from different copper alloys. Using a laser Doppler vibrometer, the structural resonances of each instrument have been established and the wall vibrations induced in each instrument when artificially blown have been measured. In this paper, these measurements are compared with findings from a series of blindfold playing tests carried out using professional musicians and with listening tests comprising notes produced by the different instruments.

9:00

2aMUa4. More experimental evidence favouring the hypothesis of significant wall vibration influence on radiated horn sound. Wilfried Kausel (Inst. f. Wiener Klangstil, Univ. f. Music, Anton von Webernplatz 1, A-1030 Vienna, Austria, kausel@mdw.ac.at), Alexander Mayer (Inst. f. Wiener Klangstil, Univ. f. Music, Anton von Webernplatz 1, A-1030 Vienna, Austria, mayer@mdw.ac.at)

The question whether wall vibrations of wind instruments do or can affect the radiated sound has not been finally answered yet. Instrument makers and musicians make a strong claim that wall thickness, material and conditioning are crucial factors for sound quality and response of wind instruments, while acousticians rather tend to question that claim. Recent experiments on horns are presented favoring the hypothesis that wall vibrations do matter. Although horn sound in general appears to change quite significantly when wall vibrations, particularly those of the bell, are dampened during artificial playing, it becomes more and more evident that more than one single mechanism has to be considered to explain the whole phenomenon. Radiation of the bell as multi-pole will add directivity and strengthen higher harmonics while absorption by the wall will rather weaken them. Dynamic interaction between air column and oscillating bore profile theoretically depends on broken symmetry. Feedback of structural resonances to the oscillating lips can be avoided by decoupling the mouthpiece from the instrument. Experiments and statistical evaluation have been laid out in order to exclude or identify certain mechanisms as well as possible artifacts like changes of mouthpiece position, bore or acoustical environment caused by damping forces.

9:20

2aMUa5. Design and manufacturing of an artificial marine conch by bore optimisation. Jef Petiot (Ecole Centrale Nantes - IRCCyN, 1 rue de la Noe, BP 92101, 44321 Nantes, France, jean-francois.petiot@irccyn.ec-nantes.fr), Francois Tavard (Ecole Centrale Nantes - IRCCyN, 1 rue de la Noe, BP 92101, 44321 Nantes, France, francois.tavard@eleves.ec-nantes.fr)

The marine conch is a traditional instrument of the brass family. The resonator is made of the inner shape of the shell, with which the lips of the player interact by the way of a hole pierced in the extremity of the shell. Several notes can be played with this instrument. Unfortunately, the marine conch has become very rare and expensive. In order to build an artificial conch (by injection moulding), we studied the acoustics property of a natural conch. This paper is dedicated to the description of the measurements we made on the conch, and to the presentation of the method used to design the bore. Firstly, the input impedance of the natural conch was measured. Secondly, the size of the bore of the natural conch was assessed by the way of pictures and image processing tools. An initial bore of the artificial

conch was designed with a CAD software, by taking into account manufacturing constraints. We developed next an optimisation procedure to improve the harmonicity of the artificial conch. Finally, an artificial conch was next manufactured by rapid prototyping. As a result, we noticed that the artificial conch manufactured can clearly be used as an interesting musical instrument.

Contributed Paper

9:40

2aMUa6. A design strategy for brass instruments. Robert W. Pyle (S. E. Shires Co., 11 Holworthy Place, Cambridge, MA 02138, USA, rpyle @post.harvard.edu)

This paper describes the processes being used to design a C trumpet. Since ″one size fits all″ is not a good approach for high-quality brass instruments, one of the goals is to create a basic design that, with minor variations, can be tailored to meet the needs of a variety of players. There are other goals, secondary to playing quality but nonetheless desirable. These include appearance and mechanical reliability (″fit and finish″), ease of manufacture (consistent quality at minimal possible cost), and ease of repair (accidents will happen, alas!). Existing instruments provide input data to the design process by way of physical dimensions, acoustic input impedance, and players' judgements. Computer modeling helps to predict the effect on intonation of changes of bore contour. Feedback from top professional players is essential to refine the design, especially the ″black magic″ aspects like brace placement and even the type of solder used to assemble the instrument.

Invited Paper

10:00

2aMUa7. Characterisation of brass musical instrument designs using the brassiness parameter. Arnold Myers (University of Edinburgh, Reid Concert Hall, Bristo Square, EH8 9AG Edinburgh, UK, am@ed.ac.uk)

Recent work has established a ″brassiness″ parameter as a measure of the support given over the sounding length of a brass instrument to non-linear propagation in sound production. This parameter is a function of the geometry (bore profile) of an instrument, and provides a useful means of relating the timbral characters to the designs of the various kinds of brass instrument. The geometries of over 1000 brass instruments belonging to museums worldwide and to individual musicians have been measured to determine the values of their brassiness parameter. Comparisons of these enable not only a more precise taxonomy but also provide a tool for studying the evolution of instrument design. This paper explores both the taxonomic analysis of the whole brasswind field and also gives examples of the use of brassiness in historical research.

Contributed Paper

10:20

2aMUa8. Influence of acoustic waveguides lengths on self-sustained oscillations: Theoretical prediction and experimental validation. Nicolas Ruty (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, nicolas.ruty@gipsa-lab.inpg.fr), Xavier Pelorson (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, pelorson@icp.inpg.fr), Annemie Van Hirtum (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, annemie.vanhirtum@gipsa-lab.inpg.fr)

Human vocal folds and lips of brass instruments players produce self-sustained oscillations due to the interaction between airflow, acoustic waveguides and deformable tissues. This interaction is commonly modelled as a distributed one or two mass-spring system coupled with a simple airflow and acoustic description. This study focuses on the influence of the acoustic waveguide length on the resulting self-sustained oscillation characteristics, i.e. the minimum pressure required to sustain oscillations, the oscillation frequency. Both fixed and varying waveguide lengths are considered. Theoretical predictions with the simplified interaction model are compared to experimental data obtained with a deformable in-vitro replica suitable to produce self-sustained oscillations in presence of an upstream (12, 24 or 32cm) and downstream (varying from 0 up to 235cm) acoustic waveguide. The current study shows the strong influence on the minimum pressure regardless the waveguide length. In addition the waveguide length is shown to impose the resonance frequency for waveguide length superior to 40cm. A rapid change in waveguide length introduces bifurcations between different oscillation regimes.

Session 2aMUb

Musical Acoustics: Interaction Between Instrument and Instrumentalist I

Gary Scavone, Cochair
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Xavier Boutillon, Cochair
Laboratoire de Mécanique des Solides, Ecole Polytechnique, Palaiseau Cedex, 91128, France

Invited Papers

11:00

2aMUb1. Motor control in drumming: Influence of movement pattern on contact force and sound characteristics. Sofia Dahl (Institute of Music Physiology and Musicians' Medicine, Hanover University of Music and Drama, Hohenzollernstr. 47, 30161 Hannover, Germany, dr.sofia.dahl@gmail.com), Eckart Altenmüller (Institute of Music Physiology and Musicians' Medicine, Hanover University of Music and Drama, Hohenzollernstr. 47, 30161 Hannover, Germany, altenmueller@hmt-hannover.de)

Whereas wind instrumentalists and string players have a continuous control of the acoustic sound parameters during playing, a percussionists' direct contact with the instrument is limited to a few milliseconds. The player has no possibilities to adjust grip or dampening during the actual contact. Whatever timbre and sound level the player is aiming for therefore has to be integrated in the entire striking gesture. How can the player control the complex interaction between drumstick and drumhead? In order to investigate how the players' grip and striking gestures influence the sound characteristics of drum strokes we recorded movements, audio, contact time and contact force during drumming. Different instructions were given with the intention to influence how the player's grip controlling the drumstick. ″Normal″ strokes were allowed to freely rebound from the drumhead. For ″controlled″ strokes the player was asked to control the ending position of the drumstick, stopping it as close as possible to the drumhead after the stroke. Preliminary analysis showed that the instructions influenced contact force, contact time, and perceptual ratings of the strokes. Further results and implications will be discussed.

11:20

2aMUb2. Two dimensional finger-string interaction in the concert harp. Jean-Loic Le Carrou (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, jean-loic.le_carrou@univ-lemans.fr), Elio Wahlen (LAUM, CNRS, Université du Maine, Lab. d'Acoustique Université du Maine, UMR CNRS 6613, 72085 Le Mans Cedex 9, France, elio@elio.de), Emmanuel Brasseur (LAUM, CNRS, Université du Maine, Lab. d'Acoustique Université du Maine, UMR CNRS 6613, 72085 Le Mans Cedex 9, France, emmanuel.brasseur@univ-lemans.fr), Joël Gilbert (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, joel.gilbert@univ-lemans.fr)

The sound of the concert harp partly depends on the way the string is plucked. The vibrating string is brought into a state of initial conditions by the finger-string interaction and then oscillates according to two transverse planes. In order to understand the plucking action of the concert harp, a one-dimensional model of this interaction has been developed in a previous paper [Le Carrou et al, Proc. ISMA (2007)]. The parameters of this model were deduced from measurements of the string's and finger's trajectories. The aim of the present paper is to extend this model to a more realistic one, including a two-transverse trajectory for each one of the interaction's elements. To do so, a special experiment with a high-speed camera, which films the interaction, is set up. Specific image processing, based on edges detection, helps to automatically track both objects' positions. The results show that the finger-string interaction takes place in two planes and permits us to obtain the parameters of the two-dimensional model for the finger-string interaction.

11:40

2aMUb3. The modern violin bow in off-string action. Knut Guettler (Norwegian Academy of Music, P.O.Box 5190 Majorstuen, 0302 Oslo, Norway, knut.guettler@nmh.no)

The convex camber of the modern Tourte-model bow permits off-string stroke techniques to be performed with resulting clean and crisp attacks. This paper discusses how different bow-, string-, and bowing parameters must combine in order for this to be achieved. Of particular interest are the geometrical changes imposed on the bow stick during bouncing strokes. A good spiccato bow is characterized by a stick that feels ″lively″, even when the bouncing is low or the hairs are not leaving the string at all. With lesser-quality bows the player has to work harder (on the lower-pitched double bass even sometimes synchronously ″shake″ the bow stick in the string-length plane) in order to induce sufficient stick oscillation. Included in this study are measurements of finger action on bow sticks of different quality.

Contributed Paper

12:00

2aMUb4. Observations on bow changes in violin performance. Matthias Demoucron (IRCAM, 1 Place Igor Stravinsky, 75004 Paris, France, demoucron@ircam.fr), Anders Askenfelt (Dept. of Speech, Music and Hearing, Royal Institute of Technology (KTH), Lindstedtsvägen 24, SE-100 44 Stockholm, Sweden, andersa@speech.kth.se), René E. Causse (IRCAM, 1 Place Igor Stravinsky, 75004 Paris, France, causse @ircam.fr)

Players of bowed instruments commonly separate notes by changing the direction of the bow motion (″bow changes″). The separation can be made more or less pronounced (detaché - portato - martellato). In contrast, long notes, requiring more than a full bow stroke, are played by making the bow changes as inaudible as possible. Acceptable bow changes require accurate control and coordination of a set of bowing parameters, in particular bow speed, bow force and bow-bridge distance. Long practice is required before optimal control is achieved. Detailed descriptions of basic bowing gestures such as bow changes are of great interest for various fields of violin-related studies, including realistic control of synthesis algorithms. We present recent measurements that provide an accurate description of the evolution of bow speed, acceleration and bow force during bow changes at the tip and the frog, respectively. Using these data for controlling a synthesis algorithm, we will discuss how modifications of the parameters influence the bow-string interaction and resulting string vibrations. The simulations are used to model efficient implementations of bow changes.

Invited Paper

12:20

2aMUb5. Extraction of lumped clarinet reed model parameters from numerically synthesised sound. Vasileios Chatziioannou (Sonic Arts Research Centre, Queen's University Belfast, BT7 1NN Belfast, UK, vchatziioannou01@qub.ac.uk), Maarten Van Walstijn (Sonic Arts Research Centre, Queen's University Belfast, BT7 1NN Belfast, UK, m.vanwalstijn@qub.ac.uk)

Fluid dynamical analysis and time-domain modelling of a single reed-mouthpiece-lip system can be used to inform the formulation of a lumped model of the woodwind excitation mechanism. Coupling this lumped model to a model of the instrument bore enables computationally efficient generation of sustained oscillations, using a small number of physical parameters that define the instrument and the way the player controls them. As such, the embouchure of the player as well as the geometry of the system is taken into account. In this paper, an attempt is carried out to use the numerically generated sound as an input to an inversion algorithm for the reed-mouthpiece-lip system. Assuming that the reed motion is proportional to the pressure difference across it, a relationship can be established between the pressure and the total flow inside the mouthpiece that allows a first estimation of the physical parameters using standard optimisation techniques. Currently we are undertaking efforts to apply the inversion to data measured under real playing conditions, i.e. effectively capturing player gesture information in the form of physical control parameters.

12:40-1:40 Lunch Break

Contributed Paper

1:40

2aMUb6. How to play the first bar of *Rhapsody in Blue*. Jer-Ming Chen (University of New South Wales, Music Acoustics, School of Physics, NSW 2052 Sydney, Australia, jerming@phys.unsw.edu.au), John Smith (University of New South Wales, Music Acoustics, School of Physics, NSW 2052 Sydney, Australia, john.smith@unsw.edu.au), Joe Wolfe (University of New South Wales, Music Acoustics, School of Physics, NSW 2052 Sydney, Australia, J.Wolfe@unsw.edu.au)

The two-and-a-half octave glissando opening Gershwin's *Rhapsody in Blue* is one of the great icons of 20th century music and one of the best known bars in music. Expert clarinettists combine unusual fingerings with even more unusual configurations of their vocal tract to achieve a nearly continuous rise in pitch. Using a novel method [1], we incorporated an acoustic impedance measurement head within a clarinet mouthpiece, allowing us to study the player's vocal tract at various stages in the glissando. We measured and compared vocal tract impedance spectra with the corresponding clarinet impedance spectra for the fingering used at that pitch. Partially uncovering an open finger-hole raises the frequency of clarinet impedance peaks in the lower register, thereby allowing smooth increases in the playing pitch. In the upper register, however, resonances in the clarinettist's vocal tract are manipulated to be comparable with those in the clarinet for frequencies in this range. Thus the pitch in the higher section of the glissando is largely controlled by smoothly varying a resonance of the player's vocal tract. [1] Chen, JM. Smith, J. and Wolfe, J., (2008) ''Experienced saxophonists learn to tune their vocal tracts''. Science, (in press).

Invited Papers

2:00

2aMUb7. Evaluating Vocal-Tract Influence in the Production of Saxophone Multiphonics. Gary Scavone (Schulich School of Music, McGill University, 555 Sherbrooke Street West, Montreal, QC H3A 1E3, Canada, gary@music.mcgill.ca), Antoine Lefebvre (Schulich School of Music, McGill University, 555 Sherbrooke Street West, Montreal, QC H3A 1E3, Canada, antoine.lefebvre2@mail.mcgill.ca), Andrey R. Da Silva (Schulich School of Music, McGill University, 555 Sherbrooke Street West, Montreal, QC H3A 1E3, Canada, andrey.dasilva@mail.mcgill.ca)

A new approach for the analysis of vocal-tract influence in single-reed woodwind instruments during performance was recently reported (Scavone et. al., 2008). Two types of vocal-tract influence were observed. When the downstream air column provides only weak support of a given note, players can use a strong and narrow-bandwidth upstream resonance to override the reed vibrations, such as when pitch bending or playing extended register notes. Performers can also use a more wide-bandwidth upstream resonance to affect subtle timbre variations when playing notes over the full range of the instrument. The research reported here addresses the performance of multiphonic tones, for which the results of the previously mentioned study were less conclusive. While it is clear that upstream influence is involved in the production of multiphonics, we are interested in determining whether performers must support a specific intermodulation component or a wider bandwidth range of components for proper production. The latest results of this research will be reported.

2:20

2aMUb8. A study of flute control parameters. Benoit Fabre (Institut Jean Le Rond d'Alembert / LAM (UPMC / CNRS / Ministère Culture), 11, rue de Lourmel, 75015 Paris, France, fabreb@ccr.jussieu.fr), Nicolas Montgermont (Institut Jean Le Rond d'Alembert / LAM (UPMC / CNRS / Ministère Culture), 11, rue de Lourmel, 75015 Paris, France, montgermont@lam.jussieu.fr), Patricio De La Cuadra (Centro de Investigación en Tecnologias de Audio (CITA), Universidad Católica de Chile, Alameda 340, Oficina 13, Casilla 114-D Santiago, Chile, pcuadra@uc.cl)

The sound of musical instruments played in self-sustained oscillations can be interpreted as a sounding transposition of the player's gesture. The playing of wind instruments requires expert control of the blowing that may be difficult to grasp and to measure because it induces only very little motion of the player, as opposed to string or keyboard instruments. In the case of instruments of the flute family, the player seems to control mainly the air jet velocity for mode selection. Flutes in which the air jet is formed between the lips also allow for a control of the total jet flow by the player, through lip adjustments. We present measurements carried on several players, in different playing conditions. The score includes technical exercises such as scales and short musical excerpts from the flute repertoire. Time evolutions of the control parameters are analysed, based on pressure and geometrical measurement, in the framework of the current knowledge on the sound production in flutes, such as jet instability. The control over different parameters will be discussed, both for basic technical exercises and in a musical melodical context.

Contributed Paper

2:40

2aMUb9. Acoustical analysis of timbral modulations on the flute as controlled by phonetic gestures. Maryse Lavoie (Laboratoire informatique, acoustique et musique, Faculté de musique, Université de Montréal, C.P. 6128, succursale Centre-Ville, Montréal, QC H3C 3J7, Canada, maryse.lavoie@umontreal.ca), Caroline Traube (Laboratoire informatique, acoustique et musique, Faculté de musique, Université de Montréal, C.P. 6128, succursale Centre-Ville, Montréal, QC H3C 3J7, Canada, caroline.traube@umontreal.ca), Marie-Hélène Breault (Observatoire international de la création et des cultures musicales (OICCM), Faculté de musique, Université de Montréal, C.P. 6128, succursale Centre-Ville, Montréal, QC H3C 3J7, Canada, mhbreault@yahoo.ca)

The purpose of this project is to study the control of timbre on the flute by varying articulatory parameters (i.e. embouchure) which correspond to distinct phonetic gestures. The main goal is to compare the production of a specific timbral modulation on the flute and its vocal reproduction (diphthong) by means of acoustical analyses of these sounds. The recordings of timbral modulations performed on the flute (e.g. by increasing the mouth opening) and reproduced vocally as diphthongs (e.g. [u] -> [a] transition) have been analyzed acoustically for intensity as well as for the first two formant trajectories. The results of the formant analyses as presented in an F1-F2 plane reveal the cardinal vowel triangle and confirm a correlation between the flute timbre modulations and the vocal diphthongs corresponding to the underlying phonetic gestures. In addition, a listening test has shown that flutists are able to perceive and recognize these timbral modulations from the sound alone.

Invited Papers

3:00

2aMUb10. Lip control of brass instruments. Donald M. Campbell (Edinburgh University, 4201 JCMB, Kings Buildings, Mayfield Road, EH9 3JZ Edinburgh, UK, d.m.campbell@ed.ac.uk)

Brass instruments are frequently described in the acoustics literature as ″lip reed″ instruments, emphasising the fact that the sounding mechanism of this instrumental class is flow modulation by the vibration of the player's lips. As well as providing the source of sound, the lips act as the major interface through which the performer controls the intonation, dynamics, articulation and timbre of the performance. Indeed, on instruments such as the natural trumpet and the alphorn, the player's lips form the only control interface. This paper reviews recent studies which have examined the vibrational properties of brass players' lips, the types of motion which they undergo when playing different instruments, and the nature of the interaction between the lips and the instrument. It also considers the extent to which current physical models of the lips are capable of encompassing these control processes.

3:20

2aMUb11. Vocal tract interactions in saxophone performance. Jer-Ming Chen (University of New South Wales, Music Acoustics, School of Physics, NSW 2052 Sydney, Australia, jerming@phys.unsw.edu.au), John Smith (University of New South Wales, Music Acoustics, School of Physics, NSW 2052 Sydney, Australia, john.smith@unsw.edu.au), Joe Wolfe (University of New South Wales, Music Acoustics, School of Physics, NSW 2052 Sydney, Australia, J.Wolfe@unsw.edu.au)

Although acousticians have debated the importance of the vocal tract in reed instrument performance, expert saxophonists report adjusting their vocal tract for advanced techniques including *altissimo* playing, subtone playing, bugling and multiphonics. Using a novel method [1], we incorporated an acoustic impedance head within a saxophone mouthpiece to study the vocal tract directly during playing. For fingerings above the first register, the operating peak in the saxophone's input impedance decreases with increasing pitch, falling to below 20 MPa.s.m^{-3} after 2.7 octaves, thus ending the standard range that is readily available to amateurs. Above this, in the *altissimo*, professional saxophonists produce peaks in the impedance of their tracts of about 20 to 40 MPa.s.m^{-3}, which they tune to select the desired note. The crossover of the relative magnitudes of saxophone and tract impedance peaks coincides with the transition from standard to *altissimo* register. While professionals use the vocal tract thus for other extended effects, inexperienced players do not tune their tract resonances and are unable to produce advanced effects. [1] Chen, JM. Smith, J. and Wolfe, J., (2008) ''Experienced saxophonists learn to tune their vocal tracts''. Science, (in press).

Contributed Paper

3:40

2aMUb12. Reproducibility of piano playing. Nicolas Principeaud (Laboratoire d'Acoustique Musicale, 11 rue de Lourmel, 75015 Paris, France, nicolas.principeaud@free.fr), Xavier Boutillon (Laboratoire de Mécanique des Solides, Ecole Polytechnique, 91128 Palaiseau Cedex, France, boutillon@lms.polytechnique.fr)

Mechanically, piano playing consists in giving to the hammer a certain escapement velocity at a certain time - t_e, v_e - by means of a traditional keyboard and piano action. Numerous replacement systems for the keyboard, action or even for the pianist's finger have been proposed. They implicitly raise the question of the accuracy with which the (t_e, v_e) information must be coded or reconstructed. The precision with which pianists are able to reproduce their playing sheds some light on this question. At this end we have asked several concert pianists to play the same passage several times with the highest reproducibility level they could achieve. The experiment was done on a traditional grand piano equipped with the Yamaha Disklavier system, used as a measurement tool for the escapement velocity of the hammers. The beginnings of a fugue by Bach and a study by Debussy were recorded. Results show that the reproducibility varies highly from note to note and can be as good as approximately 1%. Additional tests on auditors show that deviations on note amplitudes are not perceived unless they reach a level which turns out to be significantly higher than the reproducibility level achieved by the pianists.

TUESDAY MORNING, 1 JULY 2008 ROOM 250A, 8:00 A.M. TO 8:00 P.M.

Session 2aNSa

Noise, Physical Acoustics, and EURONOISE: Aeroacoustics II

Philip J. Morris, Cochair
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Christophe Bailly, Cochair
Ecole Centrale de Lyon, 36, avenue Guy de Collongue, LMFA, Ecully, 69134, France

Contributed Papers

8:00

2aNSa1. Source location prediction of subsonic isothermal jet flows. Juan Battaner-Moro (Institute of Sound and Vibration Research, University of Southampton, University Road, SO17 1BJ Southampton, UK, jpb@isvr.soton.ac.uk), Mahdi Azarpeyvand (Institute of Sound and Vibration Research, University of Southampton, University Road, SO17 1BJ Southampton, UK, ma@isvr.soton.ac.uk), Rod Self (Institute of Sound and Vibration Research, University of Southampton, University Road, SO17 1BJ Southampton, UK, rhs@isvr.soton.ac.uk)

The location of jet noise sources is a far from trivial problem that is of great importance for both understanding the noise production and radiation mechanisms and also for finding new jet noise reduction strategies. This paper presents comparisons of theoretical results with data for a number of jets. The theory used is based on the MGBK method but including a novel time scale based on the rate of energy transfer through the turbulent cascade. This new technique has been shown to give a number of advantages over existing models. The experimental results were obtained using the Polar Correlation Technique and were made at QinetiQ's Jet Noise Facility in the UK as part of the EU FP6 programme CoJeN. The high resolution jet noise images resulted from using a 64 microphone polar arc array set at two reference angles, namely 60 and 90 degrees to the jet axis. Comparisons with experimental data are made for coplanar and short cowl nozzles at different working conditions for predictions from different theoretical models. It is shown that the best agreement is obtained for the prediction methodology using the energy transfer rate timescale

8:20

2aNSa2. A hybrid method for jet noise predictions based on Large Eddy Simulation and Reynolds-Averaged Navier-Stokes simulations. Guillaume Bodard (Snecma Villaroche, Rond point René Ravaud, 77550 Moissy-Cramayel, France, guillaume.bodard@snecma.fr), Christophe Bailly (Ecole Centrale de Lyon, 36, avenue Guy de Collongue, LMFA, 69134 Ecully, France, christophe.bailly@ec-lyon.fr)

A hybrid jet noise prediction method combining steady and unsteady flow calculations is discussed. The main objective is to merge advantages of each technique to obtain a robust acoustic prediction tool, which will be able to correctly evaluate design and installation effects such as chevron nozzles or jet-pylon interaction for instance. The low-frequency component of acoustic spectra is computed using large-eddy simulations and the integral formulation derived by Ffowcs-Williams & Hawkings (1969, Phil. Trans. Roy. Soc. Lond., vol. 264). The high-frequency component associated with fine scale turbulence is obtained thanks to Tam & Auriault's mixing noise theory (1999, AIAA Journal, vol. 37) from Reynolds-Averaged Navier-Stokes simulations. The oral presentation will detail the two methods, the LES simulations for subsonic round single and coaxial jets, and the matching between the two approaches to get a complete picture of the acoustic spectra.

8:40

2aNSa3. Efficiency of optimized microjets on realistic nozzles. Alexandre Vuillemin (Snecma, Etablissement de Villaroche Sud, Rondpoint René Ravaud - Réau, 77550 Moissy-Cramayel, France, alexandre.vuillemin@snecma.fr)

Even if chevrons nozzles are an efficient way to reduce jet noise during take off, they also decrease performances during cruise. An innovative way to avoid this decrease is the use of active devices that could be switch off after take off, such as microjets nozzles. Snecma led test campaigns at Martel Facility (LEA/CEAT) to optimize microjets settings (geometric and thermodynamic parameters) and to characterize their efficiency on both 2D axisymmetric and chevrons nozzles. Acoustic and PIV measurements were done on a single stream hot jet with and without external flow, and on a double stream hot jet with pylon.

9:00

2aNSa4. Numerical investigation of the effect of nonlinear propagation distortion on helicopter noise. Penelope Menounou (University of Patras, Department of Mechanical and Aeronautical Engineering, Rion, 26504 Patras, Greece, menounou@mech.upatras.gr), Panagiotis

Vitsas(University of Patras, Department of Mechanical and Aeronautical Engineering, Rion, 26504 Patras, Greece, pvitsas@upnet.gr)

Nonlinear propagation distortion causes energy to be shifted to the high frequency end of the spectrum. This leads to underestimation of the noise levels at high frequencies. The effect has been demonstrated in the case of aircraft noise, but less attention has been given to helicopters. In the present work, the effect of nonlinear propagation distortion on helicopter noise is demonstrated based on measured data for low-speed descent and numerical calculations that predict the noise level away from the helicopter with and without nonlinear effects. It is shown that (i) for some frequency bands the difference between linear and nonlinear calculations can be as high as 8 dB, (ii) frequencies between 1000 and 3000 Hz are more affected, and (iii) the effect is highly directional depending on the receiver location around the helicopter. It is further shown that nonlinear effects manifest themselves differently than in the case of aircraft noise and that they depend on the specific helicopter noise mechanism. More specifically, the following helicopter-specific noise source types are investigated with regards to nonlinear effects: advancing Blade Vortex Interaction (BVI) noise vs retreating BVI noise, High Speed Impulsive noise, and BVI vs Very Impulsive noise.

Invited Papers

9:20

2aNSa5. Recent developments in helicopter rotor noise prediction. Kenneth S. Brentner (Penn State University, 233C Hammond Building, University Park, PA 16802, USA, ksb16@psu.edu)

Prediction of discrete frequency noise for rotorcraft in steady flight has reached a high level of sophistication and understanding. The primary challenge in such rotor noise predictions is the accurate determination of the loading on and flow field around the blades. Although current rotor noise prediction tools have been demonstrated for steady flight conditions, the utility of rotorcraft comes from their unique ability to hover and maneuver. Such maneuvers often occur near the ground in close proximity to people. This paper describes an initial study to characterize maneuver noise. A maneuver noise prediction system has been developed, which consists of a flight simulation code, free-vortex wake code, and a maneuver noise prediction code, loosely coupled together. One of the key aspects of this system is the ability to investigate the additional noise caused during the transition from one flight state to another. Several maneuvers including turns, accelerations, and pop-up/popdown maneuvers were considered. Significant increases in the low-frequency noise can occur, depending on how aggressive the maneuver is performed. The status of advanced research on predicting acoustic scattering of the rotor noise by the aircraft and the first-principles prediction of rotor broadband noise will also be presented.

9:40

2aNSa6. Aeroacoustic research complex for aircraft source noise characterization. Micah Downing (Blue Ridge Research and Consulting, 13 1/2 W. Walnut St., Asheville, NC 28801, USA, micah.downing@blueridgeresearch.com), Robert McKinley (Air Force Research Laboratory, AFRL/RHCB, 2610 Seventh St., Bldg 441, Wright-Patterson AFB, OH 45433, USA, Robert.McKinley@wpafb.af.mil), John Hall (Air Force Research Laboratory, AFRL/RHCB, 2610 Seventh St., Bldg 441, Wright-Patterson AFB, OH 45433, USA, John.Hall@wpafb.af.mil), Frank Mobley (Air Force Research Laboratory, AFRL/RHCB, 2610 Seventh St., Bldg 441, Wright-Patterson AFB, OH 45433, USA, Frank.Mobley@wpafb.af.mil), Michael James (Blue Ridge Research and Consulting, 13 1/2 W. Walnut St., Asheville, NC 28801, USA, michael.james@blueridgeresearch.com)

Aircraft noise has been traditionally measured with either a few ground-based microphones or a linear ground-plane array of microphones. These techniques capture one-dimensional and/or two-dimensional characteristics of aircraft flight noise. The US Air Force Research Laboratory has started the construction of a 3-dimensional measurement facility at White Sands Missile Range in New Mexico. This facility, the Aeroacoustic Research Complex (ARC), will allow aircraft to fly through the array, collecting fully 3D acoustic data. ARC is initially being developed in two phases The first phase includes two 91.4 m tall towers separated by 244 m and will focus on noise from rotary wing and UAV aircraft. The second phase will add two 366 m tall towers separated by 610 m and will focus on large and high performance fixed wing aircraft. This facility will allow more accurate characterization of in-flight noise directivity by providing synchronized 3-dimensional magnitude & spectral acoustical signatures from 50+ microphones. ARC responds to a critical need for validation of existing predictive acoustic models. Such models are used for aircraft design, survivability, nonlinear acoustic propagation research and assessing noise exposure to residents living adjacent to airfields.

10:00-10:20 Break

Contributed Papers

10:20

2aNSa7. Lateral directivity of aircraft noise. Walter Krebs (Empa, Ueberlandstrasse 128, 8600 Duebendorf, Switzerland, walter.krebs@empa.ch), Georg Thomann (Empa, Ueberlandstrasse 128, 8600 Duebendorf, Switzerland, georg.thomann@empa.ch)

A three-dimensional model was developed to characterise the directional sound emission of different aircraft. The model is based on spherical harmonics and defines the directional spectral sound pressure level at a reference distance. The parameters of the model are derived from acoustic measurements on real aircraft traffic. With the help of this model different physical effects on sound propagation and sound impact are analysed. Variations with respect to a rotational symmetric sound emission are outlined and compared to the engine installation corrections proposed in the revised Doc 29 3rd edition. In addition the influence of the ground effect on A-weighted sound levels is analysed for different receiver heights and ground impedances.

10:40

2aNSa8. Sonic booms, spectral analysis, and diffraction by buildings. Victor W. Sparrow (Penn State, Graduate Program in Acoustics, 201 Applied Science Bldg., University Park, PA 16802, USA, vws1@psu.edu)

An analysis has been performed upon conventional and low-amplitude N-wave sonic boom data taken by NASA outside two houses in June 2006 and July 2007. The buildings were one-floor residences which were carefully instrumented with strategically-placed, multiple microphones. The incident sonic booms had a substantial variation in rise times, most likely due to atmospheric effects. Previous work using the June 2006 data [Sparrow, Klos, and Buehrle, J. Acoust. Soc. Am. 122 (5, Pt. 2) 3084] revealed maximum pressure loads near the ground and wall facing the incident boom. A new spectral analysis of the individual booms now confirms that the diffraction of sound over the house tops is substantially affected by the spectral content of each boom. These results indicate that the pressure loading of homes due to conventional or low-amplitude sonic boom will be affected by

both the spectral content of each boom's rise phase and each house's exterior geometry. [Work supported by NASA.]

11:00

2aNSa9. Identification of aerodynamic sound sources: the key problem in noise control. Alexander Fedorchenko (Independent Researcher/Consultant, Dm. Ul'yanov Str. 27/12-1-50, 117449 Moscow, Russian Federation, fedorchenko@mail.com)

Despite the recent advances in the noise control technology, the key mechanisms of aerodynamic sound emission remained poorly studied. The ″well-recognized″ methods of TCAA did not lead to any breakthrough in this topical problem, and what seems most disappointing, the family of ″acoustic analogies″ is still used by many for definition of aerodynamic noise sources, though the sufficient set of mathematical proofs has been given by the author that this model is wrong. Experimental approaches aimed at localization of aerodynamic noise sources, including the method of acoustic imaging via using a microphone array, are considered, and their inherent limitations are pointed out. Unfortunately, no current experimental technique enables one to measure instantly all sound sources and sound disturbances inside the zone of generation, and so the latter is often regarded as a kind of ″black box″. The two-medium nonlinear theory of aerodynamic sound, based on the original decomposition of each flow variable into two components, for unsteady background flow and for acoustic field, has been created, that promotes better comprehension of the noise generation phenomena and opens new ways in flow/noise control. The main properties of this theory are now indicated in comparison with the most detrimental delusions originated from ″acoustic analogies″.

Invited Paper

11:20

2aNSa10. Direct aeroacoustic simulations based on domain decompositions. Jens Utzmann (University of Stuttgart, Institute for Aerodynamics and Gasdynamics, Pfaffenwaldring 21, 70569 Stuttgart, Germany, utzmann@iag.uni-stuttgart.de), Claus-Dieter Munz (University of Stuttgart, Institute for Aerodynamics and Gasdynamics, Pfaffenwaldring 21, 70569 Stuttgart, Germany, munz@iag.uni-stuttgart.de)

For CAA, an accurate and feasible direct simulation that considers both the generation of sound within the flow and its propagation into the far-field is hard to realize with one numerical method in a single computational domain. On the other hand, a direct approach contains automatically the interaction of the acoustic perturbations with the flow-field, a property which lacks the popular acoustic analogy models. The proposed method is basically a direct simulation, but it simplifies the problem that has to be solved for individual regions in the computational domain. The idea is to use a non-overlapping domain decomposition method where the equations, methods, grids and even the time steps are adapted to meet the local requirements. Inside the coupling framework, high-order solvers from different classes of methods are available: On unstructured grids, a reconstructed ADER finite volume method (ADER-FV) is used for linear and nonlinear problems, as well as a discontinuous Galerkin method. On structured grids, the ADER-FV and the ADER-FD method are efficiently implemented for nonlinear (FV) and linear (FV, FD) problems. These high-order accuracy methods ensure excellent wave propagation capabilities throughout the entire computational domain. In the subdomains, the Navier-Stokes, Euler and the linearized Euler equations are solved.

Contributed Papers

11:40

2aNSa11. Direct Noise Computation in subsonic and transonic flows. Frédéric Daude (LaMSID UMR EDF/CNRS, 1 avenue du Général de Gaulle, 92141 Clamart, France, frederic-externe.daude@edf.fr), Thomas Emmert (LaMSID UMR EDF/CNRS, 1 avenue du Général de Gaulle, 92141 Clamart, France, tommem@gmail.com), Philippe Lafon (EDF, 1, avenue du Général de Gaulle, 92141 Clamart, France, Philippe.lafon@edf.fr), Fabien Crouzet (EDF, 1, avenue du Général de Gaulle, 92141 Clamart, France, fabien.crouzet@edf.fr), Christophe Bailly (Ecole Centrale de Lyon, 36, avenue Guy de Collongue, LMFA, 69134 Ecully, France, christophe.bailly@ec-lyon.fr)

In order to model flow phenomena involving interactions between aerodynamics and acoustics, it is necessary to use Direct Noise Computation (DNC) instead of hybrid methods that are not suitable to take into account the feedback of acoustics on the flow. The methods that are now available in the field of Computational AeroAcoustics (CAA) allows us to deal with DNC in realistic configurations. The numerical code SAFARI (Simulation of Aeroacoustics in Fluids And with Resonance and Interactions) has been developed for this goal. The set of equations are the compressible 3-D Navier-Stokes equations. High order finite difference schemes are used. Multidomain capabilities are implemented in order to deal with complex geometries. Block decomposition is used in order to take advantage of parallel processing on large clusters Validation cases are presented: diffraction by a cylinder, shock tube. Results on realistic configurations are also shown: ducted cavity, transonic sudden enlargment, airfoil interactions.

12:00

2aNSa12. Asymptotic expressions for the directivity of round jets. Ricardo E. Musafir (UFRJ / COPPE, Universidade Federal do Rio de Janeiro, 21941-972 Rio de Janeiro, Brazil, rem@serv.com.ufrj.br)

Expressions for the directivity of round jets, in the high and low frequency limits, are derived, based on solutions of Lilley's equations. Two different forms of the equation in what concerns source terms representation are considered, along with general point sources of the appropriate type (i.e., equivalent stress and force sources and also, when necessary, volume sources), which are assumed to be statistically axisymmetric. Effects of mean shear and temperature gradients are accounted for, as well as of source movement. The expressions are compared with earlier ones, based on a plug flow model. The differences in the resulting expressions due to the choice of source description, which are relevant for hot jets, are discussed and used to derive a form consistent with both representations. Comparison with experimental data is also presented.

12:20

2aNSa13. Investigations of roughness-generated TBL sound using coupled physical-computational experiments in conjunction with theoretical development. William Blake (Naval Surface Warfare Center (Ret.), 6905 Hillmead Road, Bethesda, MD 20817, USA, hydroacoustics@aol.com), Ki-Han Kim (Office of Naval Research, Randolph St., Arlington, VA 22202, USA, kihan.kim@navy.mil), Michael Goody (Naval Surface Warfare Center (Ret.), 6905 Hillmead Road, Bethesda, MD 20817, USA, michael.goody@navy.mil), Meng Wang (University of Notre Dame, Departmernt of Aerospace and Mechanical Engineering, Notre Dame, MD 46556, USA, m.wang@nd.edu), William J. Devenport (Virginia Tech, Aerospace and Ocean Engineering, 224E Randolph Hall, Blacksburg, VA 24061, USA, devenport@vt.edu), Stewart A. Glegg (Florida Atlantic University, Department of Ocean Engineering, Boca Raton, VA 33431, USA, glegg@oe.fau.edu)

Sound produced by turbulent-boundary layers (TBL) over rough walls is being studied in a series of physical-computational experiments. At issue is the development of a knowledge of how the wall elements generate flow dipoles which directly determines how the sound is described in terms of dependent variables. The considered mechanisms include dipoles at the

roughness elements due to their shed wakes, distributed surface dipoles due to convecting turbulence impinging onto elements, Rayleigh-like scattering into sound of aerodynamic pressures of hydrodynamic wave numbers of flow above the roughness. The LES of rough-wall TBL consists of ″numerical″ experiments being used to isolate the separate mechanisms. These simulations are benchmarked with analysis and with matching physical experiments on rough wall patches in which identical geometries of wall roughness and identical Reynolds numbers are used. In the physical measurements, array-based measurements of the radiated sound are being used to characterize the directivity and magnitude of the sound and to relate the sound to aerodynamic wall pressure and to classical characteristics of the turbulent boundary layer. The LES produces comparison for both radiated sound and detailed flow structure around the roughness elements. This project is funded by the Office of Naval Research, Washington, D.C., USA.

12:40-2:00 Lunch Break

Contributed Paper

2:00

2aNSa14. Experimental investigation of sound from flow over a rough surface. Michael Goody (Naval Surface Warfare Center (Ret.), 6905 Hillmead Road, Bethesda, MD 20817, USA, michael.goody@navy.mil), Jason Anderson (Naval Surface Warfare Center (Ret.), 6905 Hillmead Road, Bethesda, MD 20817, USA, Jason.M.Anderson@navy.mil), Devin Stewart (Naval Surface Warfare Center (Ret.), 6905 Hillmead Road, Bethesda, MD 20817, USA, Devin.Stewart@navy.mil), William Blake (Naval Surface Warfare Center (Ret.), 6905 Hillmead Road, Bethesda, MD 20817, USA, hydroacoustics@aol.com)

Measurements of radiated noise and unsteady surface pressures have been carried out in order to better understand the mechanism for sound production from flow over a rough surface. In order to investigate scaling relationships, the flow speed, roughness height, geometry and element distribution were varied systematically. When considered in total, previous investigations do not present a consistent picture of the scaling behavior of roughness noise, or the underlying physical mechanism. They have reported roughness noise levels that scale on flow velocity, roughness height, and fetch area and have indicated that the sound production may be dipole or quadrupole in nature. Prevailing analytical models assume that both dipole and quadrupole sources are present. The scaling of roughness noise for large roughness height has not been investigated previously and is part of the current study. A recent developed scattering model (dipole) developed by Glegg et al is interrogated using detailed measurements of the roughness element height distribution, turbulent boundary layer properties, and array-based radiated sound levels.

Invited Paper

2:20

2aNSa15. On the sound generated by boundary-layer vorticity. Umberto Iemma (University Roma Tre, via vasca navale 79, 00146 Rome, Italy, u.iemma@uniroma3.it), Luigi Morino (via vasca navale 79, 00146 Roma, Italy, l.morino@uniroma3.it), Roberto Camussi (University Roma Tre, via vasca navale 79, 00146 Rome, Italy, camussi@uniroma3.it), Giovanni Caputi Gennaro (University Roma Tre, via vasca navale 79, 00146 Rome, Italy, gcaputi@uniroma3.it)

Turbulent boundary layers generate broadband noise as the effect of vortical-disturbances scattering into acoustic waves. The paper presents a formulation for evaluating of acoustic pressure in the field in terms of the transpiration velocity, here defined in terms of vorticity and closely related to Lighthill equivalent source. Specifically, the formulation used allows one to obtain, in the frequency domain, a matrix relationship between the transpiration velocity at a number of points on the body surface (those arising from boundary-element discretization) and the pressure at given points in the irrotational region. From this, the relationship between the corresponding PDF is easily obtained using the Wiener-Khintchine theorem. The paper will include the general formulation, validating numerical results, and comparison with experimental data. The inverse problem, that is, determining the source intensity on the body surface from field noise (in particular, the invertibility of the operator), will be also addressed.

Contributed Papers

2:40

2aNSa16. A numerical study on multimode sound propagation in lined ducts and radiation to the far field. Rie Sugimoto (ISVR, University of Southampton, Highfield, SO17 1BJ Southampton, UK, rs@isvr.soton.ac.uk) , R Jeremy Astley (ISVR, University of Southampton, Highfield, SO17 1BJ Southampton, UK, rja@isvr.soton.ac.uk), Claire R. McAleer (ISVR, University of Southampton, Highfield, SO17 1BJ Southampton, UK, crm@soton.ac.uk), Iansteel Achunche (ISVR, University of Southampton, Highfield, SO17 1BJ Southampton, UK, ia1@isvr.soton.ac.uk)

In previous articles, the authors developed a hybrid scheme for analysing bypass duct noise, in which a numerical analysis using finite element method for in-duct propagation and an analytic radiation code with fully represented effects of bypass shear layer are coupled. Such procedure permits detailed study on the interaction between duct configurations, such as geometry and acoustic liner impedances, and modal propagation and attenuation, and also the effects on the radiation pattern, within practical timescale and at modest computational cost. The scheme has been applied to realistic aero-engine bypass ducts and has been integrated with an optimisation programme. The numerical results obtained so far have revealed that for ducts with acoustic liners highly attenuated modes are not necessarily those with high mode angles, which is contrary to general anticipation. The aim of the study in the current paper is to understand the physics behind this phenomenon and its effect on the radiation to the far field. Detailed investigation on modal attenuation is performed by using the hybrid scheme. The effect of modal power distribution at the duct exit on the directivity pattern of the radiated noise to the far field is also discussed.

3:00

2aNSa17. Modeling of sound propagation in nonuniform waveguides. Wenping Bi (Laboratoire d'Acoustique de l'Université du Maine, UMR CNRS 6613, AV. O. Messiaen, 72085 Le Mans, France, wenping.bi@univ-lemans.fr), Vincent Pagneux (Laboratoire d'Acoustique de l'Université du Maine, UMR CNRS 6613, AV. O. Messiaen, 72085 Le Mans, France, vincent.pagneux@univ-lemans.fr), Denis Lafarge (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, denis.lafarge@univ-lemans.fr), Yves Aurégan (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, yves.auregan@univ-lemans.fr)

Sound propagation in waveguides is modeled by a Multimodal Method. The waveguides geometries may involve bends, variable cross-sections, or their combinations. The waveguide boundaries may involve axially or circumferentially nonuniform impedance conditions or acoustically rigid

conditions. Uniform flow may also be included for a simple uniform geometry. The pressure (displacement potential for uniform flow) is expanded in terms of the modes of acoustically rigid waveguides and an additional function that carries the information about the impedance boundary. The rigid waveguide modes and the additional function are known a priori so that calculations of the true modes of waveguides with impedance boundary, which are difficult, are avoided. By matching the pressure and axial velocity (displacement potential and axial derivative for uniform flow) at the interface between different axially uniform segments, scattering matrices are obtained for each individual segment; these are then combined to construct a global scattering matrix for multiple segments. After calculating the scattering matrix, the transmitted and reflected sound fields or intensities may be obtained for any kind of modal sources. The method allows modeling sound propagation in waveguides with axially and circumferentially nonuniform impedance boundaries up to dimensionless frequency K=70 in just hours on a personal computer, which advantageously compares with other techniques.

3:20

2aNSa18. An Active Network Representation to Predict the Flow Noise Characteristics of Corrugated Ducts. Wim De Roeck (K.U.Leuven - Dept. of Mechanical Engineering, Celestijnenlaan 300B - bus 2420, 3001 Heverlee, Belgium, wim.deroeck@mech.kuleuven.be),Vasilisa Solntseva (Andreev Acoustics Institute, Shvernik, 4, Moscow, 117036 Moscow, Russian Federation, mironov@akin.ru), Wim Desmet (K.U.Leuven - Dept. of Mechanical Engineering, Celestijnenlaan 300B - bus 2420, 3001 Heverlee, Belgium, Wim.Desmet@mech.kuleuven.be)

In this paper, an active network representation is used to numerically predict and gain more insight in the internally generated flow noise sources in corrugated ducts (the active bi-port component) as well as to describe the acoustic transmission characteristics of this type of application (the passive bi-port components) in the presence of a non-uniform mean flow. For this purpose a numerical approach is chosen, using compressible Large Eddy Simulations (LES) to predict the noise generation mechanisms and Linearized Euler Equations (LEE) with a plane pulse excitation to obtain the transmission characteristics for these components. The accuracy of the active network component determination is increased by separating the aerodynamic and the acoustic fluctuating field of the LES using both an aerodynamic/acoustic splitting technique and multiple plane mode-matching strategies. In this way, a numerical method is proposed to analyze the flow-acoustic behavior of corrugated tubes, which can be generally used for all types of duct system applications.

3:40-5:20 Posters

Lecture sessions will recess for presentation of posters on various topics in acoustics. See poster sessions for topics and abstracts.

Contributed Papers

5:20

2aNSa19. Validation of a hybrid method of aeroacoustic noise computation applied to internal flows. Mélanie Piellard (Delphi Thermal, Avenue de Luxembourg, 4940 Bascharage, Luxembourg, melanie.piellard@delphi.com), Christophe Bailly (Ecole Centrale de Lyon, 36, avenue Guy de Collongue, LMFA, 69134 Ecully, France, christophe .bailly@ec-lyon.fr)

A hybrid method of aeroacoustic noise computation based on Lighthill's acoustic analogy is first validated, and then applied to investigate the noise radiated by a low Mach number flow through a diaphragm in a duct. The simulation method is a two-step hybrid approach relying on Lighthill's acoustic analogy, assuming the decoupling of noise generation and propagation. The first step consists in an incompressible Large Eddy Simulation of the turbulent flow field, during which the Lighthill's source term is recorded. In the second step, a variational formulation of Lighthill's Acoustic Analogy using a finite element discretization is solved in the Fourier space. The validation of this method is briefly presented: a general validation is performed on the case of two corotating vortices in a medium at rest; the exit of turbulent structures from the computational domain is accounted for by a spatial filtering; and a study of spatial interpolation from the CFD mesh to the acoustic mesh shows an acceptable level of error. This method is applied to a three-dimensional diaphragm with low Mach number flow, showing good agreement with both experimental results and Direct Noise Computation performed by Gloerfelt & Lafon (Computers & Fluids, 2007).

5:40

2aNSa20. Numerical strategies for investigation of gust-airfoil interaction. Florent Margnat (Arts et Métiers Paris Tech - Sinumef Lab, 151 bd de l'Hopital, 75013 Paris, France, florent.margnat@paris .ensam.fr), Thomas Le Garrec (Arts et Métiers Paris Tech - Sinumef Lab, 151 bd de l'Hopital, 75013 Paris, France, thomas.le-garrec@paris .ensam.fr), Djaafer Fedala (Lab. d'Energétique et de Mécanique des Fluides Interne, Arts et Métiers ParisTech, 151 boulevard de l'Hôpital, 75013 Paris, France, djaafer.fedala@paris.ensam.fr),Xavier Gloerfelt(Arts et Métiers Paris Tech - Sinumef Lab, 151 bd de l'Hopital, 75013 Paris, France, xavier.gloerfelt@paris.ensam.fr), Smaine Kouidri (LIMSI-CNRS, BP 133, 91403 Orsay Cedex, France, smaine.kouidri@limsi.fr)

The noise generated by the interaction between a gust and an airfoil in a uniform flow is investigated. This problematic is of major industrial interest, regarding fans, turbomachinery, or wind turbine applications. A two-dimensional symmetric Joukowski-type airfoil is immersed without incidence in a flow at Mach number 0.5, disturbed by a harmonic gust at 45° of incidence (4th CAA Workshop on Benchmark Problems, 2004). Our methodology is first to perform a high-order direct resolution of Euler's equations of the disturbed flow over the airfoil and the associated acoustic emission, which is taken as a reference simulation. Second, the near aerodynamic field is simulated with Fluent 6.3 solver based on finite volume method with second-order schemes. The aerodynamic data thus obtained are used for far field acoustic prediction, based on Ffowcs Williams and Hawkings analogy. Finally, following another hybrid approach, the noise is predicted by using integral formulations with source field from the DNS. The aim of the study is to provide insight into the efficiency and validity of these numerical strategies commonly used. Comparisons with results of the CAA workshop are given, covering various wavenumbers values.

6:00

2aNSa21. A stochastic source model for turbulent noise prediction including sweeping time dynamics. Malte Siefert (German Aerospace Center, Lilienthalplatz 7, 38108 Braunschweig, Germany, malte.siefert@dlr.de), Roland Ewert (DLR/Institute of Aerodynamics and Flow Technology, Lilienthalplatz 7, 38108 Braunschweig, Germany, Roland .Ewert@dlr.de)

We extend a low-cost computational aeroacoustic approach by taking into account temporal effects of the modeled turbulent flow. As the noise production is determined by the turbulent frequency spectrum rather than the wavenumber spectrum, temporal properties of the turbulent flow play a role in many cases. We focus on the most dominant temporal effect, on sweeping, i. e. the advection of inertial range structures by the energy containing large scales. By introducing a feedback-mechanism of the turbulent field onto itself, we are able to incorporate this effect in the fast random particle method, which was successfully applied to different cases such as

slat noise, jet noise and others. It is shown that sweeping reproduces important properties of spatio-temporal correlations of the turbulent flow, which are not grasp by most turbulence-models. The influence on the sound generation will be discussed for aeroacoustic simulations of jet and trailing edge noise.

6:20

2aNSa22. Aeroacoustic simulation based on linearized Euler equations and stochastic sound source modelling. Hervé Dechipre (Volkswagen AG, Brieffach 1777, 38436 Wolfsburg, Germany, herve.dechipre@volkswagen.de), Michael Hartmann (Volkswagen AG, Brieffach 1777, 38436 Wolfsburg, Germany, michael.Hartmann2 @volkswagen.de), Jan W. Delfs (DLR/Institute of Aerodynamics and Flow Technology, Lilienthalplatz 7, 38108 Braunschweig, Germany, jan.delfs@dlr.de), Roland Ewert (DLR/Institute of Aerodynamics and Flow Technology, Lilienthalplatz 7, 38108 Braunschweig, Germany, Roland .Ewert@dlr.de)

In order to continually improve passenger car acoustic comfort, effective methods are needed to simulate the noise generated by external flows as well as the noise produced by duct flows in HVAC systems. The present paper illustrates the use of two methods based on the Linearized Euler equations and derivatives thereof to compute the acoustic field. The two methods implemented in the DLR's aeroacoustic code PIANO are the introduction of a perturbation (vortex) in the flow and the stochastic sound source modelling by Ewert. The application considered by this study is the noise produced by a flat plate in a two-dimensional duct at a thickness related Reynolds number of 1300. Different sizes and edge shapes of the flat plate have been tested. The simulations show a good agreement of the results obtained by the two methods as well as the existence of a vortex street behind the plate corresponding to a Strouhal number of 0.18. It could be shown computationally, that even in the absence of the classical Aeolian tone generation (pure broadband turbulence related trailing edge noise generation) resonance type phenomena occur in the duct. The computed modes for the different configurations show encouraging correlations with the Parker-type modes or resonance phenomena described by Koch.

6:40

2aNSa23. Active Perturbation on Vortex-Induced Acoustic Resonance. Li Cheng (Hong Kong Polytechnic University, Hung Hom, Hong Kong, Hong Kong, mmlcheng@polyu.edu.hk), Y Zhou (Hong Kong Polytechnic University, Hung Hom, Hong Kong, Hong Kong, mmyzhou@polyu.edu.hk) , Ming Ming Zhang (Hong Kong Polytechnic University, Hung Hom, Hong Kong, Hong Kong, mingming@jhu.edu)

Flow-induced acoustic resonance results from strong interactions between unsteady separated flows and the acoustic modes of a cavity. This work explores the feasibility of using piezoelectric-actuator-based perturbation technique in the control of vortex-induced noise through systematic experimental studies. A thick rectangular plate was used as the vortex generator and placed upstream of a cavity. Curved piezo-ceramic actuators were embedded in a slot on the top side of the plate to provide a perturbation to the flow. Uncontrolled flow-acoustic interaction was first examined to provide a baseline for comparison. Results show that noises induced by flow separation from the thick rectangular plate and the wall ahead of the cavity have very different critical flow velocities so that their effects can be well separated. Open-loop control tests indicate that vortex-induced acoustic resonance can be successfully controlled using the proposed technique. Analyses suggest that the convection of votices separated from the leading edge along the plate surface was accelerated by the surface perturbation, which interacted vigorously with the formation of the dominant trailing edge vortex, thus weakening the vortex strength in the wake of the plate. This weakened vortex strength substantially alters the flow-acoustic interaction, resulting in a significant impairment of vortex-induced acoustic resonance. (Supported by Research Grants Council of HKSAR. PolyU 5132/07E)

7:00

2aNSa24. Sound amplification in a lined duct with flow: PIV measurements. David Marx (Laboratoire d'Etudes Aerodynamiques - CNRS, Bat K, 40 avenue du recteur Pineau, 86022 Poitiers, France, david.marx@lea.univ-poitiers.fr), Yves Aurégan (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, yves.auregan@univ-lemans.fr), Helène Bailliet (Laboratoire d'Etudes Aerodynamiques - CNRS, Bat K, 40 avenue du recteur Pineau, 86022 Poitiers, France, helene.bailliet@lea .univ-poitiers.fr), Jean-Christophe Valière (Laboratoire d'Etudes Aérodynamiques (LEA), Université de Poitiers, ENSMA, CNRS, Bat K, 40 avenue du recteur Pineau, 86022 Poitiers, France, jean-christophe.valiere@lea.univ-poitiers.fr)

An experimental investigation of the acoustic behavior of a liner in a rectangular channel with grazing flow has been conducted. The liner is a locally reacting structure. When increasing the velocity of the grazing flow the transmission coefficient increases at resonance frequency. The transmission coefficient can even become larger than 1 meaning that acoustical energy is produced by the liner. This amplification of the sound wave is accompanied by an increase in the stationary pressure drop induced by the liner (up to 300%). This effect is attributed to a modification of the flow induced by the acoustic wave. Thus, the flow is measured using Particle Image Velocimetry (PIV) imaging technique and a comparison of velocity maps with and without sound amplification is performed. Some differences occur and will be presented.

7:20

2aNSa25. Acoustic PIV: Measurement of the acoustic particle velocity using synchronized PIV-technique. André Fischer (German Aerospace Center (DLR), Mueller-Breslau-Str. 8, 10623 Berlin, Germany, andre.fischer@dlr.de), Emilie Sauvage (Ecole Polytechnique de l'Univerité d'Orléans, 8 rue Léonard de Vinci, 45072 Orléans, France, milie.sauvage@gmx.net), Ingo Roehle (German Aerospace Center (DLR), Mueller-Breslau-Str. 8, 10623 Berlin, Germany, ingo.roehle@dlr.de)

This paper outlines a technique for measuring the acoustic particle velocity and the flow field simultaneously by applying synchronized particle image velocimetry (PIV). As test set-up a squared acrylic glass chamber was chosen. One side of the test section is connected to a loudspeaker, which allows a sinusoidal excitation of the chamber. To point out constrains of this method the investigation includes an analysis of excitation amplitude and frequency as well as the effect of the mean flow magnitude. Therefore a small PC fan can be mounted inside the test section to produce an adjustable mean flow. It can be shown that for a low number of averaged images (80) reasonable results can be achieved up to a certain level of fan rotation speed. Beyond this level the turbulence sensitivity increases and more images are necessary for the calculations. However, the acoustic particle velocity can be computed in the presence of turbulent flow. The presented method called acoustic PIV is a non intrusive technique, applied successfully in measuring acoustic particle velocity fields over a wide range of conditions.

7:40

2aNSa26. A finite element method for time harmonic acoustics in arbitrary flows. Anne-Sophie Bonnet- Ben Dhia (CNRS, ENSTA - 32 Boulevard Victor, 75 015 Paris, France, bonnet@ensta.fr), Jean-Francois Mercier (CNRS, ENSTA - 32 Boulevard Victor, 75 015 Paris, France, jmercier@ensta.fr), Florence Millot (CERFACS, 42 avenue Gaspard Coriolis, 31 057 Toulouse, France, millot@cerfacs.fr), Sebastien Pernet (CERFACS, 42 avenue Gaspard Coriolis, 31 057 Toulouse, France, pernet@cerfacs.fr)

The reduction of noise in aeronautics motivates an intensive research in aeroacoustics. In particular, there is a need for efficient tools to simulate acoustic propagation in a mean flow. We are interested here by solving the linearized problem in the frequency domain, by a finite element method able to take into account general geometries and flows. To our knowledge, only

the potential case has been completely handled. Recently, a new approach has been developed and validated in the case of a parallel shear flow: it relies on a regularized formulation of Galbrun's equation, well-suited for a discretization by Lagrange finite elements, combined with Perfectly Matched Layers. A drawback of the method comes from the additional term of regularization, which requires the evaluation of an oscillating integral, coupling all degrees of freedom located on the same streamline. This difficulty can be avoided by replacing this non-local term by its Low-Mach approximation We show here how to extend this Low-Mach approach to the case of a non parallel flow. Numerical experiments are done. In the case of a potential mean flow, a good agreement with the exact approach is observed, even for quite large Mach numbers.

TUESDAY MORNING, 1 JULY 2008 ROOM 251, 8:00 TO 10:00 A.M.

Session 2aNSb

Noise, Biomedical Ultrasound/Bioresponse to Vibration, ASA Committee on Standards, and EURONOISE: Session in Honor of Henning von Gierke

Paul Schomer, Cochair
Schomer and Associates, Inc., 2117 Robert Drive, Champaign, IL 61821, USA

Brigitte Schulte-Fortkamp, Cochair
TU Berlin, Institute of Fluid Mechanics and Engineering Acoustics, Einsteinufer 25, Sekr. TA 7, Berlin, D-10587, Germany

Invited Papers

8:00

2aNSb1. Where do we stand on standards for noise? William Lang (Noise Control Foundation, 29 Hornbeck Ridge, Poughkeepsie, NY 12603, USA, lang1ww@gmail.com)

Almost 40 years ago, Henning von Gierke organized and chaired a symposium at the ASA Cleveland meeting with the above title. The chairs of nine writing groups of the ANSI-predecessor working on methods for the measurement and rating of noise presented summaries of their tasks and their progress in 1968. If a symposium on the same theme were to be held today, there would be reports from fifty-four working groups of ANSI Accredited Standards Committees on Acoustics (S1), Bioacoustics (S3), and Noise (S12). These committees are cornerstones of the ASA standards program. As the first ASA Standards Director, Henning revitalized the Society's standards program to become the productive organization it is today. The progress in standardization in America during the past four decades is a tribute to his foresight and participation. He also contributed to international standardization as the leader of the U.S. delegation to ISO TC/43 (Acoustics) and its subcommittee on noise (SC1), and for three decades he chaired ISO TC/108/SC4 on human exposure to mechanical shock and vibration. Trained as an engineer, Henning was a leader of national and international standardization efforts to relate the mechanical energy in sound waves to the responses of the human organism.

8:20

2aNSb2. Henning E. von Gierke and human biodynamics. Anthony J. Brammer (Ergonomic Technology Center, Univ. of Connecticut Health Center, 263, Farmington Ave., Farmington, CT 06030, USA, brammer@uchc.edu)

In addition to his well-known involvement in protecting humans from exposure to excessive noise, Dr. Henning E. von Gierke was deeply involved in protecting humans from exposure to excessive mechanical shocks, impacts and vibration. From exploring the limits of survivability during the landing phase of space and aircraft capsules to the more everyday issues of occupational exposure to vibration, he provided scientific insight into problems and leadership in developing solutions. While most of Henning's contributions are buried in past literature, the spirit of his efforts lives on in a review chapter published in Harris' Shock and Vibration Handbook that I have had the privilege to co-author with him through two editions, now extending back more than a decade. The challenge of encompassing the original content of his contribution while introducing the results of more recent research has, so far, yielded a composition that he has endorsed. Henning's involvement and interests in human biodynamics will be illustrated by examples taken from this work.

8:40

2aNSb3. Service to his fellow men. Richard McKinley (AFRL, Wright-Patterson Air Force Base, Dayton, OH 45433-7901, USA, richard.mckinley@wpafb.af.mil)

Henning Von Gierke's life of service followed in the footsteps of his family legacy. This presentation will focus on Henning's 50 years of research at the Air Force Research Laboratory at Wright-Patterson Air Force Base, Dayton, Ohio and the recollections of Henning gathered from 20+ hours of video interviews. Henning's life and his research were dedicated to helping mankind. Beginning with his dissertation on noise produced by jet flow and continuing with the accomplishments of his division at the Air Force Research Laboratory his contributions to acoustics were outstanding. His contributions include the Air Force hearing conservation program, the EPA levels document, the BENOX report, the development of Bionics in the 1960's, car airbag research in the 60's, testing of the

original 7 US astronauts, human limits of survival for impact, human limits of performance in vibration, pioneering work in hearing protector modeling, bone conduction, and microwave hearing, speech communication, as well as his extensive work in American National Standards and international standards. His contributions have in one way or another affected nearly everyone. He was a great researcher and a great mentor. He will be missed by his colleagues at the Air Force Research Laboratory and around the world.

Contributed Paper

9:00

2aNSb4. Henning von Gierke, leader, consensus builder, mentor, and friend. Alice H. Suter (Alice Suter & Associates, 575 Dogwood Way, Ashland, OR 97520, USA, ahsuter@charter.net)

Over his long and illustrious career, Henning von Gierke has had an enormous effect on public policy in noise and vibration, not only because he was active in the field for more than half a century, but because of his intense dedication, his persistence, and his unique, almost legendary ability to build consensus. EPA's Office of Noise Abatement and Control could not possibly have produced its Criteria and Levels documents in response to the timeframe imposed by Congress without Henning's leadership. Those of us who were lucky enough to be mentored by him were able to accomplish more than we ever could have without his guidance. In addition to his invaluable assistance in my doctoral program, I knew him as ASA's Standards Director, then President of ASA, chairman of interagency task forces, chairman of several standards committees, and participant in numerous international activities. He was always willing to share the burden of work and to encourage the rest of us to forge ahead in an effort to make the world a quieter and healthier place.

Invited Papers

9:20

2aNSb5. Dr. Henning von Gierke—My mentor, everyone's mentor. Paul Schomer (Schomer and Associates, Inc., 2117 Robert Drive, Champaign, IL 61821, USA, schomer@SchomerAndAssociates.com)

Dr. Henning von Gierke was a leading force in acoustics for a majority of the 20th century. He led in human response to noise AND vibration and its assessment. He was a leading force in the US government and in the National Academy of Science (NAS) Committee on Hearing and Bioacoustics (CHABA). He was a leader in hearing assessment and conservation, and in Standards. And to many working in these various areas, he was a mentor, a confidant, and a friend. Mentoring did not stop at the doors of his laboratory, but rather, his mentoring extended to the world. He was a mentor to me in my first years after graduation when we developed the C-weighted procedure for the assessment of large-amplitude impulse noise, and in the development of related Standards and NAS studies. He thrust me into ISO work and CHABA studies, and, later, he chose me to succeed him as ISO delegation chair in acoustics and noise. Not enough can be said about Henning's impact on noise and vibration research and policy in the USA and in the world, and on the expanded reach of Henning's influence through the many like me that he mentored.

9:40

2aNSb6. Henning von Gierke's continuing contribution: Underwater whole-body vibration. Sarah Gourlie (Applied Research Laboratories, The University of Texas, P.O. Box 8029, Austin, TX 78713-8029, USA, sarah.gourlie@gmail.com), Mark F. Hamilton (Applied Research Laboratories, The University of Texas, P.O. Box 8029, Austin, TX 78713-8029, USA, hamilton@mail.utexas.edu)

Henning von Gierke was a leader in the characterization of human response to sound and vibration and in the development of international standards in this area. One focus of his research was using a multidisciplinary approach to create biodynamic models for whole-body vibration with the aim of predicting and preventing injury in vehicles. The focus of this presentation is whole-body vibration underwater with the aim of predicting discomfort or possibly injury to divers from low-frequency sonar. The frequency range of interest is 40-80 Hz, which encompasses the resonance of human lung. For this purpose, a biodynamic model developed by von Gierke to simulate thoracic, abdominal, and spinal responses to different vibrational excitations in air [H. E. von Gierke, J. Acoust. Soc. Am. 50, 1397 (1971)] is adapted for underwater conditions. It is assumed that the diver is neutrally buoyant and will therefore experience whole-body acceleration equal to the particle acceleration produced by the sound field in the absence of the diver. Emphasis is placed on determining the appropriate source distribution on the body as well as adjusting von Gierke's model to accommodate the decrease in lung compressibility with diver depth and mass loading on the chest. [Work supported by ONR.]

Session 2aNSc

Noise and EURONOISE: Physical and Psychophysical Evaluation of Vehicle Exterior Noise I

Paul Donavan, Cochair
Illingworth & Rodkin, Inc., 505 Petaluma Blvd. South, Petaluma, CA 94952, USA

Hugo Fastl, Cochair
AG Technische Akustik, MMK, TU München, Arcisstr. 21, München, 80333, Germany

Invited Papers

8:00

2aNSc1. Low speed exterior vehicle voise and the effect of pavement type. Paul Donavan (Illingworth & Rodkin, Inc., 505 Petaluma Blvd. South, Petaluma, CA 94952, USA, pdonavan@illingworthrodkin.com)

For operating conditions of cruise and moderate acceleration, the exterior noise emission of light vehicles is typically dominated by tire/pavement noise at speeds of 50 km/h or greater. At a test speed of 56 km/h, it has been found that pavement type can create a 10 dB or more variation in tire/pavement noise. This has significant implications for both community noise and vehicle noise emission testing. In this paper, the results tire/pavement noise measurements for over 40 different pavements in Europe and the United States are reported. These pavements include research surfaces, existing roadways, and ISO 10844 passby test surfaces. Measurements were conducted using an on-board sound intensity methodology that has been correlated to cruise-by noise levels. These results are discussed in terms of the revisions being considered for the newly revised ISO 362 passby test procedure and the ISO 10844 test surface specification. Additionally, a case history of community traffic noise reduction achieved by use of a quieter pavement is reviewed to demonstrate the importance of the pavement in low speed vehicle noise emissions.

8:20

2aNSc2. Noise source mapping for trucks, part 1: development and design. Kenneth J. Plotkin (Wyle Laboratories Inc., 241 18th Street S., Suite 701, Arlington, VA 22202, USA, kenneth.plotkin@wylelabs.com), Yuriy Gurovich (University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677, USA, yuriy.gurovich@wylelabs.com), William Blake (Naval Surface Warfare Center (Ret.), 6905 Hillmead Road, Bethesda, MD 20817, USA, hydroacoustics@aol.com), Paul Donavan (Illingworth & Rodkin, Inc., 505 Petaluma Blvd. South, Petaluma, CA 94952, USA, pdonavan@illingworthrodkin.com)

Mapping and quantifying noise sources on trucks under actual operating conditions on the road are important for traffic noise modeling and mitigation. The purpose of this study is to develop a practical truck noise source localization technique using acoustic beamforming. An experimental 70+ microphone elliptical array was designed and fabricated for truck testing. Beam-forming software was developed and implemented using a computerized data acquisition system. Proof-of-concept tests were performed at low-speed and high-speed truck testing facilities for a representative sample of trucks with widely different characteristics to validate the measurement system performance. The measurement system design parameters were verified experimentally, and certain improvements to the system were recommended for future implementation based on the field experience. The developed beam-forming measurement system provided adequate noise mapping and localization for various noise sources on trucks, stationary and moving with the speed up to 50 mph. The results of the proof-of-concept testing presented in an accompanying paper (Part 2) confirm that the developed microphone array, data acquisition system and beam-forming software performed generally as expected and required no major adjustments. This ongoing project is funded by the National Cooperative Highway Research Program of the Transportation Research Board of the National Academies, USA.

8:40

2aNSc3. Noise source mapping for trucks, Part 2: Experimental results. William Blake (Naval Surface Warfare Center (Ret.), 6905 Hillmead Road, Bethesda, MD 20817, USA, hydroacoustics@aol.com), Kenneth J. Plotkin (Wyle Laboratories Inc., 241 18th Street S., Suite 701, Arlington, VA 22202, USA, kenneth.plotkin@wylelabs.com), Yuriy Gurovich (University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677, USA, yuriy.gurovich@wylelabs.com), Paul Donovan (505 Petaluma Blvd, South, Petaluma, VA 94952, USA, pdonovan@illingworthrodkin.com)

The elliptical array described in the previous paper (Part 1) was deployed road-side at a test track for measuring the sound emitted from trucks during passby. Measurements were made on various truck models in various operating states with and without trailers. Extensive spherical-source calibrations confirmed the array's beamforming at both on-normal and steering angles (to 45°); they also disclosed the magnitudes of ground reflection paths for localized sources at different elevations. In all cases, the array output was computationally inverted to produce two-dimensional spatial maps of source levels in the side profile of the truck. In the cases of stationary trucks, the source maps were correlated with simultaneously-obtained intensity maps. Comparisons verified that the array-based source maps for the trucks ranked sound sources of disparate levels in the same order as did the sound power levels deduced from the sound intensity data. Acoustic source maps obtained during truck passbys were then used to provide time-histories and spatial distributions of sources and source paths from the engine, muffler, tires, and certain body components. This project is funded by the National Cooperative Highway Research Program of the Transportation Research Board of the National Academies, USA.

9:00

2aNSc4. Vehicle exterior noise from the view point of new experts. Brigitte Schulte-Fortkamp (TU Berlin, Institute of Fluid Mechanics and Engineering Acoustics, Einsteinufer 25, Sekr. TA 7, D-10587 Berlin, Germany, brigitte.schulte-fortkamp@tu-berlin.de)

The combination of physical and psycho- acoustical measurements with scientific evaluation of perceptual responses to environmental noise, known as Soundscaping, is an essential method for assessing and actualizing environments. Empirically, the perception of sounds and their evaluation will be explored under the premise of combining human judgment and physical factors. In a recent Soundscape Project, environmental noise perception and evaluation in a defined urban area were investigated in detail. Classical measurements and open, narrative, but issue-centred interviews with residents were carried out, and those interviews allowed the interviewed persons to set their own focus on certain aspects concerning the development of a public space in Berlin. Method and procedure will be discussed in detail to introduce to the New Expert Concept. The Soundscape Project is a module of the Project ″Nauener Platz - Remodelling for Young and Old″ in the framework of the research program ″Experimental Housing and Urban Development (ExWoSt)″ of the ″Federal Ministry of Transport, Building, and Urban Affairs (BMVBS)″ by the ″Federal Office for Building and Regional Planning (BBR)″. It is related to the fields of research (ExWoSt) concerned with ″Innovation of Urban Neighbourhoods for Families and the Elderly″. The project executing organization is the Regional Office Berlin-Mitte.

9:20

2aNSc5. Significance of Psychoacoustic Aspects for the Evaluation of Vehicle Exterior Noise. Klaus Genuit (HEAD acoustics GmbH, Ebertstrasse 30a, 52134 Herzogenrath, Germany, klaus.genuit@head-acoustics.de)

For 25 years acoustic engineers have been using specific measurement technologies and analyses to determine sound quality of vehicle interior noise. However, with regard to exterior vehicle noise only standard measurements, e.g. the A-weighted sound pressure level (ISO 362), have been used and interpreted so far. Extensive knowledge and experiences concerning the evaluation of interior noise are still unutilized for an advanced assessment of vehicle exterior noise. Within EU research projects, such as SVEN or QCity, exterior noise was investigated, methods established and psychoacoustic analyses carried out. The relevance of vehicle exterior noise with respect to perceived product quality was underestimated for a long time. Today, manufacturers recognize the importance of exterior noise beyond the annoyance perspective. Vehicle exterior noise conveys an impression of product quality. By optimizing the product sound quality manufacturers can compete against others and meet increased requirements. Customers prefer vehicle sounds which match their expectations, which also applies to the exterior sound of a vehicle. The acoustical fingerprint of a vehicle - inside and outside - provides enormous marketing opportunities. This paper shows how the knowledge of sound quality analyses with regard to interior noise can be transferred to the evaluation of exterior vehicle noise.

9:40

2aNSc6. Rating the Dieselness of engine-sounds. Hugo Fastl (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, fastl@mmk.ei.tum.de), Bernhard Priewasser (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, bernhard@prie.de), Markus Fruhmann (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, Markus.Fruhmann@gmx.de), Herbert Finsterhölzl (BMW Group, Knorrstr. 147, 80788 München, Germany, herbert.finsterhoelzl@bmw.de)

Sounds of idling Diesel engines were rated according to their Dieselness with two different psychoacoustic methods. On the one hand, a yes/no procedure was used. Subjects were presented sounds of two seconds duration and had to answer by yes or no to the following question: Does this sound stem from a Diesel engine? Histograms were calculated and a hypothesis was put forward as follows: The more positive answers are given to a specific sound, the larger is the Dieselness of this sound. On the other hand, the method of paired comparison was used, applying a kind of Bradley-Terry-Luce (BTL) procedure. Results are discussed with respect to the psychophysical method used and possible differences in ratings by expert versus naive listeners. Finally, psychoacoustic data are compared to predictions of Dieselness calculated by four different algorithms.

10:00

2aNSc7. Perceptive temporal features of train passbys. Jonathan Terroir (Laboratoire MRTE - UCP, 5, mail Gay-Lussac, Neuville-sur-Oise, 95031 Cergy-Pontoise Cedex, France, jterroir@u-cergy.fr), Catherine Lavandier (Laboratoire MRTE - UCP, 5, mail Gay-Lussac, Neuville-sur-Oise, 95031 Cergy-Pontoise Cedex, France, catherine.lavandier@u-cergy.fr)

This work deals with characterization of temporal features linked to train passages which may act upon annoyance. Two varying parameters in the recordings have been selected: distance from the railway (7.5 meters, 50 and 100 meters) and train category (four types of French trains). A first series of psychoacoustic tests are carried out in order to mainly extract temporal aspects which could be important for sound perception. To avoid the level saliency, a preliminary equalization has been made on L_{A5}. The selection of perceptive factors is carried out through a multidimensional analysis (INDSCAL). Dissimilarity tests highlight several dimensions: train category, slope of the level increase linked to train arrival and passage length. In addition to the dissimilarity between two stimuli estimation, subjects have to choose the one they prefer and explain their answers. The verbalization task helps illustrate multidimensional analysis axis. A second series of tests, designed with recordings varying on two independent parameters (the level and the slope of the temporal evolution) makes it possible to compare perceptive variations on temporal effects to loudness.

10:20

2aNSc8. Synthesis of pass-by railway noise. Shafiquzzaman Khan (Kungliga Tekniska Hogsklan, Department of Aeronautical and Vehicle Engineering, SE - 10044 Stockholm, Sweden, Shafik@kth.se), Mohite Ulhas (Indian Institute of Technology Roorkee, Graduta Student, Dept. of Mech. & Ind. Engg, 247667 Roorkee, India, goelvfme@iitr.ernet.in), Virendra Goel (Indian Institute of Technology Roorkee, Graduta Student, Dept. of Mech. & Ind. Engg, 247667 Roorkee, India, goelvfme@iitr.ernet.in)

Pass-by railway noise is one of the main problems in the community. Typically pass-by railway noise is estimated using Leq or Lden in dB(A) and often these methods do not reflect the listener perceptual overview. It is therefore necessary to reduce the pass-by railway noise by synthesizing the noise characters into various segments. The segments could be based on the physical properties of rail vehicles like rolling, traction, pantograph, cooling system noise, etc. There are few other segments which are also based on the perceptual view of the pass-by railway noise like squeaking, rattling or beating (tadak, tadak). In this study only a few of the physical characteristics of the railway noises are examined. The idea of synthesization is to determine what segments in the noise characters are annoying. And thereby make a virtual pleasant pass-by railway sound. Several pass-by noises from Swedish rail vehicles were recorded using binaural technology according to ISO 3095. Recorded noise signals were then analyzed with the help of spectrogram analysis using Matlab. The results of the studies indicate that the rolling noise and broadband noise are most annoying for the long distance passenger rail vehicles.

TUESDAY MORNING, 1 JULY 2008 ROOM 251, 10:20 A.M. TO 4:00 P.M.

Session 2aNSd

Noise and EURONOISE: Environmental Noise Mapping II

Kenneth Cunefare, Cochair
Woodruff School of Mechanical Engineering, The Georgia Inst. of Technol., Atlanta, GA 30332-0405, USA

Gaetano Licitra, Cochair
ARPAT - Dept. Firenze, Via Porpora, 22, Firenze, 50144, Italy

Invited Papers

10:20

2aNSd1. Managing Risk by Utilising an Integrated Approach to Quality Assurance During Strategic Noise Mapping. Simon J. Shilton (Acustica Ltd, Trident One, Styal Road, M22 5XB Manchester, UK, simon.shilton@acustica.co.uk), Alan Štimac (DARH2 Acoustics & Civil Eng. Ltd., Ljubicin prolaz 3, HR-10430 Samobor, Croatia, alan@darh2.hr), James Trow (Hepworth Acoustics Ltd, 5 Bankside, Crosfield Street, WA1 1UP Warrington, UK, james.trow@hepworth-acoustics.co.uk), Nathan Archer (Hepworth Acoustics Ltd, 5 Bankside, Crosfield Street, WA1 1UP Warrington, UK, nathan.archer@hepworth-acoustics.co.uk), Vincent Hii Jiu Ta (Hepworth Acoustics Ltd, 5 Bankside, Crosfield Street, WA1 1UP Warrington, UK, vincent.hii@hepworth-acoustics.co.uk), Nigel Jones (Extrium Ltd, Calverley House, 55 Calverley Road, TN1 2TU Tunbridge Wells, UK, nigel.jones@extrium.co.uk)

With the proclamation of the Directive on Environmental Noise (2002/49), the process of noise mapping and action planning has begun around Europe. As the development of strategic noise maps is arguably a new experience for many end-users, the first round of noise mapping could potentially lead to uncertainties within many aspects of the process. Noise maps represent a baseline for the implementation of noise management systems and any systematic errors within the noise maps could reduce the effective implementation of the whole management system. The extensive range of data inputs required in strategic noise mapping are usually obtained from different data sources or base datasets and as such, cataloguing with the use of metadata is key. The implementation of a quality assurance system is imperative to maintain consistency between technicians and within multi-disciplinary teams. It is also essential to ensure the control of processes and the ability to review inputs, intermediaries and deliverables. A similar approach may also be applied to calculation and post processing of noise levels. This paper presents collective experience of the implementation of quality assurance procedures used in several EU countries during the successful completion of projects within the first round of mapping.

10:40

2aNSd2. A methodology for mapping neighbourhood impacts. Ronny Klaeboe (Institute of Transport Economics, Gaustadalleen 21, 0349 Oslo, Norway, rk@toi.no), Erik Engelien (Statistics Norway, Oterveien 23, N-2225, 2225 Kongsvinger, Norway, erik.engelien@ssb.no), Margrete Steinnes (Statistics Norway, Oterveien 23, N-2225, 2225 Kongsvinger, Norway, margrete.steinnes @ssb.no)

A methodology for the spatial analysis of environmental impacts in urban areas and social gradients is presented. The methodology is computational intensive and requires access to geographically located data at the dwelling level. Neighbourhood information is harvested to provide contextual information, and a spatial smoothing technique used to provide powerful visualisations of residential and neighbourhood impacts. By focussing on impacts rather than exposures, the construction of accumulative non-monetarised and monetarised indicators of impacts such as disability adjusted life years (DALY) and health costs is facilitated. Since the residential dwelling location is, or can be made, the focal point for social, dwelling, environment, neighbourhood, and transportation related indicators, a rich network of information in a uniform format is available for construction of novel indicators. By focussing on neighbourhoods rather

than individual dwellings, privacy is enhanced while maintaining useful level of detail for involving the public and politicians in local planning. By combining information on social status and environmental exposures, questions on whether transportation related impacts are distributed fairly and whether social inequity is increased or decreases as a result of a particular measure can be addressed. The methodology is not applicable to rural areas and low probability events.

11:00

2aNSd3. Noise emissions at intersections: comparing microscopic and macroscopic traffic simulation approaches. Bert De Coensel (Ghent University - Department of Information Technology, Sint-Pietersnieuwstraat 41, 9000 Ghent, Belgium, bert.decoensel@intec.ugent.be), Dick Botteldooren (University Ghent - Department Information Technology, Sint-Pietersnieuwstraat 41, 9000 Gent, Belgium, dick.botteldooren@intec.ugent.be), Luc Dekoninck (Ghent University - Department of Information Technology, Sint-Pietersnieuwstraat 41, 9000 Ghent, Belgium, luc.dekoninck@intec.ugent.be), Dominique Gillis (Ghent University - Department of Civil Engineering, Krijgslaan 281 S8, 9000 Ghent, Belgium, dominique.gillis@ugent.be), Dirk Lauwers (Ghent University - Department of Civil Engineering, Krijgslaan 281 S8, 9000 Ghent, Belgium, dirk.lauwers@ugent.be)

Environmental noise mapping often involves the use of traffic simulation software. In most cases, traffic flows are simulated in a macroscopic way, considering only traffic flow parameters averaged over road segments. This approach does not allow to correctly account for the typical deceleration and acceleration patterns of traffic at intersections, which influence local noise emissions. Microscopic traffic simulation models, which model each vehicle individually, incorporate these dynamic effects. However, they require much larger amounts of input data for calibration, and are therefore less suited to map large areas. In this paper, we investigate the possible influence of the choice of simulation detail on simulated traffic flow parameters and, as a consequence, on noise emissions near intersections. As a case study, several types of road intersections under various traffic demand and traffic composition conditions are considered. Simulation results are obtained using a microscopic (Paramics) as well as a mesoscopic (MaDAM) and a macroscopic (OmniTRANS) traffic simulation model, and predicted traffic flow parameters and noise emission values are compared between these approaches.

Contributed Papers

11:20

2aNSd4. Environmental Noise Directive - results of noise mapping in Germany. Matthias Hintzsche (Federal Environment Agency, Woerlitzer Platz 1, 06844 Dessau-Rosslau, Germany, matthias.hintzsche@uba.de)

The Environmental Noise Directive intends the development of a concept to avoid, prevent or reduce on a prioritised basis the harmful effects, including annoyance, due to exposure to environmental noise and to inform the public on environmental noise and its effects. In a first step noise maps were made for all agglomerations and for all major roads, major railways and major airports. This noise maps are the basics for noise action plans. These plans have the target to prevent and to reduce environmental noise where necessary to preserve environmental noise quality where it is good. The results of the first round for noise mapping in Germany will be presented.

11:40

2aNSd5. Noise mapping of public roads in Norway: New calculation tool based on the Nord2000 Road Engineering Method. Ingunn Milford (Norwegian Public Roads Administration, Brynsengfaret 6A, P.O .Box 8142 Dep., N-0033 Oslo, Norway, ingunn.milford@vegvesen.no)

To comply with requirements in the Directive 2002/49/EC on environmental noise (END) and to national guidelines on land use and transport noise, the Norwegian Public Roads Administration has developed the calculation tool called NorStoy, which is composed of two main modules: 1. A calculation module with the Nord2000 Road Engineering Method algorithms. 2. An ArcGIS module to arrange the input data, delimit the calculation area, locate the calculation points and report the results. Nord2000 Road Engineering Model was completed in 2006, in cooperation with the Nordic countries. The model has very much in common with the coming joint European Harmonoise model. The challenge in using these models is to obtain an acceptable calculation time. To prevent NorStoy from stopping every time insufficient data occurs, we use default values based on either a typical mean value or the nearest registered value. At this time preparing input data is quite time consuming. We are able to produce the noise maps for the EC directive, but NorStoy is not yet an efficient tool for noise mapping in general. The development of NorStoy is still in progress.

12:00

2aNSd6. The national noise map of the Netherlands A trend in exposure since 1990. Rik Van Haaren (DHV, P.O. box 1132, 3800BC Utrecht, Netherlands, rik.vanhaaren@dhv.com)

Noise maps according to the environmental noise directive have been provided by several actors in the Netherlands. The results of the noise maps show the significance of each of the source types, as well as the distribution within and outside agglomerations. In the Netherlands 2.7 million people are exposed to noise levels above 55 dB(A). In some cases up to 4000 people are exposed to level higher than 75 dB(A). In the night period 2.3 million people are exposed to noise levels higher than 50 dB(A). Road noise is the dominant source for annoyance and sleep disturbance. Within agglomerations road noise is responsible for 88% of the highly annoyed population. Major roads, outside agglomerations, contribute less than 10% to the total annoyance due to road traffic. Railways are a second source in producing annoyance and sleep disturbance. 57% of the high annoyance due to railways is produced outside agglomerations. The results of the noise maps for road traffic noise have been compared with results for a selection of cities in 1990. The results show that the total number of exposed people has increased. However, the exposition to the higher bands has decreased.

12:20

2aNSd7. Noise Mapping and Noise Scoring - Software Techniques and Result Presentation. Wolfgang Probst (DataKustik GmbH, Gewerbering 5, 86926 Greifenberg, Germany, wolfgang.probst@datakustik.com)

Noise scoring is based on noise level maps and on exposure distribution. Different techniques can be used to develop the areal distribution of these values and to present them as coloured maps. Especially more complex ratings that are not only based on an exposure level and on population densities need relatively complex procedures to develop them from noise maps. It must be taken into account that there are two important steps in this procedure: One is the calculation of the noise score itself and the other is the presentation as coloured map to detect and focus on Hot Spots. Different techniques have been developed, were integrated in the software and have been applied in the frame of the QCity project. The consequences of different parameter settings are presented and discussed. The second step is the assignment of colours to a given scale of result values and even this simple process must be handled carefully to transport the needed information.

12:40-2:00 Lunch Break

Contributed Papers

2:00

2aNSd8. The Milan agglomeration Strategic Noise Map. Giovanni Zambon (Department of Environmental Sciences of the University of Milano - Bicocca, Piazza della Scienza, 1, 20126 Milan, Italy, giovanni.zambon@unimib.it), Simone Radaelli (Department of Environmental Sciences of the University of Milano - Bicocca, Piazza della Scienza, 1, 20126 Milan, Italy, s.radaelli@unimib.it)

The aim of this paper is to describe the procedure used for the Milan agglomeration Strategic Noise Mapping, as defined by Environmental Noise Directive (2002/49/EC). City of Milan is a complex case study because of the presence of a plurality of sources: road traffic, railway traffic (also tram traffic) and aircraft noise (Linate airport). The assessment method is based on the guidelines contained in the ″Good Practice Guide for Strategic Noise Mapping and the Production of Associated Data on Noise Exposure″ (WG-AEN). The study of noise generated from several sources was carried out with the integrated use of both numerical models and Geographic Information Systems (GIS). Numerical models allow us to estimate of noise levels in large areas near to a specific noise source and GIS enable the efficient acquisition, management and elaboration of geo-referenced data representative of territory, sources and buildings. By means of the study of specific GIS application it was possible to predict the global noise exposure for overall sources, by overlapping the noise maps of the single noise sources.

2:20

2aNSd9. Budapest Noise Mapping Project I. - Experiences. András Muntag (Enviroplus Kft., Telepy u. 3., 1096 Budapest, Hungary, muntaga@zajan.hu), Mihály Berndt (Enviroplus Kft., Telepy u. 3., 1096 Budapest, Hungary, berndt@mgx.hu)

The Hungarian capital Budapest, and their agglomeration have recently finished their first strategic noise map. According to the Hungarian noise regulations and to the European Directive 2002/49/EC, this is a legal obligation. The main part of the costs was supported by the EU. The preparation of the strategic noise maps based on a computer model and describing the present noise situation of the municipalities, should be seen as the first step to build a true noise information and management system, based on the computer model initially developed. The main parameters of the project were the planned area 1100 qkm, number of dwellings 440 thousand, length of the mapped roads ca. 2000 km This paper based on the experience of 23 municipal noise maps - Budapest and other 22 settlements in the agglomeration. We successfully finished the project on time - the results (all maps) can be found on the internet-page: http://terkep.budapest .hu/website/zajterkep The first paper deals with the technical experiences of the project.

2:40

2aNSd10. Budapest Noise Mapping Project II. - Results. Mihály Berndt (Enviroplus Kft., Telepy u. 3., 1096 Budapest, Hungary, berndt@mgx.hu), András Muntag (Enviroplus Kft., Telepy u. 3., 1096 Budapest, Hungary, muntaga@zajan.hu)

The Hungarian capital Budapest, and their agglomeration have recently finished their first strategic noise map. According to the Hungarian noise regulations and to the European Directive 2002/49/EC, this is a legal obligation. The main part of the costs was supported by the EU. The preparation of the strategic noise maps based on a computer model and describing the present noise situation of the municipalities, should be seen as the first step to build a true noise information and management system, based on the computer model initially developed. The main parameters of the project were the planned area 1100 qkm, number of dwellings 440 thousand, length of the mapped roads ca. 2000 km This paper is based on the experience of 23 municipal noise maps - Budapest and other 22 settlements in the agglomeration. We successfully finished the project on time - the results (all maps) can be found on the internet-page: http://terkep.budapest .hu/website/zajterkep The second paper shows the results of the project.

3:00

2aNSd11. The Port of Livorno noise mapping experience. Mario Morretta (Sintesis - Enginering and Consulting S.r.l., via Martin Luther King, 15, 57128 Livorno, Italy, mario.morretta@sintesis.toscana.it), Andrea Iacoponi (ARPAT - Dipartimento di Livorno, via Marradi, 114, 57128 Livorno, Italy, a.iacoponi@arpat.toscana.it), Fabrizio Dolinich (Sintesis - Enginering and Consulting S.r.l., via Martin Luther King, 15, 57128 Livorno, Italy, fabrizio.dolinich@gmail.com)

Livorno Municipality and its seaport are requested to produce the strategic noise map according to the European Directive only in 2012. However Livorno Port Authority, following the principle of environmental sustainability and in coherence with its EMAS status, started the investigation about the interaction between the noise generated by port activities and the nearby city. This analysis has been carried out within the Life funded project NoMEPorts, whose goal is to develop and provide tools and expertise to European ports for implementing their strategic noise maps, by means of a collaborative mapping of 8 Port Authority technical representatives from all around Europe. The work presented outlines in detail the process followed in performing the noise mapping of the Livorno port area and the port-city interface area, focusing in the problems encountered in the data collection, in port noise model building steps and in the consequent solutions adopted. The paper also talks about the result obtained by showing the noise maps in six detail levels and reports about the action plan proposed. As a result this experience emphasises the importance of noise maps as a powerful tool for decision-makers in port planning.

3:20

2aNSd12. Strategic Noise Mapping of the City of Ljubljana, Slovenia. Alan Štimac (DARH2 Acoustics & Civil Eng. Ltd., Ljubicin prolaz 3, HR-10430 Samobor, Croatia, alan@darh2.hr), Aleš Globevnik (A-PROJEKT Natasa Kepe-Globevnik s.p., Vinarje 110b, SI-2000 Maribor, Slovenia, aprojekt@siol.net), Dinko Stipanicev (DARH2 Acoustics & Civil Eng. Ltd., Ljubicin prolaz 3, HR-10430 Samobor, Croatia, dinko@darh2.hr), Lilijana Kuhelj (Environmental Agency of the Republic of Slovenia, Vojkova 1B, SI-1000 Ljubljana, Slovenia, lilijana.kuhelj@gov .si)

Currently Ljubljana is the only agglomeration according to the definition of the directive 2002/49/EC in Slovenia. For this agglomeration, separate noise maps not only for road and rail traffic but also for industrial sources were made. As for the first phase of preparation there is no precise methodology stated for representing industrial noise sources in strategic noise maps regarding END and following the vision of other EU countries, only industrial plants classified as IPPC installations were taken into consideration of noise mapping. Short term noise measurements around industrial plants conducted according to ISO 1996 during previous three years were used as reverse engineering techniques for the determination of sound power levels of industrial plants. Elaboration of road and rail traffic noise maps required development of perfect acoustical model with 3D terrain modelling including all possible obstacles (barriers, walls, bridges, viaducts, etc.). For calibration of noise maps more then 30 measurements were performed following the standard DIN 45642. During all stages of elaborating noise maps, quality assurance procedures were performed with a view to estimate accuracy of the results. Strategic noise maps of the agglomeration performed at municipal level of Ljubljana will be used mainly for drawing up action plans and spatial planning of areas with different noise levels allowed.

3:40

2aNSd13. Mixed Industry and Traffic Noise Mapping. Céline Boutin (Acouphen Environnement, Campus de la DOUA, 66, BD Niels Bohr, BP 52132, 69603 Villeurbanne, France, celine.boutin @acouphen-environnement.com), Alexis Bigot (Acouphen Environnement, Campus de la DOUA, 66, BD Niels Bohr, BP 52132, 69603 Villeurbanne, France, alexis.bigot@acouphen-environnement.com)

The authors have a large experience on Noise Mapping of both transportation sources and industrial plants. This paper provides examples of Noise Maps for Communities of different sizes with road, railway and industrial sources. The difficulties to estimate Noise Emission from industrial sites are discussed. Recommandations are provided. A Methodology to create the Industry data base for a Noise Map with a large number of industrial sites in the territory is presented. It is shown how Noise Maps can be an attractive tool to be used in the Environmental Management System of the Plant Manager to communicate on the Plant environmental integration and Noise Abatement Plan.

TUESDAY MORNING, 1 JULY 2008 ROOM 252A, 11:00 A.M. TO 3:20 P.M.

Session 2aNSe

Noise and EURONOISE: General Topics in Noise I

Contributed Papers

11:00

2aNSe1. Comparison of International Standards for Measuring Sound Power in Tool-Machines. Jose A. Ballesteros (Universidad de Castilla-La Mancha, Campus Universitario, 16071 Cuenca, Spain, Josea.Ballesteros@uclm.es), Marcos D. Fernandez (Universidad de Castilla-La Mancha, Campus Universitario, 16071 Cuenca, Spain, Marcos.Fernandez@uclm.es), Samuel Quintana (Universidad de Castilla-La Mancha, Campus Universitario, 16071 Cuenca, Spain, Samuel.Quintana@uclm.es), Ivan Suarez (Universidad de Castilla-La Mancha, Campus Universitario, 16071 Cuenca, Spain, ivan.suarezcasal@gmail.com), Laura Rodriguez (Universidad de Castilla-La Mancha, Campus Universitario, 16071 Cuenca, Spain, Laura .Rodriguez@uclm.es)

Industrial noise is one of the most important contaminant agents in the industrialised countries due to its effects on health. Sound power is used in order to characterize the noise emitted by the machines because it does not depend on aspects like place or distance of the measurement. These measurements are handled by international standards like UNE-EN ISO 374x and UNE-EN ISO 9614-x. The advantages and disadvantages of each one of these standards have been evaluated after making a representative number of measurements in several types of tool-machines and then, a set of recommendations have been derived to choose the best standard depending on the machine that is going to be measured.

11:20

2aNSe2. Activated Resonance Systems as Silencers and Sound Absorbers. Philip Leistner (Fraunhofer IBP, Nobelstrasse 12, 70569 Stuttgart, Germany, philip.leistner@ibp.fraunhofer.de)

A great variety of classical resonance systems is used for sound absorption and attenuation in technical systems. Apart from other requirements such as minimum size, high reliability and low costs the acoustic efficiency can be fulfilled in a wide range. However, the performance is limited and focused on a certain frequency band. These restrictions can significantly be overcome by activation. A very clear system comprises a membrane (mass) and a rear air volume (spring) which can be assembled by using a conventional loudspeaker in front of a cassette. Even the 'passive' system is causing a respectable sound attenuation also at low frequencies. The activation is based on a microphone close to the membrane the signal (according to the exciting sound pressure) of which is linearly amplified and fed back to the loudspeaker. A lot of details have to be considered prior to the real applicability of such an activated resonator. But the remarkable benefits can be documented for numerous practical scenarios. Theoretical and experimental results are based on a sophisticated model of the resonant structure and its acoustic environment. Additionally, several practical challenges, e.g. the presences of turbulent air flow or high temperatures, can be solved by using specific modifications.

11:40

2aNSe3. Sound radiation of the end of cylindrical duct application on industrial stacks. Thierry Simoneau (Acoustique & Conseil, 17-19 rue des Grandes Terres, 92508 Rueil-Malmaison, France, ts@acoustique-conseil .com)

In order to determine the sound field emitted by the top end of an industrial stack, a study of the acoustical radiation of unbaffled semi-infinite cylindrical duct ends was conducted. Firstly, the main publications on the subject were studied. A calculation model of the radiation directivity of the end of a chimney, based on H. Levine and J. Schwinger's results for the fundamental mode and on the Geometrical Theory of Diffraction for the modes of superior order, was established. In order to validate the model, measurements on scaled model were made in an anechoic chamber. Tallying between theoretical and experimental results is very good, especially at low frequencies, where the number of radiating modes is low. Moreover, measurements on a real size industrial site have been made. They also show satisfying tallying between the theory and experimentation.

12:00

2aNSe4. Austrian investigation on the influence of sound leakage in noise reducing devices. Marco Conter (Arsenal Research, Giefinggasse 2, 1210 Vienna, Austria, marco.conter@arsenal.ac.at), Manfred Haider (Arsenal Research, Giefinggasse 2, 1210 Vienna, Austria, manfred .haider@arsenal.ac.at)

Noise barriers are the most widely used means for road traffic noise abatement. Sound absorption and sound insulation are the key properties for noise barrier elements. The standard method of determining these properties by reverberation room measurements has recently been complemented by an in-situ method following CEN/TS 1793-5, also known as Adrienne method. This method allows flexible assessment of the acoustic performance of noise barriers in almost arbitrary places by means of mobile measurement equipment. Using the possibility to perform in-situ measurements the authors have investigated the presence of sound leakage due to structural imperfections which limit the sound insulation performance of noise barriers. This paper summarizes the results of Austrian research on different kinds of barriers regarding the difference in sound insulation between measurements carried out in front of the supporting post (where leakage is likely to occur) and in the middle of the barrier. A statistical correlation of the data was also performed.

12:20

2aNSe5. A study of twenty-one cases of low-frequency noise complaints. Christian Sejer Pedersen (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark, cp@acoustics.aau.dk), Henrik Møller (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark, hm@acoustics.aau.dk), Kerstin Persson Waye (Dept. of Environ. Medicine, The Sahlgrenska Acad. of Gothenburg Univ., Box 414, 405 30 Gothenburg, Sweden, kerstin.persson-waye@amm.gu.se)

From 203 cases of low-frequency complaints a random selection of twenty-one previously unsolved cases were investigated. The main aim of the investigation was to answer the question whether the annoyance is caused by an external physical sound or by a physically non-existing sound, i.e. low-frequency tinnitus. Noise recordings were made in the homes of the complainants, and the complainants were exposed to these in blind test listening experiments. Furthermore, the low-frequency hearing function of the complainants was investigated, and characteristics of the annoying sound was matched. The results showed that some of the complainants are annoyed by a physical sound (20-180 Hz), while others suffer from low-frequency tinnitus (perceived frequency 40-100 Hz). Physical sound at frequencies below 20 Hz (infrasound) is not responsible for the annoyance - or at all audible - in any of the investigated cases, and none of the complainants has extraordinary hearing sensitivity at low frequencies. For comparable cases of low-frequency noise complaints in general, it is anticipated that physical sound is responsible in a substantial part of the cases, while low-frequency tinnitus is responsible in another substantial part of the cases.

12:40-2:00 Lunch Break

Contributed Papers

2:00

2aNSe6. Measuring low-frequency noise indoors. Steffen Pedersen (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark, stp@acoustics.aau.dk), Henrik Møller (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark, hm@acoustics.aau.dk), Kerstin Persson Waye (Dept. of Environ. Medicine, The Sahlgrenska Acad. of Gothenburg Univ., Box 414, 405 30 Gothenburg, Sweden, kerstin.persson-waye@amm.gu.se)

At low frequencies, the sound pressure level may vary 20-30 dB in a room due to standing waves. For assessment of annoyance, mainly areas with the highest occurring levels are relevant, since persons present in such areas are not helped by the existence of lower levels in other areas. The level that is exceeded in 10% of the volume of a room L_{10} is proposed as a rational and objective target for a measurement method. In Sweden and Denmark rules exist for measuring low-frequency noise indoors. The performance of these procedures was investigated in three rooms. The results from the Swedish method were close to the L_{10} target, but, due to a doubtful use of C-weighting in the scanning, it may give too low results in case of complex sounds. The Danish method was found to have a high risk of giving results substantially below the target, unless complainants can precisely appoint measurement positions, where the sound is loudest/most annoying - which they often cannot. An alternative method using measurements in four three-dimensional corners of the room is proposed. This easy and straightforward method seems to give reliable results close to the proposed target.

2:20

2aNSe7. The use of communication device in background noise. Esko Toppila (Finnish Institute of Occupational Health, P.O:Box 486, 33101 Tampere, Finland, esko.toppila@ttl.fi), Pekka Airre (FIOH, P.O.Box 486, 33101 Tampere, Finland, erkko.airo@ttl.fi), Pekka Olkinuora (FIOH, Topeliuksenkatu 41, 00250 Helsinki, Finland, pekka.olkinuora@ttl.fi)

Communication devices are used more and more often in industrial premises. We have evaluated how the users experience these device in a call center (N=41), in a low noise warehouse (N=67) and in a noisy (L>80 dB) warehouse (N=25). For all subjects the same questionnaire was sent. The questionnaire contained questions were about the quality of communications device and self-evaluated hearing symptoms. The self-evaluated speech intelligibility was lowest in the high background environments. Still in the call center five persons and five in the silent warehouse had great difficulties with speech intelligibility. All these persons evaluated that they have always difficulties when communicating with people. In the noisy environments this relationship was not found. The need to rise the voice was highest in the warehouses ad did not depend on the background noise. Tinnitus was more frequent among process industry workers. The results suggests that lowered hearing function cause problems in low background noise. The self-evaluated nuisance depends on the complexity of the communication. The quality of the communication channel may also play a vital role in the communication problems especially for workers with hearing problems.

2:40

2aNSe8. Phase spectral processing for improved time-domain soft microphone based noise estimation. Ioannis Paraskevas (T.E.I. Piraeus / Department of Electronics, 250, Thivon str., GR-12244 Athens-Aigaleo, Greece, paraskevas@env.aegean.gr), Maria Rangoussi (T.E.I. Piraeus / Department of Electronics, 250, Thivon str., GR-12244 Athens-Aigaleo, Greece, mariar@teipir.gr), Stylianos M. Potirakis (T.E.I. Piraeus / Department of Electronics, 250, Thivon str., GR-12244 Athens-Aigaleo, Greece, spoti@teipir.gr), Stylianos Savvaidis (T.E.I. Piraeus / Department of Electronics, 250, Thivon str., GR-12244 Athens-Aigaleo, Greece, ssavaid@teipir.gr)

'Soft Microphones' (SM) constitute a cost-effective, yet quality alternative to the multiple microphones measurement, in applications related to noise mapping. SM offer a solution of great interest to real field applications, e.g., industrial plants. The SM approach, proposed in previous work of the authors, is based on the estimation of the noise signal and the calculation of noise levels over a set of points within the space of interest. To this end, a novel, frequency domain method was introduced and verified in a real field, textile plant experiment, with satisfactory results. However, in order to expand the use of SM from accurate noise mapping to a full Active Noise Control application, it is necessary to obtain accurate noise signal estimates in the time rather than the frequency domain. Further research into the deconvolution step of the proposed method reveals that discontinuities, appearing across the phase spectrum of the estimated signals, cause ambiguities that affect the deconvolution process. We propose here the use of the Hartley transform phase spectrum, which conveys fewer discontinuities as compared to its Fourier Transform counterpart, while it allows for a discontinuities compensation scheme. Experimental results verify that phase spectrum preprocessing provides accurate time domain signal estimates.

3:00

2aNSe9. Research into the improvement of the management of helicopter noise in the UK. David C. Waddington (Acoustics Research Centre, School of Computing, Science & Engineering, University of Salford, M5 4WT Salford, UK, d.c.waddington@salford.ac.uk), Paul Kendrick (Acoustics Research Centre, School of Computing, Science & Engineering, University of Salford, M5 4WT Salford, UK, p.kendrick@salford.ac.uk), Geoff Kerry (Acoustics Research Centre, School of Computing, Science & Engineering, University of Salford, M5 4WT Salford, UK, g.kerry@salford.ac.uk), Matthew Muirhead (QinetiQ Ltd, Cody Technology Park, Ively Road, GU14 0LX Farnborough, UK, mmuirhead@qinetiq.com), Ray Browne (QinetiQ Ltd, Cody Technology Park, Ively Road, GU14 0LX Farnborough, UK, rwbrowne@qinetiq.com)

Helicopter noise has a negative impact on the quality of life for many people. Effected populations are not just those living close to heliports, but include those exposed to noise from helicopters used by emergency services, the military, and commercial companies. One problem identified in the UK is that it is often difficult to complain about helicopter noise, since it is un-

clear which organisation is responsible for dealing with the complaint. Consequently, the Department for Environment, Food and Rural Affairs (Defra, UK) has commissioned research to summarise the following: i. the nature and extent of the concern about helicopter noise in the UK. ii. rules and regulations governing operations. iii. existing procedures for handling complaints. This stage of the project will produce a detailed report into the improvement of the management of helicopter noise. Also to be produced is a short non-technical guide including the means of redress for perceived disturbance. This paper will report on the findings of this project. [Work funded by Defra, UK]

TUESDAY MORNING, 1 JULY 2008 AMPHI HAVANE, 8:00 A.M. TO 12:40 P.M.

Session 2aPAa

Physical Acoustics: Acoustics of Porous Media II

Walter Lauriks, Cochair
Lab. ATF, Katholieke Universiteit Leuven, Celestijnenlaan 200D, Leuven, B-3001, Belgium

Keith Attenborough, Cochair
Open University, Department of Design, Development, Materials and Environment, Walton Hall, Milton Keynes, MK7 6AA, UK

Contributed Papers

8:00

2aPAa1. Sensitivity analysis and non linear parameters estimation of porous materials from normal sound coefficient absorption measurements. Mohammed Garoum (Ecole Supérieure de Technologie de Salé, Laboratoire Energétique Matériaux et Environnement, Avenue du Prince Héritier, BP 227, 10000 Salé, Morocco, garoum1@yahoo.fr), Rachida Idchabani (Ecole Supérieure de Technologie de Salé, Avenue du Prince Héritier, BP 227, Salé, Morocco, idchabani@mcinet.gov.ma), Mohammed Tajayouti (Ecole Supérieure de Technologie de Salé, Avenue du Prince Héritier, BP 227, Salé, Morocco, mtajayouti@gmail.com), Mohammed Rhachi (Ecole Supérieure de Technologie de Salé, Laboratoire Energétique Matériaux et Environnement, Avenue du Prince Héritier, BP 227, 10000 Salé, Morocco, mrhachi@hotmail.com), Antonio Moreno (Instituto de Acústica, CSIC, 144 Callé Serrano, 28006 Madrid, Spain, amoreno@ia.cetef.csic.es)

In literature, various models have been proposed in order to predict characteristic impedance Zc and wave number kc of porous materials. To use these models their involved nonacoustical parameters (i. e. flow resistivity, tortuosity porosityœ) are usually measured. Unfortunately measurements require specialised equipments and are often difficult on loose porous materials leading most often to erroneous values. Moreover, the inverse estimation of theses parameters from experimental data is not an easy task as believed. This is due to the non linearity of models and the presence of some parameters with weak and/or linearly dependent influences. In this work an approach based on the tradeoff between magnitude and linear independence of the sensitivity matrix components is presented in order to achieve an efficient ranking of parameters according to their inherent ease of estimation. Next this approach is associated to Genetic Algorithms to minimize the least squares norm between measured and modelled normal sound absorption coefficient. For five widely used models in literatures, the application of the proposed inverse estimation procedure on loose granular cork and vegetable fibres, shows that, unlike classical gradient approaches methods, better estimation has been obtained.

8:20

2aPAa2. Characterization of an acoustic ceramic liner in a subsonic flow by mean of Laser Doppler Velocimetry. Romain Pethieu (ONERA, 2, av. Edouard Belin - BP 4025 - Cedex 4, 31055 Toulouse, France, romain .pethieu@onecert.fr)

As aircraft traffic constantly increases, serious efforts are made to reduce engine noise. Among them, the design of high performance absorbing materials (liners) placed on the nacelle's internal walls is an exciting challenge. Optimizing these media requires to know the mechanisms of acoustical propagation near them in the conditions encountered in jet engines: high speed hot shear flows. Particular interest is being granted to the influence of the grazing flow in contact with the liners. This is due to its known effects on the liners characteristics: efficiency of absorption and frequency resonance. This influence is remarkable for those configurations based on perforated plates. Therefore, a testing method based on non-intrusive measurements has been developed at ONERA. This technique consists in measuring acoustic velocities by laser Doppler velocimetry to determine acoustic quantities under grazing turbulent high-speed flow, thanks to an Eulerian-Lagrangian description of the perturbations. Experimental activity is performed on a honeycomb material, via the achievement of acoustical pressure and intensity fields, for different Mach numbers. Moreover, these measurements bring into evidence a coupling between acoustical modes and hydrodynamic modes (from Kelvin-Helmholtz instability) which modify the material acoustical behaviour.

8:40

2aPAa3. The effect of mechanical elasticity on the surface impedance of a organic/inorganic composite aerogel. Winny Dong (California State Polytechnic University, 3801 West Temple Avenue, Chemical and Materials Engineering, Pomona, CA 91768, USA, winnydong@csupomona.edu), Wayland Dong (Veneklasen Associates, 1711 Sixteenth Street, Santa Monica, CA 90404, USA, wdong@veneklasen.com), Tanya Faltens (California State Polytechnic University, 3801 West Temple Avenue, Chemical and Materials Engineering, Pomona, CA 91768, USA, tafaltens@csupomona.edu), Elizabeth Scott (California State Polytechnic University, 3801 West Temple Avenue, Chemical and Materials Engineering, Pomona, CA 91768, USA, escott@csupomona.edu), Travis Thompson (California State Polytechnic University, 3801 West Temple Avenue, Chemical and Materials Engineering, Pomona, CA 91768, USA, trthompson@csupomona.edu)

Previous reports on the acoustical properties of monolithic silica aerogels have indicated that although the material's high porosity, pore tortuosity, and surface area contribute to a low sound velocity in the material, the rigid matrix of the aerogel exhibit a high reflection coefficient. Work by Forest et al. and others have shown that using granules instead of silica aerogel monoliths, the acoustical properties can be significantly enhanced. We report the acoustic properties of a polydimethylsilicate (PDMS)/silica composite aerogel that has both high surface area and a mechanically elastic matrix. Both monoliths and granules are studied. The surface impedance (at normal incidence) as a function of open porosity, pore size distribution, surface area, elasticity, and granule size will be reported.

9:00

2aPAa4. Numerical modeling of transient poroelastic waves in the low frequency range. Guillaume Chiavassa (Ecole Centrale Marseille MSNM-GP, Technopole de Chateau-Gombert, 13013 Marseille, France, guillaume.chiavassa@ec-marseille.fr), Bruno Lombard (Laboratoire de Mécanique et d'Acoustique LMA, 31 chemin Joseph Aiguier, 13402 Marseille, France, lombard@lma.cnrs-mrs.fr), Joel Piraux (Laboratoire de Mécanique et d'Acoustique LMA, 31 chemin Joseph Aiguier, 13402 Marseille, France, piraux@lma.cnrs-mrs.fr)

A numerical method is proposed to simulate the propagation of transient poroelastic waves across heterogeneous media, in the low frequency range. A velocity-stress formulation of Biot's equations is followed, leading to a first-order differential system. The latter is splitted in two parts: a propagative one discretized by a fourth-order ADER scheme, and a diffusive one solved analytically. Near sources and interfaces, a space-time mesh refinement is implemented to capture the small scales of evolution of the diffusive slow compressional wave. Lastly, an immersed interface method is implemented to accurately model the jump conditions at interfaces between the different media. Numerical experiments in one and two dimensions are shown, with porous/porous or fluid/porous interfaces. Comparisons with analytical solutions confirm the efficiency of the approach. [1] G. Chiavassa, B. Lombard, J. Piraux, Numerical modeling of 1-D transient poroelastic waves in the low-frequency range, soumis au J. Comput. Appl. Math., (2007), disponible sur http://hal.archives-ouvertes.fr/hal-00193103/fr/

9:20

2aPAa5. On the scattering of a plane wave by porous sound-absorbing strip. Alex De Bruijn (Independent Acoustical Consultant, Schonenvaarderstraat 18014, 7418 CC Deventer, Netherlands, alex.bruijn@planet.nl)

The analysis concerning the scattering of a plane wave by a porous sound-absorbing strip embedded in an infinite sound-hard surface is most relevant for the evaluation of measurements with respect to absorbing materials tested in a reverberation room. This diffraction phenomenon around the edges leads to an additional sound absorption, the so-called edge effect. This extra absorption can be analyzed and computed - among other well-known methods - by using a spatial Fourier field expansion just on the absorbing strip. The field in the space above the reflecting surface can be put into a contour integral of an angular spectrum in the complex plane or -in other words- a superposition of plane waves with complex angles of incidence. Matching both field representations leads to the solution of the unknown coefficients of the Fourier expansion and hence to the amplitudes of the total diffracted field. It would be interesting to compare these results with diffraction data obtained via an alternative technique: the Finite-Difference Time-Domain method using the direct Euler equations, also within the porous sound-absorbing material. The paper will show the data for the field amplitudes obtained by both methods and will discuss the agreement and differences.

9:40

2aPAa6. Analysis of porous plate/water layered structures by means of the transition terms method. Ferroudja Belhocine (LOMC FRE CNRS 3102, Université du Havre, Place Robert Schuman, 76610 le Havre, France, ferroudja.belhocine@univ-lehavre.fr), Serge Derible (LOMC FRE CNRS 3102, Université du Havre, Place Robert Schuman, 76610 le Havre, France, serge.derible@univ-lehavre.fr), Cole Franklin (1241 East dyer road, Santa Ana, CA 92672, USA, colesfranklin@yahoo.com)

This paper is devoted to the study of water-saturated porous plate/water layered structures by means of the transition terms defined from the reflection and transmission coefficients R and T as follows: $TT_{sym}=(1-(R+T))/(2i)$, $TT_{asym}=(1-(R-T))/(2i)$. They are the eigenvalues of the scattering matrix of the water-immersed structure and are directly connected with its symmetric or antisymmetric vibrations. The N porous plates associated in our structures obey Biot's theory which is first used to calculate the reflection and transmission coefficients of a unique water-saturated plate. An induction on N process allows to find the reflection and transmission coefficients of a given N plate/water-layer structure. The plates used in the experiments at normal incidence are 5mm thick. The reflection and transmission coefficients of sets of 1, 2, 3, and 4 water immersed plates, separated from each other by a 1cm water gap, are measured thanks to two wideband transducers with central frequency 0.5MHz. There are good agreements between the calculated and experimental transition terms which exhibit the symmetric and antisymmetric resonances of the structures. They obey the Breit-Wigner resonant form which characteristics can be obtained

10:00

2aPAa7. On the adsorption-desorption relaxation time of carbon in very narrow ducts. Timothy J. Mellow (Nokia, Nokia House, Summit Avenue, GU14 0NG Farnborough, UK, tim.mellow@nokia.com), Olga Umnova (University of Salford, Acoustics Research Centre, Newton Building, M5 4WT Salford, UK, o.umnova@salford.ac.uk), Konstantinos Drossos (University of Southampton: Institute of Sound and Vibration Research, University Road, Highfield, SO17 1BJ Southampton, UK, kd806@soton.ac.uk), Keith Holland (University of Southampton: Institute of Sound and Vibration Research, University Road, Highfield, SO17 1BJ Southampton, UK, krh@isvr.soton.ac.uk), Andrew Flewitt (University of Cambridge: Centre for Advanced Photonics and Electronics, 9, JJ Thomson Ave., CB3 0FA Cambridge, UK, ajf@eng.cam.ac.uk), Leo Kärkkäinen (Nokia Research Center, Itämerenkatu 11 - 13, 00180 Helsinki, Finland, leo.m.karkkainen@nokia.com)

Loudspeakers generally have boxes to prevent rear wave cancellation at low frequencies. However, the stiffness of the air in a small box reduces the diaphragm's excursion at low frequencies. Hence the size of the box is generally a compromise between low frequency performance and practicality. Activated carbon has been found to increase the apparent size of a given box through adsorption of the air molecules when the pressure increases and likewise desorption when it decreases. However, the exact viscous effects in the granular structure are difficult to model. Thus it is impossible determine the high frequency limit due to the natural adsorption/desorption relaxation time in the absence of viscous losses. In this study, a tube model is presented which takes into viscous and thermal losses with boundary slip together with adsorption. Impedance measurements are performed on an array of 12 million holes, each 2 micrometers in diameter, etched in a 0.5 mm thick silicon wafer so that the viscous and thermal losses can be verified against the model without adsorption. Impedance measurements are then performed on an array of holes coated with graphite in order to create an activated carbon-like structure, thus enabling the adsorption/desorption relaxation time to be evaluated.

10:20

2aPAa8. Acoustic pulse attenuation and transmission in rigid porous media: Experimental investigation and numerical simulations. Diego Turo (University of Salford, Acoustics Research Centre, Newton Building, M5 4WT Salford, UK, d.turo@pgr.salford.ac.uk), Olga Umnova (University of Salford, Acoustics Research Centre, Newton Building, M5 4WT Salford, UK, o.umnova@salford.ac.uk)

This paper presents an investigation on acoustic pulse attenuation and transmission in rigid porous media. Transmission measurements of acoustic pulses through porous material with a rigid frame were performed using an appropriate standing wave tube. A vertical tube was built to make measurements on granular materials. The influence of pulse duration, layer thickness and material microstructure on the transmission has been investigated. Numerical time domain simulations based on two different semi-empirical models with one and two viscous relaxation times, respectively, were performed to compare with existing data. First, material microstructure and pulse durations were adapted to investigate the influence of viscous and inertial effects on pulse propagation separately. The simultaneous contribution of both viscous and thermal effects was then investigated. In the finite difference time domain (FDTD) model, different approaches are used to correctly simulate the pulse transmission through a porous layer. Accuracy and computational time required for the numerical methodologies have been compared and their advantages and drawbacks applied to this particular case shown.

10:40-11:00 Break

11:00

2aPAa9. On the influence of the micro-geometry on sound propagation through periodic array of cylinders. Rodolfo Venegas (University of Salford, Acoustics Research Centre, Newton Building, M5 4WT Salford, UK, R.G.VenegasCastillo@pgr.salford.ac.uk), Olga Umnova (University of Salford, Acoustics Research Centre, Newton Building, M5 4WT Salford, UK, o.umnova@salford.ac.uk)

Sound propagation in rigid porous media has been widely studied by using macroscopical models. These models make use of averaged quantities in which the microscopic details of the porous media geometry are represented by macroscopical parameters and, in a certain way, the influence of the microscopic geometry is not directly identified. In this paper, homogenization theory and finite element method are used for solving the full microscopic dynamic flow and dynamic heat problems for a porous medium modelled as an idealized geometry consisting of a periodic array of cylinders. Different cross-section shapes of the cylinders (circular, ellipsoidal and square cross-section shapes) and a wide range of porosity values are considered. The influence of the microscopic features of the porous media on dynamic permeability and dynamic compressibility is also studied.

11:20

2aPAa10. Acoustic emission before avalanches in granular media. Vincent Gibiat (Université Paul Sabatier, PHASE, 118, route de Narbonne, 31062 Toulouse cedex 9, France, gibiat@cict.fr), Eric Plazza (Université Paul Sabatier, PHASE, 118, route de Narbonne, 31062 Toulouse cedex 9, France, ericvpp@hotmail.com), Pierre De Guibert (Université Paul Sabatier, PHASE, 118, route de Narbonne, 31062 Toulouse cedex 9, France, dguibert@cict.fr)

Avalanches of granular media are mainly characterized by the observation and the measurement of the main angles of avalanche corresponding first to the movement of isolated beads and to the whole movement of a great part of the grains. These characterisations do not give any information about the rearrangements of the grains inside the layer of granular beads. As any movement of a grain produces a deformation of the structure it is quite normal to expect for a sound that will propagate inside the granular medium. We present an experimental study of the precursors of avalanches on spherical granular glass beads and silica aerogels in powder (size of grains less than 80 micrometers). Acoustic emission has been recorded with two piezoelectric transducers placed on the lower part of the material layer. Our results show clearly that before any movement on the upper part of the beads layer, so for an angle less than the first angle of avalanche, movements inside the material produce pulsed sounds that can be recorded. Theses vibrating events are occurring more and more when the angle is increasing until the first angle of avalanche where acoustic emission becomes intense.

11:40

2aPAa11. On heterogeneous blankets: Analytical solution for the interaction between masses and poro-elastic layers. Kamal Idrisi (Virginia Tech, Mechanical Engineering, 143 Durham 0238, Blacksburg, VA 24061, USA, idrisi@vt.edu), Andreas Wagner (Virginia Tech, Mechanical Engineering, 143 Durham 0238, Blacksburg, VA 24061, USA, andreasw@vt.edu), Marty Johnson (Virginia Tech, Mechanical Engineering, 143 Durham 0238, Blacksburg, VA 24061, USA, martyj@vt.edu), David Bartylla (Virginia Tech, Mechanical Engineering, 143 Durham 0238, Blacksburg, VA 24061, USA, bartylla@vt.edu)

There has been substantial research over the last five decades on control of aircraft cabin noise. One new passive approach is the heterogeneous (HG) blanket where a traditional acoustic blanket treatment is altered by adding mass inhomogeneities into the poro-elastic/viscoelastic layers. These masses act like distributed vibration absorbers and can be used to reduce vibration and sound transmission by targeting modes of the fuselage. The natural frequency of a mass inhomogeneity is determined by the mass itself and by the effective stiffness of the porous layer due to the mass/poro interaction. An experimental 1st order approach to predict the effective stiffness based on the shape of the mass inhomogeneities is reviewed. An analytical model for poro-elastic media proposed by Allard et al. was simplified for low frequencies and is used to validate and extend the 1st order approach. It is shown that the effective stiffness depends on mass shape, the foam thickness and material constants such as the modulus of elasticity and Poisson's ratio. Furthermore the model can be used to calculate the stress and displacements fields in the blanket in order to give further insight into the behaviour of the HG blankets.

12:00

2aPAa12. Influence of boundary slip on the acoustical properties of microfibrous absorbents. Olga Umnova (University of Salford, Acoustics Research Centre, Newton Building, M5 4WT Salford, UK, o.umnova@salford.ac.uk), David Tsiklauri (University of Salford, Acoustics Research Centre, Newton Building, M5 4WT Salford, UK, d.tsiklauri@salford.ac.uk), Rodolfo Venegas (University of Salford, Acoustics Research Centre, Newton Building, M5 4WT Salford, UK, R.G .VenegasCastillo@pgr.salford.ac.uk)

In the past decades a variety of new highly porous materials with unusually small pores have been manufactured. In aerogels, for instance, pores can be less than 20 nm in diameter. The conventional models have to be modified when applied to describe acoustical properties of those materials. The non-slip condition on a pore surface is no longer valid and needs to be replaced by the Knudsen boundary condition. In attempt to provide an insight into the behaviour of microfibrous materials, an analytical model has been developed, which accounts for the boundary slip in a medium consisting of rigid parallel fibres assuming different directions of sound propagation with respect to fibres. It has been shown that the presence of the boundary slip leads to a significant change in model predictions. For instance, in a material with fibre radius 80 nm and 95% porosity the sound speed decreases and attenuation increases by more than 20% compared to the values obtained assuming no boundary slip. The effect is stronger for smaller size fibres, lower porosity values and for sound propagating parallel to fibres. Numerical computations have been performed to simulate oscillatory flow around the cylindrical fibres assuming Knudsen boundary conditions and the results have been compared with the analytical model predictions

12:20

2aPAa13. Time-Resolved Spectroscopy of Water confined in Vycor. Riccardo Cucini (European Lab. for Non-Linear Spectroscopy (LENS), Univ. di Firenze, via Nello Carrara 1, 50019 Sesto Fiorentino (Fi), Italy, cucini@lens.unifi.it), Andrea Taschin (European Lab. for Non-Linear Spectroscopy (LENS), Univ. di Firenze, via Nello Carrara 1, 50019 Sesto Fiorentino (Fi), Italy, taschin@lens.unifi.it), Paolo Bartolini (European Lab. for Non-Linear Spectroscopy (LENS), Univ. di Firenze, via Nello Carrara 1, 50019 Sesto Fiorentino (Fi), Italy, bart@lens.unifi.it), Renato Torre (European Lab. for Non-Linear Spectroscopy (LENS), Univ. di Firenze, via Nello Carrara 1, 50019 Sesto Fiorentino (Fi), Italy, torre@lens .unifi.it)

We present measurements on water confined in Vycor 7930 by heterodyne-detected transient grating experiments (HD-TG) [1]. We studied the acoustic wave propagation, the thermal diffusion and the liquid viscous flow through the pores as a function of temperature (-15 - 90 °C) and of the exchanged q-vector (0.63 - 2.5 μm^{-1}). The data show interesting aspects due partly to the confinement effects and partly to the peculiarities of bulk water. Acoustic results are compared with the predictions of the Biot theory finding a good agreement only for the sound velocity data, while the predictions for the acoustic wave attenuation is poor as just reported in [2]. Our technique is able to induce a thermal grating inside the sample. The analysis of the data requires an extension of the Biot theory, that takes into account the temperature variations. Surprisingly, our data show that the acoustic oscillation are not affected by the temperature. The induced thermal grating is also used to analyze the viscous flow of water inside the pores and the thermal diffusion. [1] Time-resolved spectroscopy of complex liquids, edited by Torre R. (Springer, New York) 2008 [2] A. Taschin, R. Cucini, P. Bartolini, R. Torre, Europhys. Lett., in press.

Session 2aPAb

Physical Acoustics: Diffraction of Waves on Periodical Structures: Acoustic, Ultrasonic, and Acousto-Optical Diffraction Phenomena I

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Invited Papers

8:00

2aPAb1. Spectral instruments based on acousto-optical tunable filters: advantages and prospects. Vladislav Pustovoit (Scientific Technological Center of Unique Instrumentation of RAS, ul.Butlerova, 15, 117342 Moscow, Russian Federation, np@ckbup.dol.ru), Vitold Pozhar (Scientific Technological Center of Unique Instrumentation of RAS, ul.Butlerova, 15, 117342 Moscow, Russian Federation, aoslab@ckbup.dol.ru)

A review of modern spectral techniques and instruments using acousto-optical tunable filters (AOTF) is presented. New approaches for spectrometers and spectral systems development are considered. The following instruments are described: a series of compact spectrometers and spectrophotometers of ultra-violet, visible, and infra-red ranges for Raman, fluorescence, and absorption spectroscopy for various applications, including out-of-door measurements; spectroradiometers for environment monitoring (open air, Earth surface, subsurface and depth seawater), specialised spectral equipment for industrial monitoring (microelectronics, alcoholometry), imaging spectral devices, time-resolve fluorescent spectrometers, and instruments for differential spectroscopy. Also some fundamental problems of AOTF-based spectroscopy are considered: spectrogram correction of instrument function distortions, ultimate spectral resolution of different methods, modulation techniques for instrument function synthesizing, optimization of measurement procedures, including adaptive algorithms. Basic trends of the past 30 years of AOTF-based spectroscopy development are formulated. The most promising ideas are discussed.

8:20

2aPAb2. Advances in acousto-optic devices based on frequency intermodulations suppression. Valeriy V. Proklov (IRE RAS, 11 Mokhovaya str., build.7, 125009 Moscow, Russian Federation, proklov@mail.cplire.ru), Sergey Antonov (IRE RAS, 11 Mokhovaya str., build.7, 125009 Moscow, Russian Federation, olga-ant@yandex.ru), Yury Rezvov (IRE RAS, 11 Mokhovaya str., build.7, 125009 Moscow, Russian Federation, rezvov@newmsk.tula.net), Alexander Vainer (IRE RAS, 11 Mokhovaya str., build.7, 125009 Moscow, Russian Federation, sawashi@yandex.ru)

The limitative influence of the frequency intermodulation effect on the most important parameters in a big family of acousto-optic (AO) devices is well known. The proposed presentation deals with new principles and related technical tools to perform significant suppression of the inherent intermodulation effects within broadband AO devices. The different AO interaction mechanisms in solids offering the intermodulations decrease have been considered. On this basis there were theoretically established and experimentally verified facilities to design a few types of advanced AO devices for a wideband signal processing with the extended spurious free dynamic range, as well as for a flexible laser beam forming with extremely high efficiency and etc.

Contributed Paper

8:40

2aPAb3. Periodically non-homogeneous acoustic beams and their application in acousto-optics. Vladimir I. Balakshy (Dept. of Physics, M.V.Lomonosov Moscow State Univ., Vorobyevy Gory, Bldg. 1, 119991 Moscow, Russian Federation, balakshy@phys.msu.ru), Bogumil B. Linde (Inst. of Exper. Physics, Univ. of Gdańsk, Ul. Wita Stwosza, Bldg. 57, 80-952 Gdańsk, Poland, fizbl@univ.gda.pl)

Light diffraction by ultrasonic waves is of great interest in the view of both fundamental physics and diverse applications. Acousto-optic devices are used for modulation of optical wave parameters, for optical information processing in real time, etc. In the given research, basic attention has been focused on the analysis of characteristics of anisotropic acousto-optic cells in which acoustic waves are excited by means of a phased array of transducers. Current technology makes it possible to create easily such transducers with antiphase excitation of adjacent elements of the array. The directional diagram of this composite transducer consists of several lobes whose spatial orientation varies with frequency. This peculiarity results in changing the Bragg condition. In the case of the anisotropic acousto-optic diffraction, every branch of the Bragg angle frequency dependence is split into several curves. This opens up new possibilities for optimization of acousto-optic devices. In this work, we have analyzed amplitude, frequency and angular characteristics of the phased transducer cells depending on crystal cut and transducer parameters. In particular, it has been shown that, in spite of a noticeable phase mismatch between interacting waves, the diffraction efficiency can approach 100%.

Invited Paper

9:00

2aPAb4. Differential properties of acousto-optical tunable filters in phase-modulation mode. Vitold Pozhar (Scientific Technological Center of Unique Instrumentation of RAS, ul.Butlerova, 15, 117342 Moscow, Russian Federation, aoslab@ckbup.dol.ru), Vladislav Pustovoit (Scientific Technological Center of Unique Instrumentation of RAS, ul.Butlerova, 15, 117342 Moscow, Russian Federation, np@ckbup.dol.ru), Sergey Beryoza (Scientific Technological Center of Unique Instrumentation of RAS, ul.Butlerova, 15, 117342 Moscow, Russian Federation, bruce_lee_22@mail.ru)

Tailoring the transmission function of acousto-optical tunable filters (AOTF) via acoustic waves control is one of the most challenging tasks, which is promising for optical spectroscopy applications. In our early work it was theoretically predicted and experimentally demonstrated that collinear AOTF with fast periodic phase manipulation exhibits differential properties, in particular capability of detection spectrum derivatives. In the report, the form of the instrument function of such AOTF was measured with use of Ne-lamp linear spectrum. Comparison to theoretically calculated function shows rather good agreement. Also the form of window of equivalent differential filter was determined and proved to be quite similar to AOTF classical window. Advantages of using this differential detection technique are considered. Its applicability to fluorescence spectroscopy is discussed.

Contributed Papers

9:20

2aPAb5. Quasicollinear acoustooptic tunable filters based on KDP single crystal. Vladimir Molchanov (Technological State Univ., Moscow Steel and Alloys Institute, Leninsky prospect, 4, 119049 Moscow, Russian Federation, v_molchanov@smtp.ru), Sergei Chizhikov (Technological State Univ., Moscow Steel and Alloys Institute, Leninsky prospect, 4, 119049 Moscow, Russian Federation, aocenter@mail.ru), Oleg Makarov (Molecular Technology GmbH, Rudower Chaussee 29-31, 12489 Berlin, Germany, moltech@mt-berlin.com)

The paper is devoted to the theoretical and experimental investigation of acoustooptical tunable filters, based on quasicollinear geometry of light-sound interaction in KDP single crystal. This geometry uses the effect of acoustic anisotropy in KDP, as well as peculiarities of acoustic wave reflections from the free boundary of KDP crystal. The mathematical approach for determination of optical and consructional characteristics of the filters is elaborated. Different types of quasicollinear filters are considered. The filters are intended for UV-VIS regions of optical radiation. The experimental acoustooptical filter was designed and investigated. Typical tunable spectral band is 220-500 nm, the spectral resolution doesn't exceed 0.1 nm in UV region of spectra. The experimental data are in a good agreement with the predicted one. This research is supported by RFBR grants # 07-02-12238 and # 07-02-01195.

9:40-10:00 Break

10:00

2aPAb6. Surface profile investigations by means of acousto-optic technique. Boris S. Gurevich (Scientific Instruments, Rizhsky prospekt 26, 190103 St. Petersburg, Russian Federation, bgurevich@mail.ru), Valentine V. Shapovalov (Intelligent Software Systems, Nevsky pr. 190, 193317 St. Petersburg, Russian Federation, Shapovalov@incomsys.ru), Sergey V. Andreyev (Scientific Instruments, Rizhsky prospekt 26, 190103 St. Petersburg, Russian Federation, svan51@mail.ru), Andrey V. Belyaev (Institute for analytical instrumentation RAS, Rizhsky pr. 26, 190103 St. Petersburg, Russian Federation, bgurevich@mail.ru), Ilya A. Kolesov (Intelligent Software Systems, Nevsky pr. 190, 193317 St. Petersburg, Russian Federation, ilkak@mail.ru)

Acousto-optic tunable filters (AOTF) represent a very strong tool in different kinds of investigations where high rate light wavelength variations are required. The rough surfaces profile investigation is among the problems which can be solved partially by this way. Our investigations have been connected with use of AOTF and lens components with strong chromatic aberration. One of the problems was to choose the z-distance resolving power criterion. The proposed criterion is defined by the admissible probability to miss information unit regarding the object characterization in z-distance. In our experiments we used the specially elaborated AOTF based on tellurium dioxide crystal with transmission bandwidth of several nm, depending on the central wavelength position. Also we have used the lenses fabricated with tellurium dioxide because this material provides high chromatic aberrations. The results of experiments are discussed from the point of view of the AOTF possibilities to increase the z-distance resolving power in comparison with existing systems. The 90%-probability distinguishing of defocusing while electric frequency variation of 200 kHz, can be considered as serious advantage. The device improvement can be attained by means of the noise level decreasing to the level taking place for electric frequency of 94 MHz.

10:20

2aPAb7. Transformation of Acoustic Modes in Case of Arbitrary Reflection in Acousto-Optic Crystals. Nataliya Polikarpova (M.V. Lomonosov Moscow State University, Department of Physics, Leninskie gory 1 bldg. 2, Division of Oscillations, 119991 Moscow, Russian Federation, polikarp@phys.msu.ru), Vitaly Voloshinov (M.V. Lomonosov Moscow State University, Faculty of Physics, Vorob'evy Gory, MSU, 119991 Moscow, Russian Federation, volosh@phys.msu.ru), Nico F. Declercq (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, nico .declercq@me.gatech.edu)

The phenomenon of unusual reflection of plane elastic waves is examined in the crystalline compounds of tellurium and mercury. It is predicted and registered in experiments that energy flow of one of the reflected waves may propagate practically in a back direction with respect to an incident energy flow. This phenomenon is observed in the materials with a strong anisotropy of elastic properties. In the case of inclined incidence, the reflection may take place strictly in the back direction with respect to the intrinsic energy flow. Moreover, all incident elastic energy may be reflected in a form of a back elastic wave. The analysis was carried out for all exisiting cases of inclined incidence in the crystals. It was proved that relative intensity of the unusually reflected wave might be close to a unit in a wide variety of incidence and cut angles in the crystals. A comparative analysis of the effect in new anisotropic materials and in the well-known crystals has been carried out. Since the crystals under investigation possess relatively high magnitudes of acousto-optic figure of merit, e.g. paratellurite and calomel, principal results of the research are of significance in development of new generations of acousto-optic devices.

10:40

2aPAb8. Optimization of KDP-based acousto-optic imaging filters. Konstantin B. Yushkov (Moscow State University, Phisical Faculty, Leninskie Gory, 119991 Moscow, Russian Federation, protonoa@mail.ru)

Acousto-optic devices can perform spatial and spectral tunable filtration of light with the use of anisotropic Bragg diffraction. In the near UV range

of spectrum, KDP single crystals provide the highest acousto-optic figure of merit among uniaxial crystals. The report is devoted to the problem of optimization of the filter parameters in order to maximize angle aperture of the instruments. It was found that the maximum angle aperture of wide angle tunable acousto-optic filters is equal to 3 degrees in air at output of the KDP crystals. This magnitude of the angle aperture is limited by a narrow deflection angle of light in KDP that is due to a relatively low birefringence of the material. The investigation showed that the diffraction efficiency of light depends on the dimensions of the acousto-optic cell, i.e. linear aperture and length of a piezoelectric transducer. A discussion on trade-off between transmission coefficient and the spatial resolution in the filter is presented in the report.

11:00

2aPAb9. Acousto-optic collinear diffraction of arbitrary polarized light. Sergey Mantsevich (Dept. of Physics, M.V.Lomonosov Moscow State Univ., Vorobyevy Gory, Bldg. 1, 119991 Moscow, Russian Federation, manboxx@mail.ru), Vladimir I. Balakshy (Dept. of Physics, M.V .Lomonosov Moscow State Univ., Vorobyevy Gory, Bldg. 1, 119991 Moscow, Russian Federation, balakshy@phys.msu.ru)

Acousto-optic collinear diffraction of light is a type of light scattering interesting as a physical phenomenon and important from practical point of view. This type of diffraction is used in tunable acousto-optic filters. In this case, the incident light has to be linearly polarized along one of the acousto-optic crystal anisotropy axes. If the incident light is not primordially polarized, a polarizer should be disposed at the input of the acousto-optic cell. Unfortunately, this leads to loss of half the light power. The present work is devoted to theoretical and experimental investigation of collinear acousto-optic interaction peculiarities that arise when the incident optical beam is not polarized. It is shown that the output diffraction spectrum in the general case contains four components which have different polarization and frequency. Beating of these components leads to modulation of light intensity passed through the output analyzer. In this work, dependences of modulation components amplitudes on frequency and power of the acoustic wave are examined for different polarizer and analyzer orientations.

Invited Paper

11:20

2aPAb10. Ultrasonic diffraction grating spectroscopy: particle size measurements and investigation of the inertial model for attenuation. Margaret Greenwood (Pacific Northwest National Laboratory, P. O. Box 999, Mailstop K5-26, Richland, Wa, WA 99352, USA, margaret.greenwood@pnl.gov)

The experimental setup consists of a unit with a grating machined on the surface of the unit. Send and receive transducers are placed at equal angles to the grating surface that is in contact with a slurry. The transmitted beam of spectral order $m = 1$ reaches an angle of 90° at the so-called critical frequency, where it becomes an evanescent wave that interacts with particles in the slurry. As a result of this interaction, the signal in the receive transducer exhibits a dependence upon particle size of the slurry. Data will be presented for slurries of polystyrene spheres, ranging in size from 45 μm to 467 μm, as a function of the volume fraction. The velocity of sound is also determined from the critical frequency. The distributions will be compared with the viscous and inertial models. The results show that larger particle sizes can be understood in terms of the inertial model, while the smaller sizes can be explained as a result of scattering. Also, arguments will be presented to show that this technique can probe the inertial model, without the simultaneous observation of scattering effects. Such studies can improve the theoretical interpretation of standard attenuation measurements.

Contributed Paper

11:40

2aPAb11. Non-reciprocity of acousto-optic interaction at high acoustic frequencies. Yury Dobrolenskiy (M.V. Lomonosov Moscow State University, Faculty of Physics, Vorob'evy Gory, MSU, 119991 Moscow, Russian Federation, dobrolenskiy@phys.msu.ru), Vitaly Voloshinov (M.V. Lomonosov Moscow State University, Faculty of Physics, Vorob'evy Gory, MSU, 119991 Moscow, Russian Federation, volosh@phys.msu.ru), Yury Zyuryukin (Saratov State Technical University, Politechnicheskaja str., 77, 410054 Saratov, Russian Federation, phys@sstu.ru)

Modern acousto-optic (AO) tunable filters have reached the values of spectral resolution exceeding 104 and operate with acoustic frequencies up to a few gigahertzes. At such high frequencies and such narrow bandwidths, there appear effects so far neglected. Among them there is non-reciprocity of AO interaction. It consists in the fact that acoustic frequency of light diffraction by ultrasound is different when optic beams propagate in directions opposite to each other. In the present research, the new effect has been studied theoretically and experimentally for the example of collinear AO diffraction. Theoretical analysis and calculations related to particular AO materials have shown that the effect is essential at acoustic frequencies about 1 GHz and higher. The value of the non-reciprocal shift of acoustic frequency can be as high as the frequency bandwidth of the filter. The effect has also been registered and investigated experimentally in a sample of lithium niobate crystal. The experimental data has totally confirmed the theoretical analysis. Therefore, it has been proved that the non-reciprocal effect influences the parameters of AO diffraction and, consequently, operation of AO filters at high frequencies. On the other hand, one can design devices, e.g. directional couples, exactly based on the non-reciprocity.

Invited Paper

12:00

2aPAb12. Critical cone channelling in directly bonded wafers. Evgeny Twerdowski (Institute of Experimental Physics II, University of Leipzig, Linnéstr. 5, 04103 Leipzig, Germany, twerdowski@physik.uni-leipzig.de), Moritz Von Buttlar (Institute of Experimental Physics II, University of Leipzig, Linnéstr. 5, 04103 Leipzig, Germany, vbuttlar@physik.uni-leipzig.de), Reinhold Wannemacher (Institute of Experimental Physics II, University of Leipzig, Linnéstr. 5, 04103 Leipzig, Germany, wannemacher@physik.uni-leipzig.de), Wolfgang Grill (Institute of Experimental Physics II, University of Leipzig, Linnéstr. 5, 04103 Leipzig, Germany, grill@physik.uni-leipzig.de)

Directly bonded semiconductor wafers have been investigated using ultrasonic transmission tomography and imaging of the phonon focusing patterns at ultrasonic frequencies. Beside of total disbonds, several bonded wafers contained defects that are fully transparent to normally incident waves of longitudinal polarization, and are fully opaque to those of transverse polarization. These defects, which are due to slip boundary conditions at the wafer-wafer interface, generate an additional acoustic mode by mode conversion at the interface. The additional mode is clearly observable in the experimental phonon focusing patterns and is indicative of the critical cone

channelling phenomenon, which is caused by the generation of pseudo-surface and head waves at weakly bonded solid-solid interfaces. The effect is also expected to be present in the case of weakly-bonded isotropic materials. Since the strong pseudo-surface wave resonance exists only in the case of boundary conditions that allow for the relative displacement of the two adjoining media along the interface, the critical cone channelling effect can serve as a measure of the type and quality of the bond.

Contributed Papers

12:20

2aPAb13. Sound waves in 3D periodic granular materials. Orion Mouraille (UTwente, Multi Scale Mechanics, P.O.Box 217, 7500 AE Enschede, Netherlands, o.j.p.mouraille@utwente.nl), Stefan Luding (UTwente, Multi Scale Mechanics, P.O.Box 217, 7500 AE Enschede, Netherlands, s.luding@utwente.nl)

The sound propagation mechanisms inside dense granular matter, even for periodical structures, are challenging the attempts to describe it because of the discrete nature of the material. Phenomena like dissipation, scattering, and dispersion are hard to predict based on the material state and/or properties and vice-versa. We propose here a simulation method using dynamic discrete elements in order to get more insight in this problem. The small perturbation created on one side of a dense, static regular packing of grains is examined during its propagation and when it arrives at the opposite side. Both longitudinal and shear perturbation are studied and an interesting wave-acceleration is observed [1]. Moreover the rotational degree of freedom permits to observe the role of rotations in the wave propagation. The control of the inter-particle forces like, contact potential, cohesion and friction make possible to observe the effect of these micro-parameters on the macro-behavior (at the wave scale). Long term goal is to predict the large-scale macroscopic material behavior from the microscopic structure and material parameters. [1] O. Mouraille, W A Mulder and S. Luding: Sound wave acceleration in granular materials, J. Stat. Mech. (2006) P07023.

12:40

2aPAb14. Bragg light diffraction in nontransparent crystals. Farkhad Akhmedzhanov (Navoi State Mine Institute, 27a Yuzhnaya Street, 210100 Navoi, Uzbekistan, farkhad2@yahoo.com)

As is well known, in absolutely transparent crystals an acoustooptical interaction is lacking. In this connection's acoustooptical investigations are presented greatest interest in optical range, in which the crystal is not transparent. In order to carry out similar experiments, it is necessary to use the crystal with a small light absorption coefficient at applied wavelength. The main problem is selection of an appropriate buffer crystal that the light beam can penetrate into sample by Bragg angle at required frequency. At present work the acoustooptical properties of Si crystal were investigated at light wavelength 632.8 nm. Dy_2S_3 crystals were used as a buffer sample. It was detected the diffracted light intensity is much more powerful in comparison with that for $LiNbO_3$ crystals. Acoustooptical efficiency M2 (it is defined the acoustic-optical quality of material) has been calculated from the values of optic coefficients and elastic constants for appropriate directions of sound and light. The results of executed investigations have shown the possibility of determination of acoustooptical properties of nontransparent crystals by Bragg diffraction method. At that rate, the very high intensity of diffracted light can be obtained in comparison with that in transparent crystals.

1:00-2:00 Lunch Break

Invited Paper

2:00

2aPAb15. Theory of resonant acoustic transmission through subwavelength apertures. Johan Christensen (Universidad Autónoma de Madrid, Departamento de Física Teórica de la Materia Condensada, Facultad de Ciencias, C-V, 28049 Madrid, Spain, johan.christensen@uam.es), Luis Martín-Moreno (Universidad de Zaragoza, Departamento de Física de la Materia Condensada, Instituto de Ciencia de Materiales de Aragón, C/ Pedro Cerbuna 12, 50009 Zaragoza, Spain, lmm@unizar.es), Francisco Jose García-Vidal (Universidad Autónoma de Madrid, Departamento de Física Teórica de la Materia Condensada, Facultad de Ciencias, C-V, 28049 Madrid, Spain, fj.garcia@uam.es)

The discovery of the phenomenon of extraordinary optical transmission through a two-dimensional array of subwavelength holes in a metallic film has opened a new line of research within optics. The key role played by surface plasmons in transferring light efficiently from the input side of the metal film to the output region was soon realized. This fundamental knowledge enabled extension of this surface-plasmon ability to achieve extraordinary optical transmission and strong collimation of light in a single hole surrounded by a finite periodic array of indentations. Here, we show how these ideas developed for electromagnetic radiation can be transferred to other classical waves such as acoustic waves.

Contributed Papers

2:20

2aPAb16. Theoretical and experimental ultrasonic investigation of a thin plate with regions of different periodic double-corrugations. Sarah W. Herbison (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, sherbison@gatech.edu), Nico F. Declercq (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, nico.declercq@me.gatech.edu)

In order to provide a qualitative measurement of the depth of periodic double-corrugations existing on both sides of a thin plate, theoretical and experimental ultrasonic techniques have been applied. The aesthetic galvanized steel sheets under investigation contain many regions of periodic double-corrugations, each region with a different corrugation depth. An ultrasonic technique is investigated as a means to improve existing optical testing techniques in quality and speed. Theoretical simulations using the theory of diffraction from periodically corrugated surfaces have been employed to obtain reflection spectra to facilitate qualitative observations regarding the depth of the corrugation in the different regions. Experiments to obtain reflection spectra correlate well with the theoretical simulations, and show the application of ultrasound to the qualitative measurement of the corrugation depth. Because the wavelengths necessary to make the investigations produce Lamb waves, some difficulty lies in the generation of diffracted bulk waves and Lamb waves. The research shows that correlations between theory and experiments exist, but they can only be obtained after thorough simulations that do not produce easy to apply rules of thumb.

2:40

2aPAb17. BEM analysis of plane waves scattered from periodic surfaces. Mahesh Bansal (Institute of Technical Acoustics, Tech. Univ. Berlin, Einsteinufer 25, 10587 Berlin, Germany, mbansal.iitk @gmail.com), Wolfgang Ahnert (Ahnert Feistel Media Group, Arkonastr. 45-49, 13189 Berlin, Germany, wahnert@ada-acousticdesign.de), Stefan Feistel (Ahnert Feistel Media Group, Arkonastr. 45-49, 13189 Berlin, Germany, sfeistel@afmg.eu)

Periodic surfaces like stairs, seats and repetitive designs on walls are very common and integral part of room acoustics. Effective simulation of the sound field inside enclosures requires the investigation of scattering from such periodic surfaces. In this work, we perform BEM analysis to calculate the scattering coefficients of different samples. For comparison purposes we also introduce a simple point-source based model to calculate the scattered wave fronts. Both approaches are implemented in a computational tool called EASE Scatterer. The incident plane waves are considered at various angles and scattered waves computed in both models are then compared with the measured data. It is found that while the point-source model can give reasonable asymptotic results, the BEM model matches with the measurement data significantly better in quantity and quality. Moreover, a study by varying the number of periods for the given sample is also performed.

3:00

2aPAb18. Diffraction phenomena associated with a composite plate containing an interior periodically corrugated interface. Sarah W. Herbison (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, sherbison@gatech.edu), Nico F. Declercq (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, nico.declercq@me.gatech.edu)

The interest in the study and applications of phononic crystals has naturally lead to the investigation of other novel periodic structures. The present work examines the case of a plate constructed of two solid layers of differing elastic properties separated by a periodically corrugated interface. It is shown how the dispersion curves are influenced by the internal corrugated interface and how they evolve as a function of the magnitude of this corrugation. Internal diffraction effects alter the dispersion properties and thus have an important effect on the composite when it is used as an acoustic filter. These effects are also important for the transmission and reflection of sound when the composite is used as a panel or when it is the intention to generate Lamb waves to investigate the composite plate nondestructively.

3:20

2aPAb19. Multiple scattering of acoustic waves from two transversely isotropic cylinders. Sina Sodagar (Faculty of Mechanical Engineering, K. N. Toosi University of Technology, Pardis St., Molla Sadra Ave., Vanak Sq., Postal code 1999143344, 16579 Tehran, Iran, ssodagar@alborz.kntu.ac.ir), Farhang Honarvar (Faculty of Mechanical Engineering, K. N. Toosi University of Technology, Pardis St., Molla Sadra Ave., Vanak Sq., Postal code 1999143344, 16579 Tehran, Iran, honarvar @mie.utoronto.ca), Anthony N. Sinclair (Department of Mechanical Engineering, University of Toronto, 5 King's College Road, Toronto, ON M5S 1A4, Canada, sinclair@mie.utoronto.ca)

The study of the interaction of acoustic waves with cylindrical structures has numerous applications including the ultrasonic nondestructive testing of materials. The scattered pressure field from a submerged cylinder contains valuable information about its physical properties. Scattering of acoustic waves from single cylindrical components including solid cylinders, shells, and multilayered cylinders has been of interest during the past two decades. These studies include theoretical modeling, numerical calculations, and experimental measurements of the scattered field of these objects. The more complex problem of multiple scattering from a grating of cylindrical components has also been considered during the past few years. These studies usually deal with either rigid or isotropic cylindrical objects. In this paper, the mathematical modeling for the scattering of plane acoustic waves from two adjacent infinite anisotropic solid cylinders will be presented. The type of anisotropy considered is transverse isotropy (hexagonal symmetry). The mathematical model accounts for the effects of the scattered filed of each cylinder on the total resultant pressure field. Numerical calculations are used to verify the validity of the developed mathematical model.

3:40-5:20 Posters

Lecture sessions will recess for presentation of poster papers on various topics in acoustics. See poster sessions for topics and abstracts.

Contributed Papers

5:20

2aPAb20. Phononic crystals in the diffraction regime. Vincent Laude (Institut FEMTO-ST/CNRS, 32 avenue de l'Observatoire, 25044 Besançon cedex, France, vincent.laude@femto-st.fr), Sarah Benchabane (Institut FEMTO-ST/CNRS, 32 avenue de l'Observatoire, 25044 Besançon cedex, France, sarah.benchabane@femto-st.fr), Abdelkrim Khelif (Institut FEMTO-ST/CNRS, 32 avenue de l'Observatoire, 25044 Besançon cedex, France, abdelkrim.khelif@femto-st.fr)

Phononic crystals are periodic composite materials exhibiting amazing wave propagation properties. In many works, complete band gaps are being looked for, i.e. the materials constituting the phononic crystal and its lattice arrangement are chosen such that propagation for all waves within a prescribed frequency range is forbidden. In other studies, the phononic crystal is considered a metamaterial, the anisotropic spatial dispersion of which can be tuned, and in which negative refraction can even be observed under certain circumstances. Such effects are usually considered in the sub-diffraction regime, i.e. below some critical onset frequency. In this work, we specifically examine phononic crystals in the diffraction regime. Indeed, the boundaries of a finite size phononic crystal embedded in a host propagation medium can be viewed as diffraction gratings, as we show. We will specifically consider two cases: two-dimensional phononic crystals composed of steel rods in water, and two-dimensional phononic crystals for surface acoustic waves achieved by etching cylindrical holes in a solid substrate.

5:40

2aPAb21. Methods to investigate the possibilities of using a three element periodic structure to suppress the transmission of energy in an elastic tube. Ole Holst-Jensen (Minus10dB, Stokrosevej 29, 8330 Beder, Denmark, ole.holst@minus10db.dk), Sergey Sorokin (Univ. of Aalborg, Institute for mechanical engineering, Pontoppidanstraede 101, 9220 Aalborg East, Denmark, svs@ime.aau.dk)

In industrial applications, such as pump and compressor systems, pipe vibrations can exceed an acceptable level. Periodically spaced masses added to the piping are an appropriate method of attenuation, and investigations to develop valid prediction models and measurement techniques has been initiated in the Danish Makunet network. The present investigation concern the effect of three masses attached periodically to a pipe of small diameter. The pipe is small enough that it can be treated as a beam. The masses are eccentric to the center of the beam, to achieve a large change in the moment of inertia by the added elements. The theoretical model is formulated as a system of boundary equations, which describe propagation of flexural, axial and torsion waves within each segment of a tube between periodic elements. An exact solution of this system is obtained and the power transfer is dramatically reduced in some frequency 'stop bands' regardless the excitation conditions. Transfer impedance measurements between the force input and the acceleration on selected positions are used to find the insertion loss due to the application of the periodic elements, showing good agreement with theory.

6:00

2aPAb22. Diffraction influence on the SAW tag characteristics. Boris V. Sveshnikov (Lebedev Research Center in Physics, 23, Lenin Av., 119991 Moscow, Russian Federation, bvs@ieee.org)

The diffraction of acoustic beams may influence notably on the responses of the SAW based RFID systems. In order to prevent such an influence one cannot increase the tag acoustic aperture too much because of ohmic loss in a transponder. On the other hand, usually the tag length should be large enough. There is a very specific and unique feature of the acousto-electric scattering of SAW beam on the coding sequence of the reflective metal electrodes placed on a piezoelectric substrate. The point is that the SAW itself is formed by two coupled sub-systems, namely: elastic displacements and electric field. Both of them are scattered on every discontinuity because of two kinds of the boundary condition perturbation: electric shorting beneath electrodes and mass-loading effect. A uniform metal electrode scatters differently from each other the ″elastic″ and ″electric″ subsystems with non-uniform wave-front caused by diffraction. The closed form expressions are found, allowing us to describe consistently the mentioned phenomena. Their numerical analysis has been added by the analytical one in case of parabolic approximation of the crystal slowness curve. The synthesis algorithm of SAW tags with high capacity is proposed taking into account the diffraction effect in a natural way.

6:20

2aPAb23. Frequency-domain diffraction for edges of arbitrary length with efficient numerical integration. Peter Svensson (Norwegian Univ. of Science and Technology, O.S. Bragstads plass 2B, Dept. of Electronics and Telecommunications, NO-7491 Trondheim, Norway, svensson@iet.ntnu.no), Paul Calamia (Rensselaer Polytechnic Institute, Greene Bldg., 110 8th St., Troy, NY 12180, USA, calamp@rpi.edu)

Frequency-domain edge diffraction from an infinite wedge insonified by a point source has been widely studied, with various analytical solutions dating back nearly a century. In this talk we present an alternative frequency-domain solution which can be used for finite as well as infinite edges. The expression, given as a line integral along the diffracting edge, is derived from an analytical time-domain expression presented in [U. P. Svensson et al., J. Acoust. Soc. Am. 106, 2331-2344 (1999)]. The new formulation is shown to be equivalent to an exact, analytical, contour-integral formulation for the infinite wedge via a variable transformation. Results for various finite and infinite edge cases will be presented, and numerical integration using an efficient quadrature method for highly oscillatory integrals will also be discussed.

6:40

2aPAb24. Acoustic cloaking by two-dimensional sonic crystals. Daniel Torrent (Politechnic University of Valencia, Wave Phenomena Group. Department of Electronic Engineering, C/ Camino Vera s.n, ES-46022 Valencia, Spain, datorma1@upvnet.upv.es), José Sanchez-Dehesa (Polytechnic University of Valencia, Cami de Vera s/n, 46022 Valencia, Spain, jsdehesa @upvnet.upv.es)

It has been previously shown that a two dimensional sonic crystal made of a periodic arrangement of solid cylinders in air behaves in the low frequency limit (homogenization) like an acoustic metamaterials with anisotropic effective density and isotropic effective bulk modulus [D. Torrent and J. Sanchez-Dehesa, New J. Phys. February 2008]. Here, the homogenization method is extended to the case of sonic crystals with two types of cylinders in the unit cell, and analytical expressions for the anisotropic effective density are then obtained. Moreover, it will be shown how these new acoustic metamaterials can be used to physically realize the solution for the acoustic cloaking proposed by Cummer and Schurig [New. J. Phys., vol. 9, 45, 2007] [Work supported by MEC of Spain.]

TUESDAY MORNING, 1 JULY 2008 ROOM 352A, 8:00 A.M. TO 7:00 P.M.

Session 2aPAc

Physical Acoustics: Outdoor Sound Propagation and Uncertainties I

Keith Wilson, Cochair
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Michel C. Berengier, Cochair
Laboratoire Central des Ponts et Chaussées, Centre de Nantes - Route de Bouaye, BP 4129, Bouguenais cedex, 44341, France

Invited Papers

8:00

2aPAc1. Expected mean in an environmental noise measurement and its related uncertainty. Marco Paviotti (European Commission, via e. fermi, 1, 21020 Ispra, Italy, marco.paviotti@jrc.it), Stylianos Kephalopoulos (European Commission, via e. fermi, 1, 21020 Ispra, Italy, Stylianos.Kephalopoulos@jrc.it)

In the context of the implementation of the Environmental Noise Directive 2002/49/EC a study on noise measurement uncertainty was developed. In performing any noise measurement average, there might be assumptions over the distribution and independency of the samples. In a concrete example dealing with environmental noise, this might be the case of a series of measurements of a constant noise source like an industrial plant or a fluctuating noise source like road traffic. Using a series of 1 day Leq in the first case, or a series of 15 minutes samples in the second case, the average of these values is usually considered as the expected mean, however, the error caused by the specific selection of the samples is not evaluated. Statistically speaking, before establishing an average value, at least the lognormal distribution of the samples and the effect of adding-up several uncertainties should be evaluated. This article will focus on the formulas to be used and will discuss differences in assessing the expected mean for normally distributed values, or for lognormally distributed, and will suggest an approach to properly add-up all uncertainties related to an environmental noise measurement campaign.

8:20

2aPAc2. Criterion to select meteorological factors to evaluate uncertainties in sound propagation. Panu Maijala (VTT, Tekniikankatu 1, 33101 Tampere, Finland, Panu.Maijala@vtt.fi)

Most of the environmental factors have some effect on sound propagation outdoors. Many of these factors can be properly implemented to a sound propagation model. However, it is not easy to handle sound scattering due to the turbulence, and at the same time, the turbulence is the most important source of uncertainties. Concurrently with the studies of turbulence models we have developed a concept to get an estimate of the excess attenuation using a state-of-the-art physical model and to evaluate the uncertainties using a statistical model. This statistical model is based on two years continuous measurements using extensive acoustical and meteorological measurement facilities and producing over 100 factors hourly. Many meteorological factors had a strong and significant correlation with the excess attenuation, but between each other too. To avoid instable model due to the collinearity many factors were abandoned. In this paper the criterion and methods to select best explaining factors and to form this statistical model are considered.

8:40

2aPAc3. Methodological and metrological benchmarks for determining experimental values of meteorological parameters. Benoit Gauvreau (Laboratoire Central des Ponts et Chaussées, Centre de Nantes - Route de Bouaye, BP 4129, 44341 Bouguenais cedex, France, benoit.gauvreau@lcpc.fr)

Outdoor sound propagation is largely influenced by meteorological conditions: mean refraction, intermittency and atmospheric turbulence. Thus, acoustic engineers and scientists have to rigorously characterize those propagation conditions through several parameters, both for numerical predictions (input data) and for operational reports (impact studies). One of these influent parameters is the well-known (and widely used) effective vertical sound speed gradient, which depends on spatial and temporal mean values of wind and temperature between source and receiver. However, the experimental characterization of wind and temperature (and next sound speed) vertical profiles remains delicate: it requires accurate meteorological devices and adapted measuring methods. At this purpose, an experimental campaign has been carried in 2007 on a perfectly flat and open ground, in order to quantify the confidence, uncertainty and spreading of such parameter values for different time scales. This campaign involved different meteorological devices: instrumented towers, classical sensors, 3D sonic anemometers, etc. The benchmark results will be presented and next analysed in terms of metrological and methodological requirements for acousticians in charge of environmental studies for standard organizations and/or scientific institutes.

9:00

2aPAc4. Sources and potential influences of uncertainty in ground impedance measurements and estimations. Shahram Taherzadeh (The Open University, Faculty of Mathematics, Computing and Technology, Walton Hall, MK7 6AA Milton Keynes, UK, s.taherzadeh@open.ac.uk), Keith Attenborough (Open University, Department of Design, Development, Materials and Environment, Walton Hall, MK7 6AA Milton Keynes, UK, Keith.Attenborough@ioa.org.uk)

Knowledge of acoustic impedance of ground surface is essential for determining noise levels outdoors. At distances up to a few hundred metres from the source ground effect may be the dominant factor. The short-range measurement of sound level spectra from an omni-directional source has been used as a standard method to deduce the acoustic impedance spectrum of ground surfaces (e.g. ANSI S1.18, 1999). However, there remain a number of uncertainties in such measurement methods. For example, at low frequencies the difficulties in fitting theoretical or phenomenological models to short range data may result is a large uncertainty in predictions of sound levels at larger distances. Another source of uncertainty is the variability of the acoustic impedance of apparently uniform ground. Another potential influence on predictions of long term equivalent noise levels is seasonal variation. This paper discusses the extent and potential influences of such uncertainties in measured and deduced ground impedance spectra.

9:20

2aPAc5. Facing the challenge of calculating outdoor sound propagation using a 3D multi domain approach based on linear euler equations. Fabrice Junker (EDF, 1, avenue du Général de Gaulle, 92141 Clamart, France, fabrice.junker@edf.fr), Fabien Crouzet (EDF, 1, avenue du Général de Gaulle, 92141 Clamart, France, fabien.crouzet@edf.fr), Philippe Lafon (EDF, 1, avenue du Général de Gaulle, 92141 Clamart, France, Philippe.lafon@edf.fr)

Solving the Linear Euler Equations (LEE) is the reference method that allows to take into account all the phenomena involved in Outdoor Sound Propagation. However, the huge size of the problems to be treated is still a great limitation to the practical application of this method. The concept of multi domain computations associated with the use of massively parallel computers now pushes the limits away. The code SAFARI, recently developed by EDF, solves LEE with high order numerical schemes on structured grids. To deal with complex and large geometries, a multi domain approach is used. The computational domain is composed of several partially overlapping grids (overset grids). Computations are parallelized by domain decomposition to be runned on cluster facilities. The presentation means to show the capability of SAFARI to deal with propagation over realistic 3D domains. The strategy used to carry out the calculations is detailed. The new perspectives of this kind of method are finally given.

9:40

2aPAc6. Computing the parameter sensitivities of outdoor sound propagation in a random environment. Chris L. Pettit (U.S. Naval Academy, Aerospace Engineering Dept., 590 Holloway Rd., MS 11-B, Annapolis, MD 21402, USA, petitcl@usna.edu), Keith Wilson (U.S. Army Cold Regions Research and Engineering Laboratory, 72 Lyme Road, Hanover, 03755-1290, USA, D.Keith.Wilson @erdc.usace.army.mil)

Computational forecasts of near-ground sound levels are compromised by uncertainty (e.g., randomness) and error (e.g., grid resolution) in the atmosphere and terrain representations, and by simplified or incorrect physics. For an incompletely known environment, a model's predictive power cannot be assessed without first quantifying the sensitivity of its forecasts to the full range of every parameter. Knowledge of these sensitivities throughout the spatial domain also is essential for effectively investing data-gathering resources to support sound propagation forecasts. Sensitivity analysis therefore is central to raising the relevance of computational acoustics in practical applications. These considerations should motivate practitioners of computational acoustics to adopt a consistent framework for sensitivity and uncertainty analyses. Topics to be discussed include: (1) standard uncertainty taxonomies in computational mechanics, (2) why uncertainty about a parameter should be distinguished from sensitivity of a model to that parameter, (3) sources of uncertainty in the near-ground acoustics, (4) a sampling-based sensitivity analysis framework that facilitates estimating typical and extreme values of sensitivities at each point in the spatial domain (i.e., full-field sensitivities), (5) factors to be aware of when applying sensitivity analysis to forecasts of near-ground sound propagation, and (6) ways of representing sensitivity estimates to facilitate insight.

10:00-10:20 Break

10:20

2aPAc7. Field experiments on the influences of wind speed and direction on outdoor sound propagation over flat ground. Takatoshi Yokota (Kobayasi Institute of Physical Research, 3-20-41 Higashi-Motomachi Kokubunji, 185-0022 Tokyo, Japan, t-yokota@kobayasi-riken.or.jp), Koichi Makino (Kobayasi Institute of Physical Research, 3-20-41 Higashi-Motomachi Kokubunji, 185-0022 Tokyo, Japan, makino@kobayasi-riken.or.jp), Kohei Yamamoto (Kobayasi Institute of Physical Research, 3-20-41 Higashi-Motomachi Kokubunji, 185-0022 Tokyo, Japan, yamamoto@kobayasi-riken.or.jp), Yasuaki Okada (Faculty of Science and Technology, Meijo University, 1-501 Shiogamaguchi Tempaku-ku, 468-8502 Nagoya, Japan, okada@ccmfs.meijo-u.ac.jp), Koichi Yoshihisa (Faculty of Science and Technology, Meijo University, 1-501 Shiogamaguchi Tempaku-ku, 468-8502 Nagoya, Japan, yosihisa@ccmfs .meijo-u.ac.jp)

The influences of wind on outdoor sound propagation are investigated both by field experiments and numerical simulations. At first, measurements of sound propagation at a distance of 160 m were carried out under various wind speed and direction. The relationship between vector wind speed and the variation of sound pressure level has been examined not only for the up/down wind conditions but also for the cross wind conditions. Secondly, the short term sound fluctuations were investigated by the use of a high energy impulsive sound generator and a loud speaker as sound sources. Sound propagation up to 300 m was examined with frequency range from 16 Hz to 4 kHz octave band. Finally, the sound speed profiles in the measurement field were estimated by two methods; one was based on the time interval during the sound propagation from the source to the receivers, the other was based on the wind speed at height of 0.6 m, 1.2 m, 2.4 m and 4.8m, and the temperature. By using those sound speed profiles, the excess attenuation was calculated by the PE method. The measured and the calculated values have been compared and the methods to estimate the influences due to wind condition have been discussed.

10:40

2aPAc8. Variation in measured sound level as a function of propagation environment and distance. Michelle E. Swearingen (Norwegian Defense Research Establishment/US Army, Postboks 25, 2027 Kjeller, Norway, michelle.swearingen@ffi.no), Morten Huseby (Norwegian Defense Research Establishment/US Army, Postboks 25, 2027 Kjeller, Norway, Morten.Huseby@ffi.no), Michael J. White (US Army Engineer Research and Development Center, 2902 Newmark Drive, Champaign, IL 61826, USA, Michael.J.White @usace.army.mil)

The propagation environment exerts a large influence on the range of received levels of impulsive events. This talk focuses on the variation in excess attenuation over durations of less than approximately 15 minutes. Data are presented for greatly different measurement distances (25 m to 7 km) and propagation environments (sparse vegetation to forested), illustrating the effects of distance and terrain cover on sounds from a propane cannon and an artillery source. Over sparse vegetation 7 km from an artillery source, the received CSEL varied 11 dB within a 12-minute duration. In measurements up to approximately 300 m from the source, variation in received level (both peak and SEL) was less than 1 dB within the forest, and much more in the open. The control of the forest canopy on the micrometeorology seems to explain the effect.

11:00

2aPAc9. The revision of the French method for road traffic noise prediction. Guillaume Dutilleux (Lab. Régional des Ponts et Chaussées, 11, rue Jean Mentelin, BP 9, 67035 Strasbourg Cedex 2, France, Guillaume.Dutilleux@equipement.gouv.fr), Jérôme Defrance (CSTB, 24 rue Joseph Fourier, 38400 Saint-Martin-d'Hères, France, jerome.defrance@cstb.fr), Benoit Gauvreau (Laboratoire Central des Ponts et Chaussées, Centre de Nantes - Route de Bouaye, BP 4129, 44341 Bouguenais cedex, France, benoit.gauvreau@lcpc.fr), Francis Besnard (Lab. Reg. Régional des Ponts et Chaussées de l'Est Parisien, Rue de l'égalité prolongée, BP 34, 93352 Le Bourget, France, francis.besnard@equipement.gouv.fr)

A revision of the French method for road traffic noise prediction (NMPB-Routes-2008) has been released. The major principles of NMPB-Routes-2008 are outlined. The most important modifications regarding the source are the reduction of its height and the introduction of two different spectra. In this complete revision, the main change is the replacement of the ISO 9613-2 - based ground attenuation formula in downward conditions by the formula for ground attenuation in homogeneous conditions of NMPB-Routes-1996

with corrected heights in order to take into account the mean curvature of rays (refraction) and its spreading (turbulence). The revised NMPB adds an attenuation term for an occasional cutting embankment. Regarding diffraction the Δdif formula is now suitable for low height barriers. The validation of the revised NMPB with respect to experiment is presented. It is based on measurement campaigns on 6 sites with complex geometries and shows that the predicted noise levels obtained from the revised NMPB are significantly closer to experimental results than in the case of the original method.

11:20

2aPAc10. Long range propagation of high speed train noise: Sound level variations before and after the pass-bys. Benjamin Cotte (Ecole Centrale de Lyon, LMFA, UMR CNRS 5509, Université de Lyon, Bât. KCA, 36 Avenue Guy de Collongue, 69134 Lyon, France, benjamin.cotte@ec-lyon.fr), Philippe Blanc-Benon (Ecole Centrale de Lyon, LMFA, UMR CNRS 5509, Ecully, 69134 Lyon, France, Philippe.Blanc-Benon@ec-lyon.fr), Franck Poisson (SNCF DIR, 45, rue de Londres, 75379 Paris, France, franck.poisson@sncf.fr), Cora Cremezi-Charlet (UIC, 16,rue J. Rey, F-75015 Paris, France, cremezi@uic.asso.fr)

It has been observed relatively frequently that high speed train (TGV) noise could be heard tens of seconds before or after the actual train pass-bys. This study is aimed at characterizing this phenomenon and the conditions in which it occurs, both experimentally and numerically. Acoustic measurements of TGV pass-bys have been performed under controlled conditions. A relatively strong wind was blowing from the South during the experiment. The measurements show that the TGV noise can be heard before the pass-bys when the TGV was coming from the South (same wind and train directions), and after the pass-bys when the TGV was coming from the North (opposite wind and train directions). This noise is relatively low in frequency (around 400 Hz), and corresponds to propagation distances that can exceed 1 km. Levels associated with this phenomenon can vary significantly over short time intervals (5-10 minutes), which raises the issue of the representativeness of TGV measurements at long ranges. It will be shown using numerical prediction methods (parabolic equation in the frequency domain, linearized Euler equations in the time domain) that these acoustic variations are mostly due to variations in the meteorological conditions between the pass-bys.

11:40

2aPAc11. A model based monitoring system for aircraft noise. Frank Van Den Berg (TNO - Science and Industry, Stieltjesweg 1, 2628 CK Delft, Netherlands, Frank.vandenBerg@tno.nl), Arno R. Eisses (TNO, Stieltjesweg 1, P.O.Box 155, 2600 AD Delft, Netherlands, Arno.Eisses@tno.nl), Pieter J. Van Beek (TNO, Stieltjesweg 1, P.O.Box 155, 2600 AD Delft, Netherlands, Pieter.vanBeek @tno.nl)

A new approach for an airport noise monitoring system is presented that comprises not only a number of measuring stations, but also a dedicated sound propagation model and an aircraft noise emission model. This approach enables estimation of noise levels in the whole area around the airport and not only at the location of the measuring stations. An additional advantage is that the locations of the measuring stations can be chosen more freely. Traditionally the measuring stations must be located near residential areas where the influence of other noise sources (such as cars or wind) cannot be eliminated. Better locations can be found that yield more reliable data and therefore more reliable noise levels. To update the sound propagation model frequently, the measuring stations not only measure noise levels, but also other parameters, like temperature and wind profile. The sound emission of the aircraft is derived from a directional aircraft noise emission model. A tool was developed to visualize the power of combining data and models in this model-based monitoring system. The use of models opens possibilities for interesting applications such as i) short term noise forecasts, ii) scenario studies with various aircraft distributions and iii) enforcement of noise limits.

Contributed Papers

12:00

2aPAc12. The variability of acoustical turbulence in the atmospheric boundary layer. Sylvain Cheinet (ISL, 5 Rue du General Cassagnou, BP 70034, 68301 Saint-Louis, France, sylvain.cheinet@isl.eu)

Outside sound propagation is affected by small-scale turbulence in the atmospheric boundary layer through refractive processes. These effects modulate the performance of acoustical sensors, and are at the basis of the atmospheric remote sensing by sodar. Previous studies have documented the impact of a mean turbulence profile or a statistical distribution of turbulence. In this study, we use a high-resolution atmospheric model called LES to reveal the spatial and temporal variability of the small-scale turbulence. First, we introduce some standard results on the role of various eddy sizes on acoustical propagation. Second, we discuss the methodology to derive the acoustical refractive index fluctuations from LES data. Third, we analyze the LES prediction in the case of a fair-weather boundary layer, emphasizing the impact of large-scale convective organization. Finally, we discuss some challenges in attempting to predict the acoustical turbulence from present-day weather forecasts.

12:20

2aPAc13. Sound propagation in areas with a complex meteorology: a meteorological-acoustical model. Frits Van Der Eerden (TNO - Science and Industry, Stieltjesweg 1, 2628 CK Delft, Netherlands, frits.vandereerden@tno.nl), Frank Van Den Berg (TNO - Science and Industry, Stieltjesweg 1, 2628 CK Delft, Netherlands, Frank.vandenBerg@tno .nl)

Long range sound propagation is largely affected by the vertical wind and temperature gradients. These gradients are more complicated in areas where the meteorology can be complex, such as: coastal areas, islands, and lake districts. Furthermore, the gradients usually vary as a function of the horizontal distance, for instance at water-land crossings near a coast or a lake. As a result the sound propagation in these areas cannot be calculated with ″standard″ acoustic models, such as the ISO 9613 or Harmonoise engineering model. By using an advanced meso-scale meteorological model, which is using large-scale weather forecast data, and an acoustical model, that incorporates horizontal and vertical meteorological variations, long range sound propagation in such complex areas is calculated. In this paper

we present two applications of this hybrid meteorological-acoustical calculation scheme: i) propagation of impulse noise in a coastal area, and ii) propagation of industrial noise in a rural area with a lake. For the coastal area sound level contours have been calculated as a function of the meteorology for a complete year. For the lake area the effect of the lake on the sound propagation is demonstrated.

12:40-2:00 Lunch Break

Contributed Papers

2:00

2aPAc14. Horizontal wave number spectra of atmospheric acoustic fields in range-dependent environments. Kenneth E Gilbert (National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA, kgilbert@olemiss.edu), Xiao Di (National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA, xiaodi@olemiss.edu), Santosh Parakkal (National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA, sparakka @olemiss.edu)

The parabolic equation (PE) is a powerful method for computing sound propagation in range-dependent environments. However, the PE gives only the total field, with no information on the horizontal wave number spectrum of the acoustic field. In many situations, e.g., for propagation over irregular terrain, the horizontal wave number spectrum is valuable for analyzing the modal content of the acoustic field. For example, one might want to know the strength of the surface wave component at different ranges. A spectral decomposition method developed by Gilbert and Evans for ocean acoustics is applied here to atmospheric propagation over irregular terrain. This paper shows that for a PE field $p(z)$ at range r, the horizontal wave number spectrum $\phi(\kappa,z)$ can be obtained directly and efficiently by solving a one-dimensional elliptic equation that uses the field $p(z)$ as a source term. The derivation of the spectral decomposition equation is given along with a numerical method for solving it. Examples are presented and discussed for propagation over irregular terrain. [Research supported by the U. S. Army TACOM-ARDEC at Picatinny Arsenal, New Jersey]

2:20

2aPAc15. Numerical modeling of sonic boom propagation from hypersonic aircraft. Alexandra Loubeau (Institut Jean Le Rond d'Alembert, Université Pierre et Marie Curie, Boites 161 et 162, 4 place Jussieu, 75252 Paris Cedex 05, France, loubeau@lmm.jussieu.fr), François Coulouvrat (Institut Jean Le Rond d'Alembert, Université Pierre et Marie Curie, Boites 161 et 162, 4 place Jussieu, 75252 Paris Cedex 05, France, coulouvr@ccr.jussieu.fr)

A numerical study of sonic boom propagation from hypersonic aircraft is performed including the effects of nonlinearity, atmospheric absorption and dispersion, and atmospheric stratification. A second-order split-step algorithm, which alternates application of nonlinearity in the time domain and complex absorption in the frequency domain, allows for a faster convergence of results with fewer range steps than with conventional first-order algorithms. Nonlinearity is calculated using the potential, the integral of the pressure, as proposed by Burgers and later applied to sonic booms by Hayes et al. This method, an alternative to Landau's law of equal areas, efficiently locates the shock position by selecting the maximum potential in multivalued regions. Definition of atmospheric absorption at high altitudes is important for modeling the propagation of sonic booms from hypersonic aircraft. Some aspects of an extended absorption model by Sutherland and Bass are adopted, therefore, which extend absorption predictions above the 20 km limit of the current ISO and ANSI standards. The study is completed using the meteorological conditions at two locations, Le Havre, France and Edwards Air Force Base, CA, USA, over the course of a year. [Work supported by European Union through ATLLAS AST5-CT-2006-030729, meteorological data provided by ECMWF.]

2:40

2aPAc16. Road noise: characterization and estimation of uncertainty due to meteorological effects. David Ecotiere (Lab. Régional des Ponts et Chaussées, 11, rue Jean Mentelin, BP 9, 67035 Strasbourg Cedex 2, France, david.ecotiere@equipement.gouv.fr)

Meteorological effects can lead to important temporal fluctuations of the sound level in outdoor sound propagation. These fluctuations lead to sound level uncertainties that are rarely estimated. A method based on the coupling of a model of sound propagation and a temporal micrometeorological model is used to quantify sound level fluctuations only due to meteorological fluctuations, over a very long period (typ. 10-30 years). Some statistical analysis are presented: diary or seasonal fluctuations, influence of the duration of observation on the accuracy of the estimation of a LAeq. A semi-analytical method is also proposed for characterizing the uncertainty of the sound level of a distribution of punctual sources. An application to a road noise source is presented.

3:00

2aPAc17. A numerical study of sound propagation over urban canyons. Martin Schiff (Division of Applied Acoustics, Chalmers University of Technology, 41296 Göteborg, Sweden, schiff@student.chalmers.se), Maarten Hornikx (Applied Acoustics, Chalmers University of Technology, Sven Hultins Gata 8a, SE-41296 Gothenburg, Sweden, maarten.hornikx@chalmers.se), Jens Forssén (Division of Applied Acoustics, Chalmers University of Technology, 41296 Göteborg, Sweden, jens .forssen@chalmers.se)

Because quiet areas in dense urban environments are important, there is high interest in propagation to areas shielded from direct road traffic noise. Sound levels in shielded areas are strongly influenced by distant sources, so intermediate propagation factors such as metrology, screening, and intermediate canyons must therefore be addressed in a realistic propagation model. A numerical investigation of sound propagation across the open tops of intermediate urban canyons has been performed, using the Parabolic Equation and Equivalent Sources methods. Results have been collected for various canyon geometries, and the influence of multiple canyons, canyon/rooftop absorption, variable rooftop height, and correlated versus uncorrelated source models has been investigated. By characterizing the ″insertion loss″ of canyons intermediate to the source and receiver, the influence of these intermediate canyons could be addressed simply, without the overhead of a detailed numerical calculation.

3:20

2aPAc18. A validation test for the Acoustic Footprint Mission Planning System. Doru Velea (QinetiQ North America, Technology Solutions Group, 12030 Sunrise Valley Dr., Suite 400, Reston, VA 20191, USA, dvelea@plansys.com), Troy D. Schultz (Wyle Laboratories Inc., 241 18th Street S., Suite 701, Arlington, VA 22202, USA, troy.schultz@wylelabs.com), Kenneth J. Plotkin (Wyle Laboratories Inc., 241 18th Street S., Suite 701, Arlington, VA 22202, USA, kenneth.plotkin@wylelabs.com), Andy Rogers (QinetiQ North America, Technology Solutions Group, 12030 Sunrise Valley Dr., Suite 400, Reston, VA 20191, USA, arogers@plansys.com)

QinetiQ North America, Technology Solutions Group (QNA) and Wyle Laboratories have integrated two mature, proprietary technologies into a prototype Acoustic Footprint Mission Planning System. Wyle's Noise Model Simulation (NMSim) uses validated linear noise propagation routines in combination with ray-tracing algorithms based on weather data provided by QNA's Precision Airdrop System (WindPADS) to predict the time his-

tory of spectral noise levels from a flying aircraft at any number of user-defined receptor locations. Through interpolation over a ground-based receptor grid, the impact of noise propagated through a stratified atmosphere may be quantified in terms of audibility at any point inside the acoustic footprint. Weather data for the NMSim ray-tracing routines is extracted from the WindPADS three-dimensional forecast of wind, pressure, temperature and humidity in the atmosphere as a function of time. The three-dimensional field is the result of physical and dynamic assimilation of forecast and measured atmospheric data, with wind-forcing by the underlying terrain. This paper describes the results of a validation test performed at Yuma Proving Ground, AZ. It is shown that the noise propagation model predictions agree satisfactorily with the measurements of a C-130 cargo plane. [Work supported by Natick Soldier Center.]

3:40-5:20 Posters

Lecture sessions will recess for presentation of poster papers on various topics in acoustics. See poster sessions for topics and abstracts.

Contributed Papers

5:20

2aPAc19. Modal probabilistic analysis. Christophe Heinkele (ENTPE/DGCB, Rue Maurice Audin, 69518 Vaulx-en-Velin, France, heinkele@entpe.fr), Claude-Henri Lamarque (rue Maurice Audin, 69100 Vaulx-en-Velin, France, lamarque@entpe.fr)

In this paper, we first recall a method for estimating the numerical Probability Density Function (PDF) with Parzen-Rosenblatt estimators of experimental data (like the sound absorption coefficient with the Kundt's tube). Then we focus on the model of Mikki which is used as a predictive tool. We are interested then in inversing the model and in identifying the 3 parameters (q,σ,ϕ) of Mikki's model. But we want to go further and to identify from the estimated PDF directly the PDF of the 3 parameters. We explain first when it is possible, then we give some examples to illustrate the method. This work allows for evaluating the robustness of a model from experimental data.

5:40

2aPAc20. An experimental evaluation of a new approach to aircraft noise modelling. Foort De Roo (TNO Science and Industry, Stieljesweg 1, 2628CK Delft, Netherlands, foort.deroo@tno.nl), Erik Salomons (TNO Science and Industry, Stieljesweg 1, 2628CK Delft, Netherlands, erik.salomons@tno.nl)

Common engineering models for aircraft noise, such as INM, yield noise levels by interpolation of Noise Power Distance (NPD) tables. In the European project Imagine (2004 2006), a different approach was proposed: the source is characterized by an emission spectrum and the received noise spectrum is calculated by subtracting the propagation attenuation spectrum from this emission spectrum. This is the usual approach for noise mapping of most noise sources. The aircraft emission spectrum is a function of (downward) emission direction, so each aircraft is represented by a hemisphere of emission spectra. This has been described by Butikofer in Acta Acustica 93 (2007). As hemisphere emission data are not yet available for all aircraft types, a 'reverse engineering' scheme was developed within Imagine to derive first order estimates of hemispheres from NPD tables. To gain experience with this approach, we have performed an experiment near Amsterdam airport. Various types of data were collected for a set of aircraft departures, including noise data at eleven positions and flight data. The Imagine approach was used to calculate noise contours, and noise spectra at the eleven positions. The differences between measured and calculated spectra may be used as a basis for improving the first order estimates of the hemispheres.

6:00

2aPAc21. Sound propagation in a street canyon: A study by modal decomposition. Adrien Pelat (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, adrien.pelat.etu@univ-lemans.fr), Simon Félix (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, simon.felix@univ-lemans.fr), Vincent Pagneux (Laboratoire d'Acoustique de l'Université du Maine, UMR CNRS 6613, AV. O. Messiaen, 72085 Le Mans, France, vincent.pagneux@univ-lemans.fr), Christophe Ayrault(LAUM, CNRS, Université du Maine, Av. O. Messiaen, 72085 Le Mans, France, christophe.ayrault@univ-lemans.fr), Olivier Richoux (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, olivier.richoux@univ-lemans.fr)

An urban, U-shaped, street canyon being considered as an open waveguide in which the sound may propagate, one is interested in a multimodal approach to describe the sound propagation within. The key point in such a multimodal formulation is the choice of the basis of local transversal modes on which the acoustic field is projected. For a classical waveguide, with a simple and bounded cross-section, a complete orthogonal basis can be analytically obtained. The case of an open waveguide is more difficult, since no such a basis can be exhibited. However, an open resonator, as displays for example the U-shaped cross-section of a street, presents resonant modes with complex eigen frequencies, owing to radiative losses. This work first presents how to numerically obtain these modes and, then, how they can be used as a basis for the modal decomposition of the sound field in a street canyon. Results are compared with experimental measurements on a scale model.

6:20

2aPAc22. Acoustical waves propagation in rough walls street. Bertrand Lihoreau (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, bertrand.lihoreau@univ-lemans.fr), Simon Félix (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, simon.felix@univ-lemans.fr), Claude Depollier (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, claude.depollier@univ-lemans.fr)

We present a method for the numerical calculation of a sound wave propagating in a two dimensional rough-sided street. Is this situation, the wave is subject to multiple scattering at the two boundaries. The propagation is governed by a parabolic equation. We show that the amplitude of a pulse may be expressed as the fractional Fourier transform of the incident pulse, and that the order of the transformation is related to the distance between the source and the point of observation. Numerical simulations are proposed as illustration of this model and are compared to experimental results.

6:40

2aPAc23. A high-density infrasound array of particle velocity sensors in the Netherlands. Arnout Tim Van Zon (Royal Netherlands Meteorological Institute (KNMI), PO Box 201, 3730 AE De Bilt, Netherlands, zon@knmi.nl), Laeslo G. Evers (Royal Netherlands Meteorological Institute (KNMI), PO Box 201, 3730 AE De Bilt, Netherlands, evers@knmi.nl)

A small aperture, High-Density Infrasound Array (HDIA) is being operated by the Royal Netherlands Meteorological Institute (KNMI). This 80 sensor array is part of the geophysical application within LOFAR, i.e. an astronomical low-frequency array in the Netherlands. HDIA occupies an area of 100 by 100 meter, so about the size of a noise reducer used in infrasound arrays for verification purposes. HDIA will be able to estimate the wind-noise correlation length, which has implications for noise reducer design. Of the 80 instruments, 74 are Microflowns. These measure particle velocity and have a directional sensitivity. If two of them are closely spaced, then the azimuth of the incoming sound wave can be calculated from the

amplitude ratio. The field setup has 37 elements, each containing a pair of Microflowns. Six elements also have a pressure microphone. We will present the first results of using the vector properties of the particle velocity to estimate the direction of arrival (DOA) of events. These will be compared with DOA-estimates from beamforming of 1) the pressure data, 2) the particle velocity data and 3) pressure data of a nearby 6-element microbarometer array. Furthermore, the observed correlation length of wind will be discussed.

TUESDAY MORNING, 1 JULY 2008 ROOM 362/363, 8:00 A.M. TO 1:20 P.M.

Session 2aPAd

Physical Acoustics: Photoacoustics I

Gerald Diebold, Cochair
Department of Chemistry, Brown University, Providence, Rhode Island 02912, USA

Christ Glorieux, Cochair
Lab. ATF, Katholieke Universiteit Leuven, Celestijnenlaan 200D, Leuven, B-3001, Belgium

Invited Papers

8:00

2aPAd1. On the optical generation and detection of high frequency ultrasounds: thermal and non-thermal processes. Pascal Ruello (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, pascal.ruello@univ-lemans.fr), Philippe Babilotte (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, philippe.babilotte.etu@univ-lemans.fr), Shuo Zhang (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, Shuo.Zhang@insp.jussieu.fr), Denis Mounier (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, denis.mounier@univ-lemans.fr), Jean-Marc Breteau (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, jean-marc.breteau@univ-lemans.fr), Mathieu Edely (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, mathieu.edely@univ-lemans.fr), Patrick Laffez (Université François Rabelais, Tours, IUT Blois, 6, place Jean Jaures, 41029 Blois, France, patrick.laffez@univ-tours.fr), Alain Bulou (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, alain.bulou@univ-lemans.fr), Bernard Perrin (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, bernard.perrin@insp.jussieu.fr), Vitali Gusev (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, vitali.goussev@univ-lemans.fr)

We report on the results of the generation and detection by femtosecond laser pulses of the acoustic waves at frequencies of tens to hundreds GHz in semiconductors and in oxides compounds exhibiting phase transition. We focus first on the generation mechanisms involved to achieve opto-acoustic transformations. Particular attention will be paid to the cases where classical thermal effects (thermoelastic coupling) drive the mechanism of generation of acoustic phonons and those where non-thermal effects become significant and sometimes dominant sources of acoustic phonons field. In the latter cases, we will especially focus on the phonons generation based on photo-induced modifications of microscopic internal electric fields (potential deformation) and also on the use of photoexcited carriers dynamics (carriers recombination) as a tuning parameter of the photo-generated ultrasounds spectrum. Secondly, we give the examples demonstrating that the choice of optical frequency for ultrasound detection influences not only the amplitude of the detected signal but provides an opportunity to detect separately high or low frequencies in the spectrum of ultrasounds. This study was supported by CPER and ANR project No. BLAN06-3-136284.

8:20

2aPAd2. Optical detection of longitudinal and shear acoustic waves with laser picosecond acoustics. Osamu Matsuda (Division of Applied Physics, Graduate School of Engineering, Hokkaido University, 060 8628 Sapporo, Japan, omatsuda@eng.hokudai.ac.jp), Oliver B. Wright (Division of Applied Physics, Graduate School of Engineering, Hokkaido University, 060 8628 Sapporo, Japan, olly@eng.hokudai.ac.jp), David H. Hurley (Materials Characterization Department, Idaho National Laboratory, P.O.Box 1625, Idaho Falls, ID 83415-2209, USA, David.Hurley@inl.gov), Vitali Gusev (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, vitali.goussev@univ-lemans.fr), Ken'Ichi Shimizu (University Chemical Laboratory, Keio University, 4-1-1 Hiyoshi, Kohoku-ku, 223-8251 Kanagawa, Japan, shimizuk@econ.keio.ac.jp)

The absorption of picosecond light pulses in a medium can generate sub-THz acoustic waves. These cause a transient optical reflectance change that can be monitored by delayed probe light pulses. This technique, termed laser picosecond acoustics, can be used for the nondestructive evaluation of the physical properties of thin films and substrates. This paper describes a general method for quantitative analysis of such reflectance changes. It is applicable to multiple anisotropic layers that may be opaque or transparent. Longitudinal or shear acoustic waves propagating along the stacking direction of the multilayers modulate the dielectric permittivity anisotropically and inhomogeneously through the photoelastic effect, through local rotation, or through the surface and interface displacements. We describe how the optical reflectance for obliquely incident probe light can be calculated for the modulated medium.

We then demonstrate the method with reference to experimental results for a sample consisting of a silica film on a zinc substrate in which both longitudinal and shear acoustic waves are generated and detected. The analysis yields the film thickness, sound velocity, and photoelastic tensor components, for example. The method is also applicable to various light scattering problems involving the inhomogeneous modulation of optical properties such as in photothermal experiments.

Contributed Papers

8:40

2aPAd3. Experimental studies of generation and propagation of high frequency acoustic waves in various solid materials using ultraviolet picosecond laser pulses. Eirini Tzianaki (Department of Electronics, Technological Educational Institute of Crete, Romanou 3, 73133 Chania, Greece, i.tzianaki@mail.chania.teicrete.gr), Michael Tatarakis (Department of Electronics, Technological Educational Institute of Crete, Romanou 3, 73133 Chania, Greece, m.tatarakis@chania.teicrete.gr), Makis Bakarezos (Department of Music Technology and Acoustics, Technological Educational Institute of Crete, 1 E. Daskalaki Str., 74100 Rethymnon, Greece, bakarezos@stef.teicrete.gr), Maria Elefteriou (Department of Music Technology and Acoustics, Technological Educational Institute of Crete, 1 E. Daskalaki Str., 74100 Rethymnon, Greece, marel@physics.uoc.gr), Nektarios Papadogiannis (Department of Music Technology and Acoustics, Technological Educational Institute of Crete, 1 E. Daskalaki Str., 74100 Rethymnon, Greece, npapadogiannis@stef.teicrete.gr), Spyros Kazianis (Department of Physics, University of Ioannina, 45110 Ioannina, Greece, skaziannis@in.gr), Costas Kosmidis (Department of Physics, University of Ioannina, 45110 Ioannina, Greece, kkosmidis@uoi.gr), Andreas Lyras (Department of Physics, University of Ioannina, 45110 Ioannina, Greece, alyras@uoi.gr)

The generation of high frequency acoustic waves by picosecond laser pulses in the ultraviolet region and their detection by optical interferometric schemes, is presented. The two main acoustical modes, longitudinal and shear are clearly apparent in the time resolved spectra of solid materials, for various absorbing energies, extending from the thermoelastic to the ablative regime. The ultraviolet light is strongly absorbed by insulator materials like Pyrex and thus strong elastic waves are produced. From the time separation of the longitudinal waves we have deduced values for the speed of sound in various materials and of different thickness, that are in very good agreement with those reported in the literature. Also the time bandwidth of the sound waves is measured and significant differences, originating from different sample thickness, are apparent.

9:00

2aPAd4. Effect of photoacoustic generation mechanisms on transient grating measurements of thin films on silicon. Alexei A. Maznev (Department of Applied Physics, Hokkaido University, Kita 13 Nishi 8, Kita-ku, 060-8628 Sapporo, Japan, alexei.maznev@gmail.com)

Laser induced transient gratings technique is widely used for non-contact measurements of surface acoustic waves (SAWs). One major application of such measurements is characterization of thin films used in microelectronics. A peculiar phenomenon has been observed in transient gratings measurements of thin films on a silicon substrate: if the film is transparent, there is a systematic discrepancy between the measured and expected SAW velocity values that cannot be accounted for by instrumental errors. However, as soon as the sample is coated by an opaque metal film, the discrepancy disappears. In this presentation, we will show that the origin of the effect is in the competition of thermal and electronic mechanisms of the optical generation of sound in silicon. We will also describe another effect of the same origin observed on very thin semi-transparent metal films: at a certain film thickness, the SAW signal vanishes as the thermal expansion of the film and the contraction of silicon caused by photoexcited carriers cancel each other.

9:20

2aPAd5. Thickness measurement of submicron metal coatings on transparent substrate by laser optoacoustic method. Ivan Pelivanov (International Laser Center of Moscow State University, Vorob'yovy Gory, 1, 119992 Moscow, Russian Federation, pelivanov@ilc.edu.ru), Daria Kopylova (International Laser Center of Moscow State University, Vorob'yovy Gory, 1, 119992 Moscow, Russian Federation, dskopylova@mail.ru), Nataliya Podimova (Moscow State University, MSU, 1, building 2, GSP-2, Leninskiye Gory, 119992 Moscow, Russian Federation, npodymova@mail.ru), Alexander Karabutov (Moscow State University, MSU, 1, building 2, GSP-2, Leninskiye Gory, 119992 Moscow, Russian Federation, akarabutov@gmail.com)

New nondestructive method for measurement of the thickness of submicron metal coatings on a transparent dielectric substrate is developed. Theoretical background includes the consecutive solution of the thermal and acoustic problems. The experimental part is based on the measurement of frequency dependence of the laser-ultrasound excitation efficiency on the metal coating thickness for the system where the coating is covered by a transparent liquid. Three chrome coatings of various thicknesses (0.2, 0.3 and 0.6 micrometers) deposited on a quartz substrate were tested experimentally. A rough estimate of their thickness was obtained from the sputtering time. Two different experimental modes were used: forward mode (laser pulses irradiate the metal film through the substrate, excited acoustical transients are detected in the liquid) and backward mode (both laser irradiation of the film and detection of excited acoustical transients are performed in the liquid). Spectral dependencies of the thermo-optical transformation efficiency were calculated analytically and obtained experimentally for both signal detection modes. The values of the film thicknesses were determined by the least squares fitting of the theoretical curves to experimental data. It is demonstrated, that the developed optoacoustic method can be used for metal coatings thickness measurement in the range of 50 nm - 5 microns with inaccuracy of 50 nm.

9:40-10:00 Break

Invited Papers

10:00

2aPAd6. Acoustic phenomena in porous media studied by transient grating spectroscopy: a critical test of the Biot theory. Riccardo Cucini (European Lab. for Non-Linear Spectroscopy (LENS), Univ. di Firenze, via Nello Carrara 1, 50019 Sesto Fiorentino (Fi), Italy, cucini@lens.unifi.it), Andrea Taschin (European Lab. for Non-Linear Spectroscopy (LENS), Univ. di Firenze, via Nello Carrara 1, 50019 Sesto Fiorentino (Fi), Italy, taschin@lens.unifi.it), Paolo Bartolini (European Lab. for Non-Linear Spectroscopy (LENS), Univ. di Firenze, via Nello Carrara 1, 50019 Sesto Fiorentino (Fi), Italy, bart@lens.unifi.it), Renato Torre (European Lab. for Non-Linear Spectroscopy (LENS), Univ. di Firenze, via Nello Carrara 1, 50019 Sesto Fiorentino (Fi), Italy, torre@lens.unifi.it)

The propagation of sound in a porous solid filled by a liquid can be described by a phenomenological model introduced by M. A. Biot, that is still extensively used to predict the essential acoustic properties of a wide variety of porous media. Nevertheless testing of the Biot theory at ultra/hypersonic frequencies and in porous media with nanometric scale heterogeneities remains an open issue. We

studied the propagation of acoustic waves in two liquid-filled porous glasses by heterodyne detected transient grating experiments [1]. A test of the Biot theory is presented under two new regimes, namely, for a new frequency range up to 1.3 GHz and for porous media with nanometric scale heterogeneities [2]. We show that the Biot theory describes the sound velocity data correctly, but does not account for the acoustic attenuation. We suggest that the acoustic damping is mainly due to the dissipation mechanisms intrinsic of the matrix and the liquid which are not accounted for in the theory. [1] Time-resolved spectroscopy of complex liquids, edited by Torre R. (Springer, New York) 2008. [2] A. Taschin, R. Cucini, P. Bartolini, R. Torre, Europhys. Lett. In press.

10:20

2aPAd7. Laser ultrasonic study of Lamb wave propagation in different film loading configurations. Xiaodong Xu (Modern Acoustics, Institute of Acoustics, Nanjing University, 210093 Nanjing, China, xdxu@nju.edu.cn), G. Shkerdin (Laboratorium voor Akoestiek en Thermische Fysica - Departement Natuurkunde en Sterrenkunde - Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, shkerdin@yahoo.com), Christ Glorieux (Lab. ATF, Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, christ.glorieux@fys.kuleuven.be)

A study is performed concerning the propagation Lamb waves in technologically and scientifically interesting configurations. In a first configuration a thin film was loaded with different liquids on its two sides. Three experimental laser ultrasonic configurations, line excitation with scanning detection, grating excitation with single point detection, and grating excitation with scanning probe beam, were implemented. All experimental results were found to be consistent between experimental methods and with theoretical predictions. The configuration can be implemented in monitoring applications where elastic properties of liquids that are kept in containers can be determined via the outside container wall. In a second configuration laser excited and detected Lamb waves in a rubber-metal bi-layer were used to determine the highly damped elastic properties of rubber.

Contributed Papers

10:40

2aPAd8. Zero-group velocity modes and local vibrations of an elastic plate. Claire Prada (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, claire.prada-julia@espci.fr), Dominique Clorennec (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, dominique.clorennec@espci.fr), Daniel Royer (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, daniel.royer@espci.fr)

Elastic plates or cylinders can support guided modes with zero group velocity (ZGV) at a nonzero value of the wave number. The conditions required for the existence of ZGV Lamb modes in isotropic plates are discussed. It is shown that these modes appear in a range of Poisson's ratio over the value for which the cut-off frequency curves of modes belonging to the same family intercept, i.e for a bulk wave velocity ratio equal to a rational number. An interpretation of this phenomenon in terms of a strong repulsion between a pair of modes having a different parity in the vicinity of the cutoff frequencies is given. Using laser-based ultrasonic techniques, we experimentally investigate some properties of these ZGV Lamb modes: resonance, backward wave propagation, interference between backward and forward waves. Experiments performed with materials of various Poisson's ratio demonstrate that the resonance spectrum of an unloaded elastic plate, locally excited by a laser pulse, is dominated by the ZGV Lamb modes. From these local resonance frequencies, thickness variations can be measured accurately and material properties like the Poisson's ratio, bulk acoustic wave velocities or material attenuation can be determined without any mechanical contact.

11:00

2aPAd9. Mechanical contacts probed with picosecond ultrasonics. Thomas Dehoux (Division of Applied Physics, Graduate School of Engineering, Hokkaido University, 060-8628 Sapporo, Japan, dehoux@eng.hokudai.ac.jp), Motonobu Tomoda (Division of Applied Physics, Graduate School of Engineering, Hokkaido University, 060-8628 Sapporo, Japan, mtomoda@eng.hokudai.ac.jp), Oliver B. Wright (Division of Applied Physics, Graduate School of Engineering, Hokkaido University, 060 8628 Sapporo, Japan, olly@eng.hokudai.ac.jp)

We investigate the contact between a thin metal film and a ball bearing indented to nanometre depths using picosecond ultrasonics. The area of contact is acoustically imaged to micron spatial resolution using GHz acoustic pulses produced by ultrashort pulsed optical excitation of the film through a transparent substrate. In particular, acoustic echoes are detected through transient optical reflectance changes that are monitored by probe optical pulses. In this way we image the acoustic reflection coefficient at the interface between the film and the indenter. In addition, by imaging the arrival time of the acoustic echoes we determine the penetration profile of the indenter to nanometre precision. Furthermore, imaging the transient thermoreflectance gives a different means for measuring the contact area through spatial variations in thermal diffusion. We thus demonstrate that picosecond ultrasonics and thermoreflectance provide powerful tools for the non-contact evaluation of mechanical contacts. These techniques could be applied to the in situ characterization of contact interfaces between machine elements.

Invited Paper

11:20

2aPAd10. Photoacoustic imaging and laser-ultrasonics using Fourier domain reconstruction methods. Peter Burgholzer (Upper Austrian Research, Hafenstr. 47, 4020 Linz, Austria, peter.burgholzer@uar.at), Thomas Berer (Upper Austrian Research, Hafenstr. 47, 4020 Linz, Austria, thomas.berer@uar.at), Bernhard Reitinger (Upper Austrian Research, Hafenstr. 47, 4020 Linz, Austria, bernhard.reitinger@uar.at), Robert Nuster (Karl-Franzens-Universität Graz, Universitätsplatz 5, 8010 Graz, Austria, ro.nuster@uni-graz.at), Günther Paltauf (Karl-Franzens-Universität Graz, Universitätsplatz 5, 8010 Graz, Austria, guenther.paltauf@uni-graz.at)

Laser-ultrasonics as well as photoacoustic imaging use optically generated acoustic waves detected at the sample surface to image its interior. In laser-ultrasonics a laser pulse is absorbed at the sample surface generating an ultrasound pulse that propagates into the sample, is subsequently reflected at internal structures, and finally detected at the surface by an interferometer. In photoacoustic imaging ultrasound is generated by heating of light-absorbing structures inside of an optical semitransparent sample. The goal in photoacoustic imaging is to recover the spatial distribution of the absorbed energy density inside the sample from the acoustic pressure signals measured outside the sample (photoacoustic inverse problem). Fourier reconstruction is based on the decomposition into plane waves and is a fast and efficient method used in photoacoustic imaging. Interpolation is needed when signal Fourier components are mapped to

source Fourier components. We have shown that the synthetic aperture focusing technique (SAFT) in frequency domain, which needs no interpolation, and the Fourier reconstruction method are mathematically equivalent if the step size of the spatial discretization goes to zero. Both imaging methods are compared using simulated data and measurement data acquired with our interferometer set-up. This work has been supported by the Austrian Science Fund (FWF), project P18172-N02 and project L418-N20.

Contributed Papers

11:40

2aPAd11. Stroboscopic interferometric full-field imaging of laser-induced surface acoustic waves. Bart Sarens (Laboratorium voor Akoestiek en Thermische Fysica - Departement Natuurkunde en Sterrenkunde - Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, bart.sarens@fys.kuleuven.be), Osamu Matsuda (Division of Applied Physics, Graduate School of Engineering, Hokkaido University, 060 8628 Sapporo, Japan, omatsuda@eng.hokudai.ac.jp), Xiaodong Xu (Modern Acoustics, Institute of Acoustics, Nanjing University, 210093 Nanjing, China, xdxu@nju.edu.cn), Georgios Kalogiannakis (Laboratorium voor Akoestiek en Thermische Fysica - Departement Natuurkunde en Sterrenkunde - Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, Georgios.Kalogiannakis@vub.ac.be), Robbe Salenbien (Lab. ATF, Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, robbe.salenbien@fys.kuleuven.be), Renaud Côte (Lab. ATF, Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, renaud.cote@fys.kuleuven.be), Christ Glorieux (Lab. ATF, Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, christ.glorieux@fys.kuleuven.be)

Surface acoustic waves (SAW) have the interesting property that they specifically interact with surface and sub-surface regions, rendering them suitable for non-contact investigation of sub-surface properties and heterogeneities. Transmission, reflection and diffraction effects of SAW propagation are analysed in order to reveal information on the region over which they propagate. Laser excitation allows to efficiently excite SAW with short wavelengths, enhancing both the lateral and the depth resolution. Typically information on the waves is collected by scanning a part of the surface of interest with a fast laser probe, using the surface displacement or slope as a real time witness of the wave field. Here we present results obtained by a full field imaging technique [1], in which the sample is repetitively excited by a pulsed pump laser, and the wave displacement field is stroboscopically gathered by illuminating the sample surface with an electronically delayed pulsed probe laser, whose displacement induced optical path variations are visualized by interferometrically analyzing it. In this work, the technique is applied on different samples, varying the geometry of the pump laser beam pattern, the interferometric configuration, and the heterogeneities of the sample. [1] ″Phase mask-based interferometer: operation principle, performance, and application to thermo-elastic phenomena″, C. Glorieux, J.D. Beers, E.H. Bentefour, K. Van de Rostyne and K.A. Nelson, Rev. Sci. Instrum. **75**(9), 2906-2920 (2004).

12:00

2aPAd12. Laser-ultrasonic defectoscope with focused transducer. Alexander Karabutov (Moscow State University, MSU, 1, building 2, GSP-2, Leninskiye Gory, 119992 Moscow, Russian Federation, akarabutov@gmail.com), Nataliya Podimova (Moscow State University, MSU, 1, building 2, GSP-2, Leninskiye Gory, 119992 Moscow, Russian Federation, npodymova@mail.ru)

The novel focused laser-ultrasonic transducer for ultrasonic microscopy of constructive materials is presented. Short and sharp ultrasonic pulse is produced by optoacoustic effect - excitation of ultrasound by absorption of nanosecond laser pulse in special layer. Absorbing layer is integrated with the wide-band ultrasonic receiving system into common unit - laser-ultrasonic transducer. Ultrasound beam is focused by acoustical lens with relatively low numerical aperture (convergence angle - 15°, focal distance - 20 mm). An acoustic impedance of the lens matches that of the laser-ultrasonic transducer. The duration of the irradiated ultrasonic pulse was ~0.1 μs. The focused laser-ultrasonic transducer was mounted on 3-D linear translation system. The object under control and the transducer was placed in a tank with distilled water. The transducer was moved across the object with 2.5 μm step. The caustic of the probe ultrasonic beam was analyses by sliding the transducer across sharp beard of a knife. The reflected ultrasonic signal was recorded by 12-bit ADC. The diameter of the focal area was 0.53 mm. The length of caustic of ultrasonic was 28 mm. The amplitude of the probe ultrasonic pulse was ~20 kPa at laser energy 0.1 mJ. The dynamic range of ultrasonic trace measurement exceeded 90 dB.

12:20

2aPAd13. Acoustic waves generated by a laser point pulse in a micrometric fiber. Damien Segur (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, d.segur@lmp.u-bordeaux1.fr), Alexander Shuvalov (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, a.shuvalov@lmp.u-bordeaux1.fr), Yong Dong Pan (Institute of Acoustics, Tongji University, 200092 Shangai, China, ypan@mail.tongji.edu.cn), Nikolay Chigarev (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, Nikolay.Chigarev@univ-lemans.fr), Clément Rossignol (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, c.rossignol@lmp.u-bordeaux1.fr), Bertrand Audoin (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, b.audoin@lmp.u-bordeaux1.fr)

Having emerged in the 1980s, the laser ultrasonics technique with its non-contact generation and detection process overpasses the difficulties of coupling piezoelectric transducers with curved surfaces. To date, the authors [1] have been interested in acoustic generation for cylinders opaque at a given laser wavelength and for the acoustic source located at the cylinder surface. In this presentation, assuming point focusing of the laser pulses, we propose a three-dimensional (3D) semi-analytical model for acoustic waves generation and propagation in a partly transparent isotropic cylinder. First, the radial displacement at any position on the free surface is derived, in a 3D Fourier domain, for an inner point source. The response to a volume-source distribution along a radius is obtained as a convolution of the above Green function with the corresponding source distribution caused by optical absorption. Three inverse transforms are then applied to obtain the radial displacement at the cylinder surface. Picosecond ultrasonics experiments are performed on different micrometric fibers and compared with calculated waveforms for different optical absorptive properties. References [1] Y.D. Pan, C. Rossignol and B. Audoin, Appl. Phys. Lett. **82**, 4379 (2003).

12:40

2aPAd14. Simulations of Thermally Induced Photoacoustic Wave Propagation Using a Pseudospectral Time-Domain Method. Yae-Lin Sheu (Dept. Electrical Engineering, National Taiwan University, No.1, Sec. 4, Roosevelt Road, 106 Taipei, Taiwan, b88901147@ntu.edu.tw), Pai-Chi Li (Institute of Biomedical Electronics and Bioinformatics, National Taiwan University, No.1, Sec. 4, Roosevelt Road, 106 Taipei, Taiwan, paichi@cc.ee.ntu.edu.tw)

Physical models used to evaluate thermally induced photoacoustic waves in biomedical applications are mostly approximations based on certain hypotheses, such as the thermal and stress confinements, for the sake of obtaining analytical results. On the other hand, using numerical methods to solve the general photoacoustic wave equations gives detailed information of wave phenomena without making as many assumptions. The photoacoustic wave generated by thermal expansion involves the heat conduction theorem and the state, continuity, and Navier-Stokes equations. In this study a numerical approach was developed in 2.5D axis-symmetric cylindrical coordinates using a pseudospectral time-domain (PSTD) scheme. The method is efficient for large scale simulations in that only two grids for the smallest

wavelength are required, where in conventional methods 10~20 grids are typically needed. The numerical techniques include Berenger's perfectly matched layers (PMLs) for free wave simulations, and linear-perturbation analytical solutions are used to validate the simulation results. The numerical results using 2 grids for the minimum wavelength in simulation domain agree with theory to within an error of 7×10^{-3} in the absolute differences. On the other hand, conventional methods such as finite-difference time-domain method requiring 10 grids result in an error of 1.3×10^{-3}.

Invited Paper

1:00

2aPAd15. Laser ultrasonic study of Lamb wave propagation in different film loading configurations. Xiaodong Xu (Modern Acoustics, Institute of Acoustics, Nanjing University, 210093 Nanjing, China, xdxu@nju.edu.cn), G. Shkerdin (Laboratorium voor Akoestiek en Thermische Fysica - Departement Natuurkunde en Sterrenkunde - Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, shkerdin@yahoo.com), Christ Glorieux (Lab. ATF, Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, christ.glorieux@fys.kuleuven.be)

A study is performed concerning the propagation Lamb waves in technologically and scientifically interesting configurations. In a first configuration a thin film was loaded with different liquids on its two sides. Three experimental laser ultrasonic configurations, line excitation with scanning detection, grating excitation with single point detection, and grating excitation with scanning probe beam, were implemented. All experimental results were found to be consistent between experimental methods and with theoretical predictions. The configuration can be implemented in monitoring applications where elastic properties of liquids that are kept in containers can be determined via the outside container wall. In a second configuration laser excited and detected Lamb waves in a rubber-metal bi-layer were used to determine the highly damped elastic properties of rubber.

TUESDAY MORNING, 1 JULY 2008 ROOM 241, 8:00 A.M. TO 8:00 P.M.

Session 2aPPa

Psychological and Physiological Acoustics and ASA Committee on Standards: Applications of Psychoacoustics I

Patricia Davies, Cochair
Ray W. Herrick Lab., School of Mechanical Engineering, Purdue Univ., 140 S. Martin Jischke Drive, West Lafayette, IN 47907-2031, USA

Hugo Fastl, Cochair
AG Technische Akustik, MMK, TU München, Arcisstr. 21, München, 80333, Germany

Invited Papers

8:00

2aPPa1. Determination of filtering parameters for dichotic-listening binaural hearing aids. Yôiti Suzuki (R.I.E.C., Tohoku University, 2-1, Katahira, Aoba-ku, 980-8577 Sendai, Japan, yoh@ais.riec.tohoku.ac.jp), Atsunobu Murase (Panasonic Shikoku Electronics Co., Ltd, 600, Saedo-cho, Tsuzuki-ku, 224-8539 Yokohama, Japan, murase.atsunobu@jp.panasonic.com), Motokuni Itoh (Matsushita Electric Industrial Co., Ltd, 600, Saedo-cho, Tsuzuki-ku, 224-8539 Yokohama, Japan, ito.gempo@jp.panasonic.com), Shuichi Sakamoto (R.I.E.C., Tohoku University, 2-1, Katahira, Aoba-ku, 980-8577 Sendai, Japan, saka @ais.riec.tohoku.ac.jp)

Sensorineural hearing-impaired people have difficulty in hearing sounds not only because of the increased hearing threshold, but also because of their reduced dynamic range of hearing (loudness recruitment), as well as the large and extensive masking resulting from the reduced frequency selectivity, especially masking by which middle-frequency and high-frequency components are masked by the intense low-frequency component, the so-called upward spread of masking. Dichotic listening, listening to complementary filtered speech signals given to the two ears, has been proposed to cope with these problems. We introduce a dichotic listening technique for binaural hearing aids and the relationship between the effect of this technique and auditory characteristics of hearing-impaired people. The results of listening tests imply that some relationship exists between the width of the auditory filter and the effect of this technique. Moreover, intelligibility test results obtained using the low-frequency-boosted sounds suggest that this dichotic listening reduced the upward spread of masking. We are applying this technique to hearing aid systems and have begun evaluating the performance of this hearing aid.

8:20

2aPPa2. Binaural auralization based on spherical-harmonics beamforming. Wookeun Song (Brüel & Kjær Sound & Vibration Measurement A/S, Skodsborgvej 307, DK-2850 Nærum, Denmark, wksong@bksv.com), Wolfgang Ellermeier (Institut für Psychologie, Technische Universität Darmstadt, Alexanderstraße 10, D-64283 Darmstadt, Germany, ellermeier@psychologie.tu-darmstadt.de), Jørgen Hald (Brüel & Kjær Sound & Vibration Measurement A/S, Skodsborgvej 307, DK-2850 Nærum, Denmark, JHALD@bksv.com)

The binaural auralization of a 3D sound field using spherical-harmonics beamforming (SHB) techniques was investigated and compared with the traditional method using a dummy head. The new procedure was verified by comparing simulated room impulse responses with directly measured ones both monaurally and binaurally. The objective comparisons show that there is good agreement in the frequency range between 0.1 to 6.4 kHz. Psychoacoustic attributes of multi-channel reproduced sounds were measured in a listening experiment to validate the method subjectively. The results show that subjective ratings of the width, spaciousness and preference of different audio reproduction modes auralized based on SHB were not significantly different from those obtained for dummy head measurements. Thus binaural synthesis using SHB may be a useful tool to reproduce a 3D sound field binaurally while saving considerably on measurement time because head rotation can be simulated based on a single recording.

8:40

2aPPa3. Simulation of wave field synthesis. Florian Völk (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, florian.voelk@mytum.de), Josef Konradl (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, koj@mmk.ei.tum.de), Hugo Fastl (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, fastl@mmk.ei.tum.de)

Wave field synthesis utilizes a large number of loudspeakers to generate a desired wave field. It therefore is necessary to drive each speaker with an independent signal, which requires as many amplifier and soundcard channels as there are loudspeakers. These enormeous hardware costs make research and development very expensive and time consuming. Additionally, different rooms influence the wave field synthesis arrays in different ways. For this reason a simulation technique is of advantage which permits the evaluation of the perceived properties of arbitrary wave field synthesis configurations without the need to physically construct them. This paper proposes a simulation system capable to simulate wave field synthesis systems in different rooms based on physical measurements of one loudspeaker in each room. The techniques used are presented, and possiblities as well as limits of the system are discussed.

9:00

2aPPa4. The next generation of artificial heads. Janina Fels (Institute of Technical Acoustics, RWTH Aachen University, Neustr. 50, 52056 Aachen, Germany, Janina.Fels@akustik.rwth-aachen.de), Michael Vorlaender (Institute of Technical Acoustics, RWTH Aachen University, Neustr. 50, 52056 Aachen, Germany, mvo@akustik.rwth-aachen.de)

Standardised artificial heads are vital means when it comes to describing the binaural transmission from the sound field into the ear canal or rather the eardrum. In recent years numerous fields of application were created ranging from room acoustics, to product sound design or telecommunications, all based on the well-known KEMAR standard IEC TR 959. In the meantime, however, it has become a well-known fact, that a) specific artificial heads with natural heads or replicas in hearing experiments are superior to standard artificial heads as far as the quality of spatial hearing is concerned and that b) the standardised heads do not comply with the dimensions of an average population (the standard heads are too small). In this contribution first of all today's situation will be assessed and then a possible way will be outlined that could lead to a new future artificial head standard. This includes an adequate match with adult population for various continents, and also new approaches such as children-size artificial heads to measure and fit hearing aids or for new measurement techniques for classroom acoustics.

9:20

2aPPa5. Auditory memory and evaluation of environmental sounds. Sonoko Kuwano (Graduate School of Human Sciences, Osaka University, 1-2 Yamadaoka, 565-0871 Suita, Osaka, Japan, kuwano@see.eng.osaka-u.ac.jp), Seiichiro Namba (2-7-5-604 Obana, 666-0015 Kawanishi, Hyogo, Japan, QZW00041@nifty.com), Tohru Kato (Otemon Gakuin University, 2-1-15 Nishiai, 567-8502 Ibaraki, Osaka, Japan, tkatou@res.otemon.ac.jp)

A series of experiments were conducted concerning the memory of environmental sounds. Twelve kinds of sound were presented with soft background noise of about 6 min to participants. They were asked to recall or recognize the sound sources and to judge the loudness of the recalled or recognized sounds some period after they listened to the sounds. The recalled or recognized loudness was examined in relation to the length of the period between the presentation of sounds and the judgment of loudness, the method to measure the memory, LAeq of each sound source, etc. The results suggest that it is possible to judge the loudness of the memorized sounds and that the judgments seems reliable.

9:40

2aPPa6. Comparison of subjective impression of copy machine noise between Japanese, American and German participants. Tatsuya Furukawa (Ricoh Co., Ltd., 16-1 Shinei-cho, Tsuzuki-ku, 224-0035 Yokohama, Japan, furukawa@rdc.ricoh.co.jp), Osamu Takehira (Ricoh Co., Ltd., 16-1 Shinei-cho, Tsuzuki-ku, 224-0035 Yokohama, Japan, osamu.takehira@nts.ricoh.co.jp), Masaki Nagamiya (Ricoh Co., Ltd., 16-1 Shinei-cho, Tsuzuki-ku, 224-0035 Yokohama, Japan, masaki.nagamiya@nts.ricoh.co.jp), Sonoko Kuwano (Graduate School of Human Sciences, Osaka University, 1-2 Yamadaoka, 565-0871 Suita, Osaka, Japan, kuwano@see.eng.osaka-u.ac.jp), Seiichoro Namba (Graduate School of Human Sciences, Osaka University, 1-2 Yamadaoka, 565-0871 Suita, Osaka, Japan, namba.seiichiro@nifty.com), Hugo Fastl (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, fastl@mmk.ei.tum.de)

Psychological evaluations of copy machine noise are in progress to lower acoustical noise and to avoid disturbing the office environment. Psychological experiments were made for Japanese, American and German participants to examine cultural differences in sensitivity to copy machine noise using the Semantic Differential. Seventeen copy machine and laser printer sounds were used for the experiments. The results were analyzed statistically. The experiments revealed that, just as with Japanese, Americans and Germans were mostly sensitive to ″sound pressure levels.″ However, the second influential factor for Japanese participants was ″impulsiveness″ but ″sharpness″ for Americans and Germans. Evaluation formulas for ″pleasing″ noise using physical values were decided for each country. This suggests that copy machine noise will be evaluated without psychological experiments. The information gathered will possibly be used to adapt copy machine noise to levels that are comfortable to users in different countries.

10:00

2aPPa7. Applications of psychoacoustics to information technology products. Willem M. Beltman (Intel Corporation, 2111 NE 25th Avenue, M/S JF2-86, Hillsboro, OR 97124, USA, willem.m.beltman@intel.com), Rina A. Doherty (Intel Corporation, 5200 NE Elam Young Parkway, Hillsboro, OR 97124, USA, rina.a.doherty@intel.com), Eric Salskov (University of Twente, Dept. Mechanical Engineering, P.O. Box 217, 7500 AE Enschede, Netherlands, eric.salskov@intel.com), Philip J. Corriveau (University of Twente, Dept. Mechanical Engineering, P.O. Box 217, 7500 AE Enschede, Netherlands, philip.j.corriveau@intel.com), Doug Gabel (Intel Corporation, 2111 NE 25th Avenue, M/S JF2-86, Hillsboro, OR 97124, USA, doug.gabel@intel.com), Eric Baugh (Intel Corporation, 15400 NW Greenbrier Parkway, Beaverton, OR 97006, USA, eric.baugh@intel.com)

Emerging usage models for computing devices require low acoustic noise, for example in home entertainment systems. Studies have shown that not only the overall level, but also the psychoacoustic aspects matter. This paper provides an overview of testing techniques that are used in the information technology industry and outlines two specific case studies. First, an extensive subjective psychoacoustic study was designed and conducted in multiple geographies to determine the aspects of sound that best describe the annoyance to sound from information technology products in a home type environment. Over 200 participants in four countries participated in this carefully controlled experiment and rated typical steady state sounds on a 5 point annoyance scale. The relevant sound quality metrics were extracted and geographical variations quantified. Second, in a paired comparison study the influence of modulation on annoyance was investigated by superimposing different frequency and amplitude modulated sounds onto a baseline sound. The results indicate that modulation can have a significant effect on subjective perception.

10:20-10:40 Break

10:40

2aPPa8. Identification of transient events from a hard disk drive using non-stationary loudness. Dave Ali (Western Digital, 5863 Rue Ferrari, San Jose, CA 95127, USA, dave.ali@wdc.com)

The Hard Disk Drive (HDD) industry has been pushed so much to lower its limits of acoustic emissions by OEMs and customers that the levels of most HDDs flirt with the threshold of hearing and in some cases go below. With these much reduced levels come different problems and complaints from customers, namely transient events; latching, de-latching, spin up, spin down, etc. The difference in amplitude and duration of these events typically stand out to the human ear above the steady state nature of the HDD during idle and sometimes when it's active. This paper shows a method of characterizing these events with confidence to help in designing a better product for the industry.

11:00

2aPPa9. Automotive tire/road sound quality. Gabriella Cerrato Jay (Sound Answers Inc, 4856 Alton Drive, Suite 100, Troy, MI 48085, USA, gabriella.cerratojay@soundanswers.net), Todd Freeman (Sound Answers Inc, 4856 Alton Drive, Suite 100, Troy, MI 48085, USA, todd.freeman@soundanswers.net), Chris Raglin (Cooper Tire & Rubber Company, 701 Lima Avenue, Findlay, OH 45840, USA, caraglin@coopertire.com), Timothy Carson (IAC North America, 47785 W Anchor CT, Plymouth, MI 48170, USA, tcarson@iacna.com)

Tire/road noise and sound quality are increasingly important factors for customer satisfaction. As vehicle interior sound levels decrease, the noise from tires and their interaction with the road become more noticeable. Both tire and vehicle manufacturers need to assess as early as possible the impact of any tire-vehicle combination on perceived interior sound quality. In this paper, we describe two projects in which psychoacoustics concepts were applied to help, on one hand, tire manufacturers to screen for designs likely to generate poor sound quality in vehicle, and, on the other hand, vehicle manufacturers to measure the impact of road noise and interior acoustic treatment on speech transmission in the cabin. In both projects, the activities were driven by the understanding of the psychoacoustic features of tire/road noise and by the need to improve customer satisfaction.

Contributed Paper

11:20

2aPPa10. Acoustical aspects of travel comfort in the aircraft cabin. Ingo Baumann (Oldenburg University, Institute of Physics - Acoustics, Carl-von-Ossietzky Str. 9-11, 26111 Oldenburg, Germany, ingo @aku.physik.uni-oldenburg.de), Sandra Buss (Oldenburg University, Institute of Physics - Acoustics, Carl-von-Ossietzky Str. 9-11, 26111 Oldenburg, Germany, sandra@aku.physik.uni-oldenburg.de), Nils Freese (Oldenburg University, Institute of Physics - Acoustics, Carl-von-Ossietzky Str. 9-11, 26111 Oldenburg, Germany, nils_f@aku.physik.uni-oldenburg.de), Volker Mellert (Oldenburg University, Institute for Physics, 26111 Oldenburg, Germany, volker.mellert@uni-oldenburg.de), Reinhard Weber (Oldenburg University, Institute of Physics - Acoustics, Carl-von-Ossietzky Str. 9-11, 26111 Oldenburg, Germany, Reinhard.Weber@uni-oldenburg.de)

In several European projects tests were conducted during real flights and in aircraft simulators in order to identify important parameters of travel comfort. Physical environmental, physiological and questionnaire data were measured and investigated with respect to human perception. Besides numerous intrinsic quantities affecting health, well-being and awareness of ambience defined measurable environmental parameters influence flight and cabin crews' as well as passengers' perception, psychology and physiology. Sound and vibration are relevant environmental parameters with impact on passenger and crew. Other important influence is caused by e.g. air quality, pressure, local climate. Statistical analysis of the collected data reveals significant correlations between environment and human response for selected groups of test persons. A specific acoustic comfort is not well defined but part of common well-being and comfort. The demand for a general human response model is discussed, which relates ambience and perception.

Invited Papers

11:40

2aPPa11. The influence of speed bumps on perceived annoyance. Anna Preis (Institute of Acoustics, Adam Mickiewicz University, Unultowska 85, 61-614 Poznan, Poland, apraton@amu.edu.pl), Tomasz Kaczmarek (Institute of Acoustics, Adam Mickiewicz University, Unultowska 85, 61-614 Poznan, Poland, tomek@spl.ia.amu.edu.pl), Barbara Griefahn (Institute for Occupational Physiology, Ardeystr. 67, 44139 Dortmund, Germany, griefahn@ifado.de), Truls Gjestland (O.S. Bragstads plass 2, N-7464 Trondheim, Norway, Truls.Gjestland@sintef.no)

Recently, several attempts to use speed bumps as a noise reduction method have been made. Objective analyses of the effect of speed bumps on noise have been shown to result in a rather small reduction of noise. In the present paper the influence of speed bumps on perceived annoyance is investigated. The annoyance rating of a situation in which a passenger car approaches with constant velocity, then decelerates, crosses the bump, accelerates, and then recedes at a constant speed was compared with a car pass-by at a constant velocity without a bump. Three different velocities were analyzed: 40, 50 and 60 km/h, and two types of driving conditions: normal, and aggressive. Listeners judged their annoyance for all the investigated scenarios using the ICBEN scale (0-10) for annoyance assessment. Objective analyses showed a significant reduction of LAeqT in the bump situation for all tested velocities, and for both driving conditions. The results of this psychoacoustic experiment show no effect of the bump on annoyance rating for normal driving conditions. However, in aggressive driving conditions the bump resulted in a significant increase in annoyance. In the light of these results, speed bumps cannot be considered as a noise reduction method.

12:00

2aPPa12. The Adequate Sound Levels for Acoustic Signs for Visually Impaired in the Sound Environment with Ambient Musics from shops. Katsuya Yamauchi (Faculty of Engineering, Nagasaki University, 1-14 Bunkyo-machi, 852-8521 Nagasaki, Japan, yamauchi@cis.nagasaki-u.ac.jp), Koji Nagahata (Faculty of Symbiotic Systems Science, Fukushima University, 1 Kanayagawa, 960-1296 Fukushima, Japan, nagahata@sss.fukushima-u.ac.jp), Mari Ueda (Faculty of Human-Environment, Kyushu University, 6-19-1 Hakozaki, Higashik-ku, 812-8581 Fukuoka, Japan, mari-u@gsd.design.kyushu-u.ac.jp), Shin-Ichiro Iwamiya (Kyushu University, 4-9-1 Shiobaru, Minami-ku, 815-8540 Fukuoka, Japan, iwamiya@design.kyushu-u.ac.jp)

Providing auditory signs for the visually impaired is one of the most effective ways to support their orientation and mobility. Although the use of such sounds by the visually impaired has been revealed qualitatively, the acoustical properties of ideal sound-designs for them have not known sufficiently, even a basic property such as the sound level of these sounds. On the other hand, the advertising sounds from shops such as ambient musics are the one of the typical sounds in the Japanese downtown. These sounds also disturb their sound information listening and make their mobility difficult. Our previous studies revealed the adequate sound levels of acoustic signs under the road traffic noise environment. This study discusses the the effect of the ambient musics from shops on the adequate sound levels of acoustic signs through the psychoacoustical experiment. The results showed that the difference between the adequate sound level of acoustic signs and the environmental noise levels are larger than that under the road traffic noise environment. This trend is thought to be attributed to the frequency characteristics of the acoustical signs and the environmental noise.

12:20

2aPPa13. An algorithm modelling the Irrelevant Sound Effect (ISE). Sabine J. Schlittmeier (Work, Environmental and Health Psychology, Catholic University of Eichstaett-Ingolstadt, Ostenstr. 26-28, 85072 Eichstaett, Germany, sabine.schlittmeier@ku-eichstaett.de), Tobias Weissgerber (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, tobias@p-weissgerber.de), Stefan Kerber (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, stefan@ihr.mrc.ac.uk), Hugo Fastl (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, fastl@mmk.ei.tum.de), Juergen Hellbrueck (Work, Environmental and Health Psychology, Catholic University of Eichstaett-Ingolstadt, Ostenstr. 26-28, 85072 Eichstaett, Germany, juergen.hellbrueck@ku-eichstaett.de)

Verbal short-term memory capacity is reduced significantly during certain background sounds. Remembering a series of digits is significantly impaired by speech or music with prominent staccato passages whereas, for example, music with prominent legato passages does not disturb performance in comparison to silence. This so-called Irrelevant Sound Effect (ISE) occurs although the background sounds are irrelevant with respect to the digit sequences to be remembered. Until now, a multitude of cognitive psychological experiments explored the ISE and collected behavioral performance data during different sound conditions. The talk presents an algo-

rithm which models performance data in ISE experiments, i.e. the detrimental impact of background sounds on memory performance. The data base of this algorithm is about 50 background sounds and corresponding performance data, which have been collected in cognitive psychological experiments at the KU Eichstaett-Ingolstadt. The algorithm is based on the instrumental measuring of the hearing sensation fluctuation strength and is able to reconstruct the performance results in about 90% of cases within the interquartile ranges. The algorithm will be discussed within the scope of cognitive short-term memory models, which claim to explain the ISE and with respect to practical implications.

12:40-2:00 Lunch Break

Invited Papers

2:00

2aPPa14. Fusion and masking threshold of a tone in narrowband noise. Hans Hansen (IRCAM - Sound Perception & Design, 1 place Igor Strawinsky, 75004 Paris, France, Hans_Hansen@gmx.de), Reinhard Weber (Oldenburg University, Institute of Physics - Acoustics, Carl-von-Ossietzky Str. 9-11, 26111 Oldenburg, Germany, Reinhard.Weber@uni-oldenburg.de)

When comparing different pitch phenomena, it is not clear whether the expression pitch strength points towards a unique perceptive phenomenon or object. As many studies concentrate on one special phenomenon, the question, what is actually judged, seems only implicitly answered. Kubovy and Van Valkenburg (2001) define a perceptual object as ″that what is susceptible to figure-ground segregation″. Pitch plays a major role in these grouping processes. In this context the judgment of pitch strength refers to two perceptual cases. The first one is the pitch strength of a tone-in-noise, i.e. the case where the pitch is linked to a separate contour within the narrowband noise (NBN), while the second is the pitch strength of tonal noise. Here, the noise evokes a pitch sensation that is not related to a separate object. In order to explore the transition from case /1 tone-in-noise to case /2 tonal noise, the identification threshold hearing a tone in NBN centered on the tone is determined in a lab experiment. This segregation or fusion threshold is compared to the measured masking threshold for center frequencies 250-4000 Hz octave-wise at 60 dB SPL noise level. The bandwith of the NBN is varied from 50-250 Hz accordingly.

2:20

2aPPa15. Are absolute thresholds and loudness judgements influenced by different colours? Daniel Menzel (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, menzel@tum.de), Elias Faccinelli (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, fac@mmk.ei.tum.de), Hugo Fastl (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, fastl@mmk.ei.tum.de)

Previous experiments showed that images of differently colored trains and sports cars can influence the loudness ratings given by subjects via free magnitude estimation. Red vehicles caused subjects to rate the loudness of simultaneously presented train or car sounds higher relative to green vehicles. To investigate whether these loudness differences correspond to shifts in absolute threshold, subject's threshold in quiet was measured via Békésy-tracking while viewing red, green, and neutral color patches. Also, the influence of color on the loudness of broadband noise was measured using a method of adjustment, in which subjects had to adjust the level of a test sound until it was perceived as loud as a reference sound. In both cases, no influence of color on either absolute threshold or loudness perception measured via adjustment could be found. These results support the hypothesis that, compared to previously used methods like magnitude estimation, methods that require subjects to concentrate on the auditory stimulus do not seem to be as applicable for measurements of audio-visual interactions.

2:40

2aPPa16. An efficient masking model for audio coding exploiting spectro-temporal masking. Steven Van De Par (Philips Research Europe, Digital Signal Processing (MS WO02), High Tech Campus 36, 5656 AE Eindhoven, Netherlands, steven.van.de.par@philips.com), Jeroen Koppens (Philips Research Europe, Digital Signal Processing (MS WO02), High Tech Campus 36, 5656 AE Eindhoven, Netherlands, jeroen.koppens@philips.com), Armin Kohlrausch (Philips Research Europe, Digital Signal Processing (MS WO02), High Tech Campus 36, 5656 AE Eindhoven, Netherlands, armin.kohlrausch@philips.com), Werner Oomen (Philips Research Europe, Digital Signal Processing (MS WO02), High Tech Campus 36, 5656 AE Eindhoven, Netherlands, werner .oomen@philips.com)

Perceptual audio coding achieves part of its coding efficiency by spectrally shaping the quantization noise such that it is masked by the audio signal to be encoded. In order to determine how much quantization noise is allowed within each frequency band and time interval a masking model is used to predict a masking curve specifying the maximally allowed quantization noise level within each frequency band. In most audio coders only spectral masking properties of the audio signal are used. The model by Dau et al. [J. Acoust. Soc. Am. 99, Vol. 3615, 1996] provides an interesting approach to also model temporal masking. Since this model operates as an artificial observer it only predicts whether the quantization noise is audible or not in the presence of the audio signal. In order to determine the most efficient quantization noise shape, the encoder needs to iteratively adapt the noise shape and evaluate each option with the model. This implies a highly computational complex encoding algorithm. In this contribution we will present an efficient masking model based on the Dau et al. model that only requires a single evaluation of the input signal to determine the maximally allowed noise level within each frequency band.

3:00

2aPPa17. Factor analyses of critical-band-filtered speech of British English and Japanese. Kazuo Ueda (Perceptual Psychology Unit, Kyushu University, 4-9-1 Shiobaru, Minami-ku, 815-8540 Fukuoka, Japan, ueda@design.kyushu-u.ac.jp), Yoshitaka Nakajima (Perceptual Psychology Unit, Kyushu University, 4-9-1 Shiobaru, Minami-ku, 815-8540 Fukuoka, Japan, nakajima@design.kyushu-u.ac.jp)

Two-hundred sentences of British English and Japanese, each uttered by 10 native speakers (5 females and 5 males) in each language, were analyzed through 20 bands of critical band filters. Smoothed power fluctuations derived from the filters were submitted to principal component analyses followed by varimax rotation. The first three factors explained 34-37% of variance. One of the factors exhibited two peaks along frequency axis in the standardized scores and two of the factors showed one peak for each. These three factors divided the whole frequency range of speech sound into four bands. The structure of the factors and frequency bands was essentially the same across the two languages. These frequency bands can be used for speech perception in general, because intelligible noise-vocoded speech sounds can be synthesized with the frequency bands.

Contributed Paper

3:20

2aPPa18. A Structuralistic Approach to Acoustic-Auditory Functions of Meaning. Ute Jekosch (Chair of Communication Acoustics, TU Dresden, Helmholtzstr. 10, 01069 Dresden, Germany, ute.jekosch@tu-dresden.de), Ercan Altinsoy (Chair of Communication Acoustics, TU Dresden, Helmholtzstr. 10, 01069 Dresden, Germany, ercan.altinsoy@ias.et.tu-dresden.de), Sebastian Merchel (Chair of Communication Acoustics, TU Dresden, Helmholtzstr. 10, 01069 Dresden, Germany, sebastian.merchel@tu-dresden.de)

In this paper we introduce a methodology of semio-acoustics to get information on how human listeners associate meaning to acoustic-auditory events. We concentrate on identifying cues in the auditory stream listeners base the association of meaning on as well on modelling major characteristics of the reference system of meaning. The methodology we use is closely related to structuralism, an approach that has its origins in semiotics. In principle, structuralism differentiates between creating functions, carrying functions and changing functions of systems of meaning. We concentrate on carrying and changing functions here using the following procedure: a sign-carrier (in our case an acoustic-auditory event the association of meaning is based on) is decomposed into sub-units. By a minimal pair analysis we investigate carrying and changing functions of acoustic-auditory features with regard to the associated meaning. We will introduce the methodology used and discuss first results of a pilot study.

3:40-5:20 Posters

Lecture sessions will recess for presentation of poster papers on various topics in acoustics. See poster sessions for topics and abstracts.

Contributed Papers

5:20

2aPPa19. Optimization of a dual recognition tasks for speech quality assessment. Virginie Durin (France Télécom, 2 avenue Pierre Marzin, 22300 Lannion, France, virginie.durin@orange-ftgroup.com), Laetitia Gros (France Télécom, 2 avenue Pierre Marzin, 22300 Lannion, France, laetitia.gros@orange-ftgroup.com)

This paper deals with perceptive test methodologies to assess speech quality of telecommunication systems. Faced with drawbacks of typical methodologies recommended by ITU-T, a new way to assess speech quality is investigated. The new approach requires collecting reaction times and performances when subjects are achieving tasks involving degraded speech signals; it is shown that reactions times lengthen and performances decrease in a specific task when quality is impaired. The proposed task is a dual task with a digit recognition memory task and a letter recognition task. Three different quality levels are applied to audio signals describing digits and letters. Different experimental designs are examined to reinforce the effect of speech quality on performances and reaction times. The results show significant differences of performances and reaction times between the three quality levels, depending on the experimental design.

5:40

2aPPa20. Electronic pass-through hearing protection and directional hearing restoration integrated in a helmet. Wouter K. Vos (TNO, Kampweg 5, 3769ZG Soesterberg, Netherlands, wouter.vos@tno.nl), Adelbert W. Bronkhorst (TNO, Kampweg 5, 3769ZG Soesterberg, Netherlands, adelbert.bronkhorst@tno.nl), Jan A. Verhave (TNO, Kampweg 5, 3769ZG Soesterberg, Netherlands, jan.verhave@tno.nl)

Compared to standard earplugs, electronic pass-through earplugs provide better sound localisation. Provided that the bandwidth is sufficiently wide and the earplugs do not change the shape of the pinnae. However, when a helmet is worn that partially or completely covers the ears, the directional hearing capability is diminished. We attempt to restore directional hearing when wearing a helmet by attaching a microphone array to the helmet. The signals from the microphone array are filtered with Finite Impulse Response (FIR) filters to recreate an individual or generic open-ear Head Related Transfer Function (HRTF). The filters are designed by minimisation of an error measure in the frequency domain. The error measure incorporates both the log magnitude and the phase differences between the original and the recreated HRTF. The global minimum is found using modern optimisation techniques like Particle Swarm Optimisation (PSO) or Differential Evolution (DE). The total system is evaluated with subject experiments. Participants have to localise sounds and rate the quality of sounds. Independent variables are the number of microphones that should be used and their positions on the helmet.

6:00

2aPPa21. How many psycho-acoustic attributes are needed? Torben Holm Pedersen (Delta Acoustics & SenseLab, Venlighedsvej 4, 2970 Hørsholm, Denmark, thp@delta.dk), Nick Zacharov (Delta Acoustics & SenseLab, Venlighedsvej 4, 2970 Hørsholm, Denmark, nvz@delta.dk)

Sounds may be characterized by objective perceptive attributes (for which there may exist physical metrics) or by subjective (affective or connotative) attributes. This paper will deal with the perceptive attributes. Within product sound quality the metrics for classical the psycho-acoustic attributes (loudness, sharpness, roughness and fluctuation strength) -maybe supplemented with tone and impulse prominence- are often used as the only attributes to characterize the sounds. But are these 4-6 attributes or dimensions sufficient to characterize a sound? Within room acoustics and reproduced sound many other attributes are used and in the language around 100 direct sound describing words may be found. This paper will give an overview over attributes used within different acoustic areas. The latter part of the paper will discuss the role of sensory evaluation methods as a means to systematically developing attributes for the objective qualification and quantification of sound characteristics.

6:20

2aPPa22. Noise perception of wall-hung gas boilers. Patrick Chevret (01dB-Metravib, 200 Chemin des Ormeaux, 69578 Limonest Cedex, France, patrick.chevret@01db-metravib.com), Anne Coulon (CETIAT, Domaine Scientifique de la Doua, 25 avenue des Arts, BP 2042, F-69603 Villeurbanne cedex, France, anne.coulon@cetiat.fr), François Bessac (CETIAT, Domaine Scientifique de la Doua, 25 avenue des Arts, BP 2042, F-69603 Villeurbanne cedex, France, francois.bessac@cetiat.fr), Etienne Parizet (Laboratoire Vibrations Acoustique, Insa Lyon, 25 bis, av. J. Capelle, 69621 Villeurbanne Cedex, France, etienne.parizet@insa-lyon.fr)

The aim of this study is to assess the quality image for domestic wall-hung gas-fired boilers, based on their noise. Nine boilers were recorded using an acoustic manikin in a hemi-anechoic room, for different operating conditions. Two of these operating conditions (maximum heat input, hot water tapping) were first studied. Five-second sequences were presented (through headphones) to sixty listeners using the mixed assessment method allowing for the comparison between signals. Analyses showed several assessment strategies: according to listeners, the relevant noise parameters could be the loudness, the sharpness or the presence of tonal components. A second experimental phase focused on tonal components by artificially modifying some sounds to offer a relevant perceptive indicator. The results of this second experiment will also be presented.

Invited Papers

6:40

2aPPa23. Methodological aspects in the determination of the auditory filters and critical band at low and mid-frequencies. Carlos A. Jurado (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark, cjo@es.aau.dk), Henrik Møller (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark, hm@acoustics.aau.dk), Christian Sejer Pedersen (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark, cp@acoustics.aau.dk)

In order to evaluate loudness or audibility of complex sounds, knowledge of the auditory filter characteristics is necessary. At low frequencies, where both the threshold of hearing and dynamic range become considerably frequency dependent, care must be taken to account for this both in the psycho-acoustical model and the methodological approach. To account for variation in hearing sensitivity at low frequencies, equal loudness contours have been used to weight the stimuli accordingly. At mid and high frequencies, threshold of hearing curves have been used. These stimuli weightings can be applied before or after the experiment, normally being applied afterwards. Due to the non-linear characteristics of the cochlear amplifier, it is arguable whether post-experimental weighting is a proper approach, or whether at low frequencies there will be any difference between pre or post stimuli weighting. Listening experiments are then to be performed to test possible differences in pre or post filtering the stimuli. The most appropriate approach will then be discussed. Measurements will be done at low and mid frequencies. To obtain accurate auditory filter estimates, individual ELC or threshold curves will be determined. Methods such as the notched-noise method and the classical band-widening approach will be tested with these conditions.

7:00

2aPPa24. Effect of recording/playback technique and experimental method on assessments of noise. Emine Çelik-Christensen (Aalborg University / Rockwool International A/S, Rockwool International, Building Knowledge Center, Hovedgaden 584, Entrance C, 2640 Hedehusene, Denmark, emine.christensen@rockwool.com), Kerstin Persson Waye (Dept. of Environ. Medicine, The Sahlgrenska Acad. of Gothenburg Univ., Box 414, 405 30 Gothenburg, Sweden, kerstin.persson-waye@amm.gu.se), Henrik Møller (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark, hm@acoustics.aau.dk)

The study investigated possible effects of recording/playback technique and experimental method on assessments of annoyance, loudness and unpleasantness. A possible effect of exposure duration was also studied. Sounds were recorded with two different techniques: monophonic and binaural (dummy-head technique). In addition, they were reproduced with three different techniques: monophonic recordings presented through a loudspeaker system, binaural recordings presented through closed (circum-aural) and completely open (free-of-the-ear headphones. The study adopted three psychometric methods for collecting responses from test-subjects. Fifty-four subjects participated, and three types of sounds were used: everyday restaurant sound, road traffic sound and ventilation sound dominated by low frequencies. Each sound was played back at three different levels. The results show that there is no significant main effect of recording and playback technique for any of the three perceptual attributes; however, significant interactions between techniques and sounds were found. Since the effect of recording and playback technique differs depending on sound, this finding is of importance for future design of experiments and interpretation of results. The results also show that long-term annoyance and unpleasantness are poorly predicted by short-duration methods.

Contributed Papers

7:20

2aPPa25. Relation between the overall unpleasantness of a long duration sound and the one of its event6: application to a delivery truck. Emilie Geissner (Laboratoire Vibrations Acoustique - INSA Lyon, 25 bis avenue Jean Capelle, Bâtiment Saint-Exupéry, F-69621 Villeurbanne cedex, France, emilie.geissner@insa-lyon.fr), Etienne Parizet (Laboratoire Vibrations Acoustique, Insa Lyon, 25 bis, av. J. Capelle, 69621 Villeurbanne Cedex, France, etienne.parizet@insa-lyon.fr)

The goal of this study was to investigate the link between the unpleasantness assessment of an unstationary long duration sound composed of several distinct sound events and the corresponding judgments of each of those events. For that purpose, a sound sequence of a delivery truck was evaluated by 16 listeners during a test in laboratory: first, subjects had to continuously quantify the perceived unpleasantness of the sequence by moving a sliding cursor along a five levels graduated scale and then give a global rating by using the same scale. In a second step, listeners had to express their overall judgment of unpleasantness for eight samples of the delivery sequence. As previously shown for loudness by Kuwano and Namba (1985), the global rating of the unpleasantness of long sound could not be estimated by the arithmetic mean of the continuous assessment. It also appeared that the overall judgment corresponds to the arithmetic mean on the local values of unpleasantness of each main sound event. This last result was similar to the conclusions of Hellbrück et al. (2001) for the loudness scaling of traffic noise.

7:40

2aPPa26. Just noticeable differences of loudness and sharpness for earth moving machines. Francesca Pedrielli (CNR-Imamoter, via Canal Bianco 28, 44100 Ferrara, Italy, f.pedrielli@imamoter.cnr.it), Eleonora

Carletti(CNR-Imamoter, via Canal Bianco 28, 44100 Ferrara, Italy, e.carletti@imamoter.cnr.it), Camilla Casazza (CNR-Imamoter, via Canal Bianco 28, 44100 Ferrara, Italy, c.casazza@imamoter.cnr.it)

This paper describes some results of a research concerning the improvement of the noise climate at the operator station of construction machines during real working conditions. Binaural noise signals were previously recorded and then used in subjective listening tests aimed at identifying the set of acoustic and psychoacoustic parameters which affect the auditory perception of these signals with respect to the annoyance sensation. Results showed that loudness and sharpness are the parameters best correlated to the annoyance. In order to verify the efficacy of some noise control solutions in improving the operator comfort conditions, the next necessary step is to build a specific metrics able to show the minimum differences in these parameters which are subjectively perceived. This paper describes the results of specific listening tests carried out in order to evaluate the differential thresholds of loudness and sharpness by the Method of Limits. The test was repeated at two different signal presentation levels. The loudness value of each original stimulus was varied by interval steps of 0,3 sone, while the Sharpness by interval steps of 0,02 acum. At this moment in time, the data analysis is still in progress and the ultimate results will be presented in the manuscript.

Session 2aPPb

Psychological and Physiological Acoustics: Auditory Perception and Signal Processing by Prostheses I

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Invited Papers

8:00

2aPPb1. Advantages and disadvantages of fast and slow compression in hearing aids. Brian Moore (University of Cambridge, Department of Experimental Psychology, Downing Street, CB2 3EB Cambridge, UK, bcjm@cam.ac.uk)

Compression is used in hearing aids to compensate for the effects of loudness recruitment. However, there is no consensus about the ″best″ compression speed. The theoretical advantages and disadvantages of slow and fast compression will be discussed. Studies comparing the relative merits of slow and fast compression have led to a great variety of outcomes. It is argued, following the work of Gatehouse and colleagues, that this is partly the result of a failure to consider individual differences and the auditory ecology of each individual. It is argued that listening in the dips of a fluctuating background sound, such as a competing talker, depends on the ability to process the temporal fine structure (TFS) of sounds, as represented in patterns of phase locking in auditory neurons. For people with a good ability to process TFS, fast compression can amplify sounds in the dips, increasing the effectiveness of dip listening. However, for people with a poor ability to process TFS, envelope cues may be critical for speech intelligibility and fast compression may disrupt such cues. It is proposed that a test of the ability to process TFS might be useful for selecting compression speed for an individual.

8:20

2aPPb2. Dynamic compression in hearing aids based on an auditory model. Volker Hohmann (Carl von Ossietzky Universität Oldenburg, Ammerländer Heerstraße 114-118, 26111 Oldenburg, Germany, volker.hohmann@uni-oldenburg.de)

A multichannel dynamic compression algorithm is proposed that uses a novel nonlinear auditory filterbank which aims at effectively describing the basilar membrane (BM) response to arbitrary signals. It is based on a linear Gammatone filterbank, a subsequent instantaneous compression stage and a frequency-synthesis stage. In order to model the linear response to off-frequency tones and suppression effects from signals below the characteristic frequency (low-side suppression) the compression characteristics is controlled in each filter band by the deviation of the current sub-band instantaneous frequency from the band's center frequency. If the deviation is small, on-frequency components are prominent, and full gain and compression is applied. If the deviation is large, off-frequency components are prominent, and gain and compression is reduced. Simulations of responses to sinusoids as a function of frequency and level at a fixed BM place (i.e., within a fixed filter band) and for all filter bands (i.e., BM excitation patterns) show good correspondence with psychoacoustical excitation pattern models. Responses to two-tone stimuli quantitatively simulate psychoacoustical two-tone suppression. Likewise, nonlinear growth of simultaneous masking is quantitatively modeled. First results show that speech reception in modulated noise may be improved by the system for a subgroup of hearing-impaired subjects.

8:40

2aPPb3. Speech perception in fluctuating noise with signals compensated for hearing loss. Joost M. Festen (VU University Medical Center, de Boelelaan 1117, 1081 HV Amsterdam, Netherlands, jm.festen@vumc.nl)

For speech reception in noise normal-hearing listeners gain from masker modulations up to about 12 dB depending on rate, duty cycle, and depth of the modulations. Listeners with sensorineural hearing loss need a better signal-to-noise ratio to improve signal quality as a compensation for their auditory deficits. Generally, a larger compensation is needed for fluctuating interferences leading to

reduction or even absence of release from masking for modulated maskers. With the Speech Intelligibility Index adapted for modulated maskers, SIImod [Rhebergen and Versfeld, J. Acoust. Soc. Am. 117, 2181-92 (2005)] it is shown that these elevated thresholds are needed to compensate not only reduced hearing sensitivity but also impaired auditory and non-auditory processing. After frequency-dependent compensation for hearing loss as offered by a hearing aid, more of the speech is presented at impaired frequency regions. As a consequence the effect of a hearing aid on speech intelligibility will be less than predicted by SIImod. Speech reception thresholds in noise are affected by peripheral spectro-temporal processing (bottom-up) and by cognitive processing capabilities, like working memory (top-down). Effects of cognitive processing on speech perception are found especially in fluctuating noise as this masker presents a more complex and demanding environment than steady noise.

9:00

2aPPb4. Signal processing algorithms for speech in fluctuating noise. Peggy Nelson (University of Minnesota, 164 Pillsbury Drive SE, Minneapolis, MN 55455, USA, peggynelson@umn.edu), Janet Rutledge (University of Maryland Baltimore County, 1000 Hilltop Circle, Baltimore, MD 21250, USA, jrutledge@umbc.edu), Juan Carlos Tejero-Calado (University of Malaga, Boulevar Louis Pasteur 35, Campus de Teatinos - Complejo Tecnologico, 29071 Malaga, Spain, jctejero@uma.es)

Fluctuating background noise is a significant problem for listeners with sensorineural hearing loss (SNHL). Data indicate that fluctuating noise significantly affects both speech understanding and satisfaction with hearing aids. Listeners with SNHL do not take advantage of momentary dips in the noise and thus to not experience release from masking in fluctuating noise, as normal-hearing listeners do. Our results have shown that listeners with SNHL obtain about half of the masking release of their normal-hearing counterparts, and that this masking release is generally related to the audibility of the speech in the noise dips. Current slow-acting amplitude compression is based on the level of the background noise and does not improve the audibility of speech in the dips of fluctuating noise. Fast-acting multiband compression can improve audibility but may produce unwanted artifacts. Fast-acting algorithms based on spectral peaks shows potential for maximizing the audibility of speech in fluctuating noise to improve speech intelligibility, with limited artifact. Portions of this work are supported by the University of Minnesota and Starkey Laboratories.

9:20

2aPPb5. Signal processing in hearing aids: results of the HEARCOM project. Jan Wouters (ExpORL, Dept. Neurosciences, K.U. Leuven, Herestraat 49 bus 721, O. & N2, B-3000 Leuven, Belgium, jan.wouters@med.kuleuven.be), Heleen Luts (ExpORL, Dept. Neurosciences, K.U. Leuven, Herestraat 49 bus 721, O. & N2, B-3000 Leuven, Belgium, heleen.luts@med.kuleuven.be), Koen Eneman (ExpORL, Dept. Neurosciences, K.U. Leuven, Herestraat 49 bus 721, O. & N2, B-3000 Leuven, Belgium, koen.eneman@med.kuleuven.be), Ann Spriet (ESAT/SISTA, K.U. Leuven, Kasteelpark Arenberg 10, B-3001 Leuven, Belgium, ann.spriet@esat.kuleuven.be), Marc Moonen (ESAT/SISTA, K.U. Leuven, Kasteelpark Arenberg 10, B-3001 Leuven, Belgium, marc.moonen@esat.kuleuven.be), Michael Büchler (University Hospital, 8091 Zürich, Switzerland, michael.buechler@usz.ch), Norbert Dillier (University Hospital, 8091 Zürich, Switzerland, norbert.dillier@usz.ch), Wouter A. Dreschler (AMC, Clinical and Experimental Audiology, 1105 Amsterdam, Netherlands, w.a.dreschler@amc.uva.nl), Matthias Froehlich (Siemens Medical Solutions SAT, 91058 Erlangen, Germany, Matthias.froehlich@siemens.com), Giso Grimm (Carl von Ossietzky Universität Oldenburg, Ammerländer Heerstraße 114-118, 26111 Oldenburg, Germany, giso.grimm@vegri.net), Volker Hohmann (Carl von Ossietzky Universität Oldenburg, Ammerländer Heerstraße 114-118, 26111 Oldenburg, Germany, volker.hohmann@uni-oldenburg.de), Rolph Houben (AMC, KNO-Audiologie, 1105 Amsterdam, Netherlands, a.c.houben@amc.uva.nl), Arne Leijon (KTH, Royal Institute of Technology, 10044 Stockholm, Sweden, arne.leijon@ee.kth.se), Anthony Lombard (University of Erlangen, 91058 Erlangen, Germany, lombard@nt.e-technik.uni-erlangen.de), Dirk Mauler (Ruhr University of Bochum, 44780 Bochum, Germany, dirk.mauler@ruhr-uni-bochum.de), Henning Puder (Siemens Medical Solutions SAT, 91058 Erlangen, Germany, henning.puder@siemens.com), Michael Schulte (Hoerzentrum Oldenburg, 26129 Oldenburg, Germany, M.Schulte@hoerzentrum-oldenburg.de), M Vormann (Hoerzentrum Oldenburg, Hoerzentrum Oldenburg, 26129 Oldenburg, Germany, matthias.vormann@web.de)

Digital hearing aids of today allow the application of advanced signal processing strategies. In recent years a number of promising signal processing approaches have been designed and developed. However, most of these different evolutions have been evaluated only in a limited way. Within the framework of the HEARCOM EU-research project a number of signal enhancement techniques have been further developed and evaluated based on a representative set of real-life recordings and physical performance measures. Different auditory profiles, representing common categories of hearing aid users, have been taken into account. A selection of 5 of these signal enhancement techniques (single-channel noise suppression, blind source separation, dereverberation, multi-microphone adaptive processing, feedback reduction) has been implemented on a single common hard- and software test platform, the Master Hearing Aid (MHA). These signal processing strategies have been evaluated perceptually based on speech reception thresholds, listening effort and preference rating, at 5 different test-sites for a number of speech-and-noise listening scenarios. Fifty normal hearing subjects and 100 hearing aid users according to 2 auditory profiles, took part in this study.

9:40

2aPPb6. Model-based objective assessment of noise reduction systems for hearing aids. Birger Kollmeier (Universität Oldenburg, Medizinische Physik, Carl-von-Ossietzky Str. 9-11, 26111 Oldenburg, Germany, birger.kollmeier@uni-oldenburg.de), Rainer Huber (Kompetenzzentrum HörTech, Marie-Curie-Str. 2, 26129 Oldenburg, Germany, Rainer.Huber@HoerTech.de), Thomas Rohdenburg (Universität Oldenburg, Medizinische Physik, Carl-von-Ossietzky Str. 9-11, 26111 Oldenburg, Germany, thomas.rohdenburg@uni-oldenburg.de), Rainer Beutelmann (Universität Oldenburg, Medizinische Physik, Carl-von-Ossietzky Str. 9-11, 26111 Oldenburg, Germany, rainer.beutelmann@uni-oldenburg.de), Volker Hohmann (Carl von Ossietzky Universität Oldenburg, Ammerländer Heerstraße 114-118, 26111 Oldenburg, Germany, volker.hohmann@uni-oldenburg.de)

Since the ultimative goal of hearing-aid development is the (subjective) judgment of the individual hearing-impaired listener, time-consuming tests with the end user are indispensable. However, time- and effort-saving objective methods to assess the potential benefit of different versions and parameter sets of hearing aid algorithms are gaining importance. This contribution reviews perception-model-based approaches to predict the hearing-impaired judgement and speech reception performance achieved with various noise reduction

schemes. The perceptual similarity measure PSM evaluates the similarity between a tested condition and an ″ideal″ reference condition not on the physical level, but rather on the perceptual level at the output of a perception model for the individual hearing-impaired listener. The binaural extention of the SII approach uses a binaural preprocessing stage followed by a speech intelligibility index (SII)-based prediction scheme capable of predicting the relative benefit of binaural signal presentation and signal enhancement in complex spatial signal and noise source configurations. Both model-based schemes can be combined to assess the effect of noise reduction algorithms (such as adaptive beamformers) and to optimize their respective performance for different acoustical situations.

Contributed Papers

10:00

2aPPb7. Quantifying and modeling the acoustic effects of compression on speech in noise. Koenraad S. Rhebergen (AMC - Dept. of Clinical and Experimental Audiology, AMC, Clinical and Experimental Audiology, 1105 Amsterdam, Netherlands, k.s.rhebergen@amc.uva.nl), Niek J. Versfeld (AMC - Dept. of Clinical and Experimental Audiology, Meibergdreef 9, 1105AZ Amsterdam, Netherlands, n.j.versveld@amc.uva.nl), Wouter A. Dreschler (AMC, Clinical and Experimental Audiology, 1105 Amsterdam, Netherlands, w.a.dreschler@amc.uva.nl)

In this presentation a method is proposed that is able to separate a speech signal out of a noise signal after processing of the signal through wide-dynamic-range compression (WDRC). This technique reconstructs the speech signal and noise signal sample by sample separately using the gain factor of the WDRC, and can be used to quantify the acoustic effects of WDRC in noise. It will be shown that this technique is more accurate than a frequently used inversion technique, because the method is not affected by phase shifts that introduce distortion products in the reconstructed speech signal. As a result, the acoustic effects of WDRC can be measured more accurately. In addition, this reconstruction method allows modeling the speech intelligibility after non-linear signal processing in the Speech Intelligibility Index. With the aid of Speech Reception Threshold data it will be shown that this approach can give a good account for most existing data.

10:20

2aPPb8. Acoustical frequency discrimination and pitch matching in bimodal and hybrid hearing. Uwe Baumann (Univ. of. Frankfurt - ZHNO - Audiologische Akustik, Theodor-Stern-Kai 7, Haus 8D, 60590 Frankfurt a.M., Germany, uwe.baumann@kgu.de), Tobias Rader (Univ. of. Frankfurt - ZHNO - Audiologische Akustik, Theodor-Stern-Kai 7, Haus 8D, 60590 Frankfurt a.M., Germany, tobias.rader@kgu.de), Silke Helbig (Univ. of. Frankfurt - ZHNO - Audiologische Akustik, Theodor-Stern-Kai 7, Haus 8D, 60590 Frankfurt a.M., Germany, silke.helbig@kgu.de), Wolfgang Gstöttner (Univ. of. Frankfurt - ZHNO - Audiologische Akustik, Theodor-Stern-Kai 7, Haus 8D, 60590 Frankfurt a.M., Germany, a.jan@em.uni-frankfurt.de)

Frequency discrimination and pitch matching of implantees using combined electric and acoustic stimulation in either the same ear (EAS) or the opposite ear (bimodal condition) was assessed by means of adaptive procedures. EAS patients received either the MED-EL standard electrode or the recently introduced FLEX design with reduced diameter. Acoustic JNDF in EAS patients ranged from close to normal to grossly abnormal compared to a group of matched SNHL listeners. The median JNDF was 7.1% in the SNHL and 7.5% in the EAS group. There was no statistically significant difference in terms of JNDF between both groups of listeners. Frequency mapping was studied by means of an adjustment method where subjects were instructed to control the pitch of an acoustically presented sinusoid in reference to electrical stimulation. The findings demonstrate that the insertion of an intra-cochlear electrode does not significantly hamper the average frequency discrimination ability in EAS patients.

Invited Paper

10:40

2aPPb9. Combining hearing aids and cochlear implants to solve the cocktail party problem. Fan-Gang Zeng (University of California Irvine, 364 Med Surge II, Irvine, CA 92697, USA, fzeng@uci.edu)

The cocktail party problem refers to the difficulty in speech recognition in noise that a hearing-impaired listener must face in daily life. Combining a hearing aid with a cochlear implant can provide complementary information that may have a great potential to solve this problem. On one hand, a hearing aid may provide low-frequency temporal fine structure cues that are not conveyed by a cochlear implant. On the other hand, a cochlear implant can provide high-frequency temporal envelope cues that are not effectively delivered by a hearing aid. This talk will provide psychophysical and speech recognition evidence for combining hearing aids and cochlear implants to solve the cocktail party problem. One interesting finding along this line of research is that in many important functional tasks, the hearing aid and cochlear implant combination provides a more effective solution than bilateral cochlear implants. Another interesting finding is that the fundamental frequency cue alone can significantly improve speech perception in noise, especially when the noise is a competing voice. The latter finding suggests that combining a tactile aid and a cochlear implant can potentially achieve the same benefit as combining a hearing aid and a cochlear implant in patients with no residual acoustic hearing.

Session 2aPPc

Psychological and Physiological Acoustics: Binaural Perception by Hearing-Aid Wearers

Sridhar Kalluri, Cochair
Starkey Hearing Research Center, 2150 Shattuck Ave, Suite 408, Berkeley, CA 94704, USA

Michael A. Akeroyd, Cochair
MRC Institute of Hearing Research, Glasgow Royal Infirmary, 16 Alexandra Parade, Glasgow, G31 2ER, UK

Invited Paper

11:20

2aPPc1. Binaural function and its benefits for wearers of hearing aids. Sridhar Kalluri (Starkey Hearing Research Center, 2150 Shattuck Ave, Suite 408, Berkeley, CA 94704, USA, sridhar_kalluri@starkey.com), Brent Edwards (Starkey Hearing Research Center, 2150 Shattuck Ave, Suite 408, Berkeley, CA 94704, USA, brent_edwards@starkey.com)

Hearing-aid design has been predicated traditionally on improving the reception of speech in background noise, a goal that has, at least in part, been successfully attained. Despite the advances, hearing-aid wearers continue to have difficulty in complex acoustic environments and in auditory tasks where normal-hearing listeners benefit greatly from binaural hearing. In order to improve hearing-aid wearers' performance in such challenging conditions, recent research has begun examining the effects of hearing aids on binaural perception. Interest in the topic is heightened by the rapidly approaching prospect of bilateral hearing aids that communicate wirelessly and thereby allow implementation of more complex signal-processing algorithms than currently possible. This talk will review binaural function and its benefits, with particular emphasis on the aspects that hearing aids can affect. The talk will also speculate on which hearing-aid algorithms can affect binaural perception.

Contributed Paper

11:40

2aPPc2. Auditory localization with linear and compression hearing aids. Helen J. Simon (Smith-Kettlewell Eye Research Institute, 2318 Fillmore St., San Francisco, CA, CA 94115, USA, helen@ski.org), E. William Yund (VAMC, 150 Muir Road, Martinez, CA 94553, USA, yund@ebire.org), Harry Levitt (Advanced Hearing Concepts, PO Box 610, Bodega Bay, CA 94923, USA, harrylevitt@earthlink.net)

The question of how well hearing-impaired individuals can localize sound (with or without amplification) is still not fully resolved. This study was designed to compare sound localization with two types of hearing-aid (HA) processing, wide dynamic range multichannel compression (WDRMCC) and linear amplification (LA) with compression limiting, during the first 32 weeks of HA use. HAs from two different manufacturers were included to compare different digital signal processing implementations, (1) fast Fourier transform (FFT), necessitating a 10 ms delay, and (2) non-FFT signal processing with a shorter time delay (~1 ms). We found an initial degradation of sound localization, relative to original unaided performance, for both WDRMCC and LA in both FTT and non-FTT platforms. We found no difference between WDRMCC and LA processing. However, sound localization with non-FFT platform improved consistently throughout 32 weeks of HA use and was better than the original unaided measurements at 16 and 32 weeks. In contrast, localization with the FFT platform showed no consistent change throughout the 32-week test period and remained inferior to original unaided performance. The continuing localization problems present for the FFT, but not the non-FFT, implementations of LA and WDRMCC may be due to its 10-ms processing delay.

Invited Papers

12:00

2aPPc3. Effects of noise type and location on binaural benefit in asymmetric directional fittings. Benjamin Hornsby (Vanderbilt University, Room 8310 Medical Center East, South Tower, 1215 21st Ave. South, Nashville, TN 37232-8242, USA, ben.hornsby@vanderbilt.edu)

The benefits of bilateral directional processing for improving speech understanding in noise are well documented. However, these fittings are not universally accepted by hearing aid wearers. Research suggests that an asymmetric fitting (omnidirectional in one ear-/directional in the other) may provide benefit in noise comparable to symmetric directional fittings (directional in both ears). This study evaluated factors that may affect the relative benefit provided by an asymmetric directional fitting. Specific factors evaluated included noise configuration, reverberation and noise type. Twenty individuals with mild-moderate SNHL participated. Aided speech understanding in cafeteria babble was assessed in bilateral omnidirectional and directional modes and in an asymmetric mode in four (4) different noise configurations. Measures were made in both an anechoic and reverberant (RT ~620 ms). In a second experiment the effects of noise type were evaluated by comparing performance in symmetric and asymmetric modes in both steady state noise and cafeteria babble. Results suggest that noise configuration has a significant effect on the relative benefit provided by asymmetric fittings. No significant differences due to noise type were observed. Individual differences in bilateral directional benefit also appear to have a significant effect on the reduction in benefit resulting from an asymmetric fitting.

12:20

2aPPc4. Speech-in-noise enhancement and sound localization with improved binaural hearing instruments. Jan Wouters (ExpORL, Dept. Neurosciences, K.U. Leuven, Herestraat 49 bus 721, O. & N2, B-3000 Leuven, Belgium, jan.wouters@med.kuleuven.be), Simon Doclo (ESAT/SISTA, K.U. Leuven, Kasteelpark Arenberg 10, B-3001 Leuven, Belgium, simon.doclo@esat.kuleuven.be), Marc Moonen (ESAT/SISTA, K.U. Leuven, Kasteelpark Arenberg 10, B-3001 Leuven, Belgium, marc.moonen@esat.kuleuven.be), Tim Van Den Bogaert (ExpORL, Dept. Neurosciences, K.U. Leuven, Herestraat 49 bus 721, O. & N2, B-3000 Leuven, Belgium, tim.vandenbogaert@med.kuleuven.be)

Multi-microphone noise reduction schemes have become standard in commercial hearing aids and cochlear implants. Recent studies with bilateral hearing aids have shown that common adaptive directional microphone systems tend to distort localization cues, leading to inappropriate and reduced spatial awareness for bilateral hearing aid users. Here we show that binaural multi-microphone signal processing based on multi-channel Wiener filter (MWF) are capable of combining noise reduction with the preservation of directional hearing. Physical simulations and perceptual results from 10 listeners have been studied for different noise source scenarios, in different reverberant conditions, and for a number of signal processing schemes using up to 4 microphone inputs (2 each side). An overview is given of the localization performance and the speech reception benefits in these different listening conditions for the different noise reduction strategies. An adaptive directional microphone system (ADM) is used as a reference system. Signal processing based on MWF does, unlike ADM, provides a combination of noise reduction and preservation of spatial awareness. Moreover, in some conditions it even offers an improved spatial release from masking. The MWF outperforms the ADM in terms of localization and noise reduction if signals are not arriving from the most forward field of view.

12:40-2:00 Lunch Break

Invited Papers

2:00

2aPPc5. Interaction between stimulus and compression type in precedence situations with hearing aids. Bernhard U. Seeber (MRC Institute of Hearing Research, Science Rd / University Park, NG7 2RD Nottingham, UK, seeber@ihr.mrc.ac.uk), Cheryl Eiler (Starkey Hearing Research Center, 2150 Shattuck Ave, Suite 408, Berkeley, CA 94704, USA, cheryl_eiler@starkey.com), Sridhar Kalluri (Starkey Hearing Research Center, 2150 Shattuck Ave, Suite 408, Berkeley, CA 94704, USA, sridhar_kalluri@starkey.com), Ervin R. Hafter (University of California, Department of Psychology, 3210 Tolman Hall, Berkeley, CA, CA 94720-1650, USA, hafter@berkeley.edu), Brent Edwards (Starkey Hearing Research Center, 2150 Shattuck Ave, Suite 408, Berkeley, CA 94704, USA, brent_edwards@starkey.com)

Multiband compression in hearing aids has the potential to interfere with binaural perception by altering binaural cues. We compared binaural precedence in patients fitted with compression hearing aids or linear hearing aids and tested one month later with a variety of stimuli. In an open field environment simulated in an anechoic chamber, subjects localized the direction of a leading sound accompanied by a lagging copy played from a different direction. Precedence, defined by a strong influence of the first sound, was found in 6/7 of the subjects when the stimulus was a sentence, but in only 4/7 when it was a high-pass or a wideband burst of noise. The fact that increasing the noise-bandwidth to include low frequencies did not produce precedence in the two patients who had shown it with a sentence may indicate a need for them to accumulate information through the successive pseudo onsets in the speech-envelope. The choice of amplification, compressive or linear, did not significantly affect precedence for any sound, except for one subject who showed weak influence of compression with the high-pass noise. We conclude that although compression might alter interaural level cues, for most, especially with long sounds, localization dominance appears unimpaired.

2:20

2aPPc6. Effect of hearing aids on distance perception. Michael A. Akeroyd (MRC Institute of Hearing Research, Glasgow Royal Infirmary, 16 Alexandra Parade, G31 2ER Glasgow, UK, maa@ihr.gla.ac.uk)

The two primary auditory cues to distance in rooms are the overall level of the sounds received by the listener and the ratio of the level of the direct sound to the reverberant sounds. In previous work we have demonstrated that hearing-impaired listeners showed no overall deficits in the ability to use the overall-level cue, but they did have deficits in the ability to use the direct-to-reverberant cue [M. Akeroyd, S. Gatehouse, and J. Blaschke, J. Acoust. Soc. Am., 121, 1077-1089 (2007)]. These deficits would be expected to contribute to the auditory disability suffered by the listeners. But both of these auditory cues are level cues, as they require some measurement of intensity, and so they would be expected to be affected adversely by the amplitude compression found in most modern hearing aids. Initial results from an experiment measuring the just-noticeable difference (JND) for changes in distance (at 2 m and 5 m) with experienced hearing-aid users suggest, however, that their JNDs are no different to those of unaided but impaired listeners. If confirmed by the final results, this surprising result may be interpretable if listeners have acclimatized to the effects of their aids on level.

2:40

2aPPc7. Binaural hearing abilities of bilaterally fitted hearing aid users assessed using objective and subjective outcome measures. Thomas Behrens (Eriksholm Research Centre, Oticon A/S, 243 Kongevejen, 3070 Snekkersten, Denmark, tbs@oticon.dk), Tobias Neher (Eriksholm Research Centre, Oticon A/S, 243 Kongevejen, 3070 Snekkersten, Denmark, ton@oticon.dk)

Aided spatial hearing in the hearing impaired remains a rather sparsely explored topic. Therefore we do not know enough about the ability of the hearing impaired to exploit auditory cues for spatial hearing. In an attempt to advance our knowledge the following

experiment was set up. A group of 21 experienced hearing aid users took part in a test using modern completely in the canal hearing aids in the field for at least 7 weeks. After acclimatization to the devices they were tested on a number of outcome measures. These included spatial unmasking, an interview administered selection of questions from the Speech Spatial and Qualities of hearing scale, and a baseline speech in noise measure. Spatial unmasking was assessed using three concurrent female talkers, with the target talker always presented directly ahead and the maskers presented either at +/- 50 degrees or both at 180 degrees. This presentation will provide results from the study described above along with possible relations to auditory and non-auditory predictors of spatial hearing performance.

Contributed Paper

3:00

2aPPc8. The effect of binaural processing techniques on speech quality ratings of assistive listening devices in different room acoustics conditions. Johan Odelius (Luleå University of Technology, Dept Human Work Sciences, Div of Sound and Vibration, SE-97187 Luleå, Sweden, johan.odelius@ltu.se), Örjan Johansson (Luleå University of Technology, Dept Human Work Sciences, Div of Sound and Vibration, SE-97187 Luleå, Sweden, orjan.johansson@ltu.se)

External microphone systems, referred to as assistive listening devices (ALD), are used in classrooms for hearing impaired students. The objective is to investigate the effect of binaural processing techniques in different room acoustic conditions. A listening experiment was conducted with 10 normal hearing adults. Response variables were judgements of clarity, pleasantness, listening effort and overall speech quality. Design variables were binaural processing, room acoustics and ALD bandwidth. Stimuli were generated using the room acoustic modelling software CATT Acoustic. Three speech sources, two male voices and one female voice, were placed at a table in the centre of a room and one Brown noise source was placed in one corner of the room. Microphones were placed 0.5 m in front of each speech source. Target source was a random choice of one of the two male voices. The binaural processing was utilized by a simple HRTF filtering. Depending on the angle to the source from a fictitious listening position at the table, corresponding interaural time difference (ITD) and the interaural level difference (ILD) was applied to the signal. Stimuli were presented by loudspeakers using cross-talk cancellation. The hypothesis is that binaural processing will give a significant improvement in speech quality.

TUESDAY MORNING, 1 JULY 2008 ROOM 242B, 8:00 TO 10:40 A.M.

Session 2aSAa

Structural Acoustics and Vibration and EURONOISE: Vibration and Radiation from Complex Structural Systems II

David Feit, Cochair
Applied Physical Sciences Corp., Ste. 300, 2 State St., New London, CT 06320, USA

Jean-Louis Guyader, Cochair
INSA de Lyon - LVA, Bâtiment St. Exupéry, 25 bis avenue Jean Capelle, Villeurbanne Cedex, F-69621, France

Contributed Papers

8:00

2aSAa1. On the comparison of symmetric and unsymmetric formulations for experimental vibro-acoustic modal analysis. Morvan Ouisse (FEMTO-ST UMR CNRS, 23 chemin de l'Epitaphe, 25000 Besançon, France, morvan.ouisse@univ-fcomte.fr), Emmanuel Foltete (FEMTO-ST Applied Mechanics, 24 chemin de l'épitaphe, 25000 Besançon, France, emmanuel.foltete@univ-fcomte.fr)

The classical u-p formulation for vibro-acoustic problems is very convenient for experimental vibro-acoustic modal analysis since the physical variables are directly those which are measured by operators. In this particular context, the objective is to identify from experimental measurements a reduced model which has the same behaviour as the measured one. The complex mode shapes which are identified using this technique must satisfy a properness condition. When they do not verify it, they should be modified to be able to represent the behaviour of a physical system. Some techniques have been proposed in order to develop a strategy to obtain the modified eigenshapes, but this is a quite difficult point because of the unsymmetric topology of the equations. In this paper, a symmetric formulation is used in order to be able to directly apply the classical methodology which has been developed for structural modal analysis to obtain the physical reduced system. The methodology is described and compared with the u-p formulation, in terms of efficiency and precision, in particular when some absorbing devices are considered. All results are first presented on an ideal numerical test-case, and applications on experimental data are finally shown.

8:20

2aSAa2. Numerical modeling of Panphonics's G1 flat loudspeaker. Krisztián Gulyás (Budapest University of Technology and Economics, BME Dept. of Telecommunications, Magyar tudósok körútja 2, H-1117 Budapest, Hungary, gulyas@hit.bme.hu), Fülöp Augusztinovicz (Budapest University of Technology and Economics, BME Dept. of Telecommunications, Magyar tudósok körútja 2, H-1117 Budapest, Hungary, fulop@hit.bme.hu)

The basic idea behind AABC (Active Acoustic Barrier Control) is to reduce the sound radiation of a structure acoustically without influencing the vibration behavior of the structure. The large surface acoustic polymer material actuator/sensor systems are primarily meant to form the actuator/sensor system for the AABC concept. The main component of this package is a special active device: the rEMA - revised Elastic Mass Actuator based on the Panphonics' G1 panel loudspeaker element. This paper deals with the modeling and the low-frequency performance issues of the G1 flat loudspeaker. The aim was to create an accurate mathematical model to understand the operational principles of the loudspeaker, and then this model was used to optimize its low-frequency performance. Due to the special

structure and the optimization tasks, a new numerical method was developed to model special multi-layer coupled vibro-acoustics systems based on the Finite Difference and Boundary Element Method. The investigation focuses on the mechanical behavior of the panel and describes the sound radiation properties also. The performed work was a part of the research of the project InMAR (Intelligent Materials for Active Noise Reduction) which was funded by the European Union.

8:40

2aSAa3. Metrology and prediction for integrating a subsystem on a vehicle: Application to a fan system attached to the front end of a car. Marie-Hélène Moulet (CEVAA, Technopole du Madrillet, 2 Rue Joseph Fourier, 76800 Saint Etienne du Rouvray, France, mh.moulet@cevaa.com), Saul Mapagha (CEVAA, Technopole du Madrillet, 2 Rue Joseph Fourier, 76800 Saint Etienne du Rouvray, France, s.mapagha@cevaa.com), Vincent Martin (Institut Jean Le Rond d'Alembert, UMR CNRS 7190, UPMC, 2 Place de la Gare de Ceinture, 78210 Saint-Cyr l'Ecole, France, vmartin@ccr.jussieu.fr)

The car industry must satisfy physical acoustic and vibratory objectives in order to comply with safety and comfort norms. Among others, efforts entering the chassis should be limited. When a host structure is excited by a vibratory system, called subsystem, it also vibrates and may radiate an acoustic field. Usually, the subsystem is first tested on a bench and the question is then to deduce the efforts entering the chassis from those entering the test bench. Globalising notions of impedance already made evident via deformable structure configurations are used presently through measurements in a technological configuration, the complexity of which does not allow modelling. The case of a fan system attached to the front end of a car is under study. Some metrological conclusions, as well as more dedicated ones concerning the fan system itself, will be given. Finally, on the calculation and prediction sides, expected properties that are not satisfied (such as symmetry in measured matrices) are seen to be of no great consequence in the present case.

9:00

2aSAa4. Numerical study of the transition to chaos in nonlinear forced vibrations of plates. Cédric Camier (ENSTA, Chemin de la Hunière, 91761 Palaiseau cedex, France, cedric.camier@ensta.fr), Cyril Touze (ENSTA, Chemin de la Hunière, 91761 Palaiseau cedex, France, cyril.touze@ensta.fr), Olivier Thomas (CNAM, 2, rue Conté, 75003 Paris, France, olivier.thomas@cnam.fr), Stefan Bilbao (University of Edinburgh, Room 7306B, JCMB, King's Bldgs., Mayfield Rd., EH9 3JZ Edinburgh, UK, sbilbao@staffmail.ed.ac.uk)

Geometrically nonlinear vibrations of free edge circular plates subjected to a harmonic excitation are discussed. Particularly, transition from periodic to chaotic motion is observed when increasing the amplitude of the forcing. The present work is devoted to reproduce numerically these highly nonlinear behaviours. The temporal integration of such dynamics, including instabilities and chaotic regimes, is not straight forward because a stiff problem with a very large number of dofs is at hand. Consequently, numerical instabilities are observed when typical Runge-Kutta schemes are applied. To settle the matter, two methods have been tested and compared. They both rely on a modal approach applied to the von Kármán's model for large amplitude vibrations of plate. For the first one, the energies of the plates are expressed at the continuous level. The Hamiltonian of the system is then derived and discretized using the eigenmodes. The Hamiltonian formulation ensures the conservation of energy. An implicit time discrete scheme is then chosen to approximate the equations of motion. For the second one, the Gear's BDF method, implemented in the IMSL Fortran library, is used to integrate the nonlinear oscillator equations.

9:20

2aSAa5. Numerical investigations into the squeal propensity of a railway disc brake. Olivier Chiello (INRETS, Transport and Environment Lab., 25 avenue François Mitterrand, 69675 Bron cedex, France, olivier.chiello@inrets.fr), Xavier Lorang (SNCF, Innovative & Research Dept., 45 rue de Londres, 75379 Paris, France, xavier.lorang@sncf.fr)

This paper comes within the scope of a research program concerning with the reduction of the squeal noise generated by high power railway disc brakes. It focuses on the numerical results provided by a finite element model of the brake including unilateral contact and Coulomb friction at the disc/pad interfaces. In particular, the dynamic stability of the sliding equilibrium is investigated by performing a complex eigenvalue analysis of the linearized equations verified by the structural displacement fields. Complex eigenvalues and complex modes are used to estimate the squeal propensity of the brake in a given frequency range. The effect of various mechanical and geometrical parameters is studied in order to better understand the mechanism leading to the system instability.

9:40

2aSAa6. Three dimensional orthogonality of the Lamb modes in layered plates of elastic and viscoelastic materials and their implementation to the far field evaluation. Dmitry Zakharov (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, dmitrii.zakharov@gmail.com)

The 3D guided waves in the linearly viscoelastic laminates are considered. On the plate surfaces any of the homogeneous boundary conditions are allowed, e.g., the Lamb waves, waves in clamped plates, etc. are taken into account. The fundamental property of these waves is their generalized orthogonality, which is deduced and discussed. The applications of the orthogonality relations for solving some particular boundary value problems are suggested. A method for the exact calculation of the far field caused by an acoustic source of a finite size is suggested. The only restriction is that the distance required must exceed the longitudinal radius of the source. The obtained results can be used for evaluating the fields radiated by ultrasonic transducers of arbitrary aperture and by other realistic sources.

10:00

2aSAa7. Sound Radiation of a Large Truck Oil Pan: Estimation and Experimental Investigation. Olaf Heintze (German Aerospace Center (DLR), Institute of Composite Structures and Adaptive Systems, Lilienthalplatz 7, 38108 Braunschweig, Germany, olaf.heintze@dlr.de), Volker Wittstock (Physikalisch-Technische Bundesanstalt, Bundesallee 100, 38116 Braunschweig, Germany, volker.wittstock@ptb.de), Carl Fredrik Hartung (Volvo Technology Corporation, Götaverksgatan 10, 405 08 Göteborg, Sweden, carl.fredrik.hartung@volvo.com)

The oil pan of large diesel engine trucks has been identified as a significant contributor to their external noise radiation. This undesired broadband noise is caused by the oil pan's structural vibration and can not be treated effectively by passive measures especially in the low frequency regime up to 500Hz. In order to address this challenge, an active structural acoustic control system consisting of structural sensors and actuators is suitable to alter the oil pans vibrations in a sound reducing manner. A first step is however to classify the broadband sound radiation such that it allows for a proper and efficient sound power estimation resulting from structural measurements. Therefore, an acoustical model was set up based on a geometrical surface scan of a serial production large truck oil pan mounted in a laboratory test stand. This model served for the numerical computation of a set of principle velocity patterns contributing independently to the active sound power, where its hybrid estimation has been performed employing additionally the measured structural response of the oil pan assembly due to a shaker excitation. Finally, the sound power radiation of the test stand has been measured in a reverberation room to validate this hybrid estimation.

10:20

2aSAa8. Sound propagation on a high pressure gas pipe. Rolf Schirmacher (Müller-BBM, Robert-Koch-Straße 11, 82110 Planegg, Germany, Rolf.Schirmacher@MuellerBBM.de), Robert Baars (M+P raadgevende ingenieurs B.V., Visserstraat 50, 1431 GJ Aalsmeer, Netherlands, robertbaars@mp.nl)

At a two stage high pressure gas compressor with intermediate cooler, the cooler radiate a tone at the compressor rotational frequency of approx. 160 Hz. For the design of noise reduction devices, the mechanism of sound propagation in between the compressor an the cooler on a DN 400 steel pipe with 24 mm wall thickness and 160 Bar internal gas pressure was to be determined. By non-invasive vibration measurements on the pipe, the dominance of the fluid borne sound (natural gas) over the structure borne sound was found. Later, pressure measurements in the pipe clearly approved this result. The concepts upon which the measurements and data evaluation were based (propagating modes of coupled system, wave separation, etc) as well as the results are presentend.

TUESDAY MORNING, 1 JULY 2008 ROOM 243, 8:00 A.M. TO 12:20 P.M.

Session 2aSAb

Structural Acoustics and Vibration and EURONOISE: General Topics in Structural Acoustics and Vibration II

Wolfgang Kropp, Cochair
Chalmers University of Technology, Division of Applied Acoustics, Gothenburg, SE-41296, Sweden

Sean Wu, Cochair
Wayne State University, 5050 Anthony Wayne Dr., Detroit, MI 48202, USA

Contributed Papers

8:00

2aSAb1. Wavelet-based data processing for comparative study of noncontact measurement techniques for vibroacoustics. Dan Borza (National Institute of Applied Sciences of Rouen, INSA Rouen, LMR, Ave l'Universite, BP8 76800 Rouen, France, dan.borza @insa-rouen.fr), Ioana Nistea (Institut National des Sciences Appliquées de Rouen, BP8 avenue de l'Université, 76801 Saint-Etienne du Rouvray, France, ioana.nistea@insa-rouen.fr)

In vibration analysis of structures complex measurement information is required in order to perform modal analysis, therefore the choice of the measurement technique to be applied is essential. Non contact measurements are preferred to classical transducer based methods, mainly due to the absence of influence upon the structure under test. The various techniques available today in vibroacoustics produce results which are quite different in terms of spatial and temporal resolution or measured quantities and therefore a choice has to be made of the experimental tool best adapted for different fields of research (acoustics, mechanical structures, dynamics). In this paper, we present a comparative study of several optical, acoustical and numerical techniques for vibration measurement or simulation, namely Digital Speckle Interferometry, Laser Doppler Vibrometry and a FE model. The tests were made for the free and the forced vibrations of a highly dumped, non-metallic plate. In the data processing stage, discrete wavelet decomposition has been applied on the experimental data in order to match up the spatial maps of vibration amplitudes.

8:20

2aSAb2. Analysis of the acoustic signals backscattered by a tube using the time-frequency representations. Mustapha Laaboubi (Ibn Zohr University, FS Agadir, 80000 Agadir, Morocco, laaboubi @gmail.com), Elhoucien Aassif (Ibn Zohr University, FS Agadir, 80000 Agadir, Morocco, aassif@hotmail.com), Rachid Latif (ESSI - ENSA, BP 1136, Ibn Zohr University, 80000 Agadir, Morocco, latif@ensa-agadir.ac.ma), Gerard Maze (LAUE, Université du Havre, Place Robert Schuman, F-76610 Le Havre, France, gerard.maze @univ-lehavre.fr),Dominique Decultot(LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, dominique.decultot@univ-lehavre.fr), Ali Moudden (Ibn Zohr University, FS Agadir, 80000 Agadir, Morocco, ali_moudden@yahoo.fr), Abdelilah Dariouchy (LMTI, univirsté ibn zohr Faculté des Sciences LMTI bp 8106, 80000 Agadir, Morocco, abdelilah _dariouchy@yahoo.fr)

The normal excitation of a tube immersed in water by the acoustic plane wave, circumferential waves are generated inside the shell. These circumferential waves, standing form stationary waves on the circumference of the tube for some frequencies. These stationary waves, constituting resonances of the tube which are perfectly visible on the backscattered spectrum. Moreover, the studies carried out on the diffusion of a plane acoustic wave by target were based primarily on the use of the monodimensional methods (Temporal domain and/or frequencial domain). To exceed the disadvantages of these methods, in this work, we used the time-frequency representations such as the Short-Term Fourier Transform (STFT), Wigner-Ville Distribution (WVD) and Wavelet Transform method. These representations are applied to a theoretical signal backscattered by a tube of aluminium, copper and steel with radii ratio b/a = 0.95 (a is the external radius, and b the internal radius). From the time-frequency images obtained we have visualized the dispersion of circumferential waves (S0, A1, S1,œ) and identified these different waves. This analysis permits to compare between these time-frequency representations. And also we have compared between the cut-off frequencies of circumferential waves obtained from these representations and those computed by the proper modes theory of the vibration.

8:40

2aSAb3. Weak radiator design using dimples. D. Chih-Chun Cheng (National Chung Cheng University, Dept. of Mechanical Engineering, 160, San-Hsing, Ming-Hsiung, 621 Chia-Yi, Taiwan, imeccc@ccu.edu.tw), Wen-Nan Cheng (National Chung Cheng University, Dept. of Mechanical Engineering, 160, San-Hsing, Ming-Hsiung, 621 Chia-Yi, Taiwan, zzs1003@hotmail.com), Cary H. Koopmann (Pennsylvania

State University, Dept. of Mechanical and Nuclear Engineering, 157 Hammond Building, University Park, PA 16802, USA, ghk@engr.psu.edu)

A design method for achieving minimum radiation of sound from a beam by creating cylindrical dimples on its surface is presented. Two strategies of determining the dimple size and its location on the beam are presented. The first is based on the optimization method, in which the dimple size and its location are the design variables. The design variables that minimize the sound power are obtained directly using an optimization subroutine. The second is to synthesize the beam's weak radiator mode using a set of dimples, in which the mode shape of the dimpled beam is a close fit to the beam's weak radiator mode. As a comparison in sound power reduction between these strategies, numerical results for a simply supported beam in a rigid baffle excited by a harmonic force are presented.

9:00

2aSAb4. Estimation for vibration mode of membrane by NAH method. Nabilah Ibrahim (Shibaura Institute of Technology, 3-7-5, Toyosu, Koto-ku, 135-8548 Tokyo, Japan, m107068@sic.shibaura-it.ac.jp), Hideo Shibayama (Shibaura Institute of Technology, 3-7-5, Toyosu, Koto-ku, 135-8548 Tokyo, Japan, sibayama@sic.shibaura-it.ac.jp), Toru Itakura (Shibaura Institute of Technology, 3-7-5, Toyosu, Koto-ku, 135-8548 Tokyo, Japan, m106009@sic.shibaura-it.ac.jp)

Near Field Acoustic Holography (NAH) is the method of measuring sound information on the radiant surface in the near field as a hologram. By use the technique, we can visualize the vibration of the radiant surface. Thus, we can easily understand the object of the acoustic conditions in acoustic image processing. We made the measurement on one side of the drum using array microphone while the driven source is attached to the other side. In addition, the measurement is not touched on the vibrating membrane surface. This paper describes the estimated results of the vibration mode on the vibration membrane of a drum. And this is possible by apply it to the vibration controlling field for detecting the sound structure on the membrane.

9:20

2aSAb5. Computing high frequency vibrations of polygonal plates by the Image Source Method. Jacques Cuenca (LAUM, CNRS, Université du Maine, Lab. d'Acoustique Université du Maine, UMR CNRS 6613, 72085 Le Mans Cedex 9, France, Jacques.Cuenca.etu@univ-lemans.fr), François Gautier (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, francois.gautier@univ-lemans.fr), Laurent Simon (LAUM, CNRS, Université du Maine, Lab. d'Acoustique Université du Maine, UMR CNRS 6613, 72085 Le Mans Cedex 9, France, Laurent.Simon@univ-lemans.fr)

The aim of this paper is to show that the Image Source Method (ISM) can be used for predicting medium and high frequency vibrations of arbitrarily shaped polygonal plates with controllable precision. Modeling the vibrations of polygonal plates by ISM consists in superposing the contributions of elementary image sources, which allows the plate boundary conditions to be satisfied. For particular plate shapes leading to a spatially periodic pattern of image sources, it is shown that ISM allows the exact computation of the Green's function by means of a Poisson summation formula. For arbitrarily polygonal plates, a criterion for truncating the image source generation process is examined and used as a control parameter for the precision of the estimations. The given examples show that the results are in good agreement with exact analytical solutions for simple plate shapes and with results obtained by the finite element method for more complex shapes. The convergence towards reference solutions as the number of image sources is increased is investigated. The main advantage of the method is that the accuracy is improved with structural damping and with frequency for a given number of image sources.

9:40

2aSAb6. Estimation of received acoustic levels in the near-field of a ship. Christopher Barber (Penn State University, PO Box 30, State College, PA 16804-0030, USA, cbarber@psu.edu)

Measured acoustic levels at the face of a receiver in close proximity to a ship in shallow water can differ substantially from the levels predicted by applying standard propagation models to a source level determined from a far-field radiated noise measurement. At distances typical of mine-ship engagements, the dimensions of the source (ship) can be large with respect to range, ocean depth and acoustic wavelength, and the ship cannot be approximated as a single point source, nor is the receiver necessarily located outside the acoustic near field. Direct free-field measurements of ship radiated noise at short range are problematic, so that the received acoustic level in the near-field of a ship must be estimated from other available information. Options for generating such estimates are presented, including a hybrid approach combining empirically-derived transfer functions, a virtual distributed-source representation of the ship, and in-situ hull vibration measurements. The potential for computational methods such as BEM - FEM and measurements such as Nearfield Acoustic Holography to provide an improved understanding of the vibration sources, structural responses, radiation mechanisms and propagation paths associated with the radiated noise field of realistic ship structures is also explored. (Work sponsored by ONR Code 331).

10:00

2aSAb7. Some notes on the sound reduction index of pax cabins panels on cruise ships. Marco Biot (DINMA, University of Trieste, Via Valerio, 10, 34100 Trieste, Italy, biot@units.it), Francesco De Lorenzo (Fincantieri SpA, Passeggio S. Andrea, 6, 34100 Trieste, Italy, Francesco.Delorenzo@fincantieri.it)

The issue of comfort of passenger ships has became in the last years of paramount importance; in this context the noise and vibration control plays a leading role. The concept of comfort on board is subjective and it is impossible to define it with simple formula. On the other hand, a few number of significant parameters is commonly used to characterize the level of comfort on board of ships, and the sound insulation index in one of the most important. As known, it has a paramount importance in defining the contractual comfort on board of ships. The report is related with the problem of setting up the sound insulation index for pax cabins, specifically in relation to the nature and characteristics of the panels between cabins.

10:20-10:40 Break

10:40

2aSAb8. Acoustic and vibration measurements of NASA spacesuits. Durand R. Begault (NASA Ames Research Center, Mail Stop 262-2, NASA ARC, Moffett Field, CA 94035, USA, Durand.R.Begault@nasa.gov), James L. Hieronymus (NASA Ames Research Center, Mail Stop 262-2, NASA ARC, Moffett Field, CA 94035, USA, JimH@riacs.edu), Bernard D. Adelstein (NASA Ames Research Center, Mail Stop 262-2, NASA ARC, Moffett Field, CA 94035, USA, Bernard.D.Adelstein@nasa.gov)

Research was conducted to examine interactions between acoustic signals and mechanical factors relevant to the design of improved auditory displays and noise mitigation for future NASA Spacesuits. Data were collected at NASA Johnson Space Center for the current Mark III advanced prototype surface suit during walking, walking with arm motion, and while seated. Measurements were made using multiple microphones (both head mounted and helmet mounted) and a tri-axial accelerometer to evaluate the effects of suit-borne vibration on overall background noise. Footfall impacts were found to create temporary depolarization of condenser microphone signals, and microphone positioning had a strong effect on signal-noise ratio, partially due to the effect of the helmet enclosure. The results indicate noise mitigation and signal conditioning strategies for improving lunar suit audio voice communications under extra-vehicular (EVA) conditions. Funded by a directed research program of NASA's Space Human Factors Engineering project and by NASA-JSC's Constellation EVA system audio group.

11:00

2aSAb9. Influence of material properties on sound radiation for flat loudspeakers. Jaime Ramis Soriano (DFISTS. Univ. de Alicante, Carretera de Sant Vicent del Raspeig s/n, 03690 San Vicente del Raspeig, Spain, jramis@ua.es), Jesús Alba Fernandez (Escola Politècnica Superior de Gandia, Universitat Politècnica de València, Crtra Natzaret-Oliva s/n, 46730 Gandia, Spain, jesalba@fis.upv.es), Jorge Frances Monllor (DFISTS. Univ. de Alicante, Carretera de Sant Vicent del Raspeig s/n, 03690 San Vicente del Raspeig, Spain, jfmonllor@ua.es), Jorge P. Arenas Bermudez (Universidad Austral de Chile, Institute of Acoustics, 567 Valdivia, Chile, jorgep.arenas@gmail.com)

This work aims to analyze numerically, analytically and experimentally, the influence on sound power radiated, frequency response, and directionality of different materials in flat loudspeakers. Concerned experimental setup, the excitation force consists in a shaker which his mechanical characteristics are completely known. Finite Element Method has been used to simulate this configuration .The analytical model used couples the electrical, mechanical and acoustical system of the panel. The parameters of the panel material which we are focused are elastic modulus and loss factor. The analytical formalism serves as the basis to study the structural and vibrational behaviour of the system that also is analyzed with experimental results. The results obtained from different materials and different excitation position serves as basis for optimum configuration in this type of speakers.

11:20

2aSAb10. Acoustical Effectiveness of Damping Coating. Raymond Fischer (Noise Control Engineering Inc, 799 Middlesex Tnpk, Billerica, MA 01821, USA, rayf@noise-control.com), Leonid Boroditsky (Noise Control Engineering Inc, 799 Middlesex Tnpk, Billerica, MA 01821, USA, leob@noise-control.com)

Damping coatings have again become popular in the shipbuilding industry to reduce onboard and underwater noise. Understanding of the actual effectiveness of damping tiles applied to ship structures is important for noise prediction during the design stage and for optimizing noise control. The commonly held opinion that energy loss is proportional to loss factor values is not confirmed by tests conducted on actual structures. This paper discusses differences between loss factors measured on the Oberst beam and on two-dimensional real structures. The paper shows how loss factor change may influence transmission loss, radiation efficiency and acceptance. The results of loss factor measurements in different conditions including water-loaded media are demonstrated. influences of resonant and non- resonant modes on damping tile effectiveness is discussed. Corrections for SEA noise prediction algorithms, taking into account results of this research, are proposed.

11:40

2aSAb11. On the prediction of absorption coefficient of porous materials with Finite Elements. Sabine Langer (TU Braunschweig, Institute of Applied Mechanics, Spielmannstr. 11, 38106 Braunschweig, Germany, s.langer@tu-bs.de), Meike Wulkau (TU Braunschweig, Institute of Applied Mechanics, Spielmannstr. 11, 38106 Braunschweig, Germany, m.wulkau@tu-bs.de)

The absorption coefficient gives the relation between incited and reflected sound power on a surface and is a measure for the damping properties of materials. Experimentally it can be investigated by using the impedance tube for perpendicular incitation and the echo chamber in the case of a diffuse sound field as is known. To optimize the sound absorption properties of materials in the pre-prototype stage, numerical simulation can support the design of materials. A detailed finite element simulation based on Biots Theory for poroelastic-media is used to predict the absorption coefficient of materials with open-pored surfaces. Prospects and limits of this strategy are discussed.

12:00

2aSAb12. Ground-borne vibration case studies in residential constructions. James Perry (Cerami & Associates, 404 Fifth Avenue, New York, NY 10018, USA, jperry@ceramiassociates.com)

Case studies are examined for several new high-rise residential and hotel buildings proposed for construction over or near underground and surface transit systems. Baseline vibration measurements were taken on grade and existing building structures and projected against standard functional vibration criteria (ANSI S3.29 / ISO 2631 2). Through post-construction testing and client feedback, the vibration and structure-borne noise impacts were field verified. Study results offer insight into the correlation between these vibration criteria and the subjective human perceptions of various sensitivities. Further items of discussion include the relative effects of various architectural constructions, building types, and applications as well as mitigating sound and vibration controls.

Session 2aSAc

Structural Acoustics and Vibration and EURONOISE: Source Characterization in Structure Borne Noise Problems I

Evan Davis, Cochair
The Boeing Company, P.O. Box 3707, Seattle, WA 98124, USA

Charles Pezerat, Cochair
Laboratoire Vibrations Acoustique - INSA Lyon, 25 bis avenue Jean Capelle, Bâtiment Saint-Exupéry, Villeurbanne cedex, F-69621, France

Invited Papers

11:00

2aSAc1. Structure-borne sound transmission from machines into ribbed structures. Barry M. Gibbs (University of Liverpool, School of Architecture, Abercromby Square, L693BX Liverpool, UK, bmg@liv.ac.uk), Andreas R. Mayr (Stuttgart University of Applied Sciences, Schellingstrasse 24, 70174 Stuttgart, Germany, andreas.mayr@hft-stuttgart.de)

The total structure-borne sound power of an installed machine is a function of the source activity and mobility, and the receiver mobility, at each contact and for each component of excitation. The data and computational requirements for prediction therefore are large. Manufacturers view their products as single entities and desire corresponding single values of source strength. A laboratory reception plate measurement procedure has been proposed which yields single equivalent values of source strength and source mobility. The source data, in combination with an estimate of the single equivalent value of receiver mobility, yields the approximate total installed power. The accuracy of the estimate is dependent on the spatial variation in contact conditions over the connections. In addition, phase information has been lost in the simplification. Case studies are described for multiple contact sources on a non-homogeneous plate (a timber-joist floor) where the approximate estimates of structure-borne power are compared with exact values obtained from full mobility formulations.

11:20

2aSAc2. Time domain identification of loads on plate-like structures using an array of acoustic velocity sensors. Quentin Leclere (Laboratoire Vibrations Acoustique - INSA Lyon, 25 bis avenue Jean Capelle, Bâtiment Saint-Exupéry, F-69621 Villeurbanne cedex, France, quentin.leclere@insa-lyon.fr), Charles Pezerat (Laboratoire Vibrations Acoustique - INSA Lyon, 25 bis avenue Jean Capelle, Bâtiment Saint-Exupéry, F-69621 Villeurbanne cedex, France, charles.pezerat@insa-lyon.fr)

The FAT (Force Analysis Technique) method has been developped to identify loads on beams or plates from the knowledge of their flexural displacements. The method is based on a local discretisation of the differential operator of the studied structure : all derivatives of the displacement field are assessed at a given point from a finite difference scheme. The estimation of the operator gives as an output the value of the force distribution. Up to now, applications of the FAT method have been made in the frequency domain, scanning the studied structure with accelerometers or with a laser vibrometer, and using phase references to get the phase relation between different points. The aim of the present study is to show that the FAT method allows to identify loads in the time domain. This operation requires the simultaneous measurement of at least 13 points on the plate, that can be realized without contact using an array of acoustic velocity sensors in the very near field of the plate. The method has been applied on a plate excited by an acoustic diffuse field. The identified force distribution is compared to the parietal acoustic pressure measured in the reverberant room.

11:40

2aSAc3. Measurements of the bending moment at boundaries of a structure. Simon Chesne (LaMCoS - INSA-Lyon - CNRS UMR5259, 18-20, rue des Sciences, Bâtiment Jean d'Alembert, F-69621 Villeurbanne, France, simon.chesne@insa-lyon.fr), Baptiste Chomette (LaMCoS - INSA-lyon - CNRS UMR5259, 18-20, rue des Sciences, Bâtiment Jean d'Alembert, F-69621 Villeurbanne, France, baptiste.chomette@insa-lyon.fr), Charles Pezerat (Laboratoire Vibrations Acoustique - INSA Lyon, 25 bis avenue Jean Capelle, Bâtiment Saint-Exupéry, F-69621 Villeurbanne cedex, France, charles.pezerat@insa-lyon.fr)

In the vibration transmission process, the part due to the moment excitation is often neglected, because of the difficulty to measure them, even if these terms are sometimes important. Indeed, several studies show that the influence of the moment in structure borne power transmission can be higher than simple force. Moment measurement or identification is an old problem especially at boundaries. It has been mainly investigated in mobility methods. In fact, bending moment expression can be seen as spatial derivatives of displacements. These derivatives can be approximated from measured displacements but two major difficulties appear: derivatives are highly sensitive to measurement errors and the usual methods used to obtain them (finite differences, modal approach, etc.) are not well adapted at boundary points. In this paper, three different moment identification methods are investigated where the considered structure is a beam. Both of these approaches are already developed by authors, the third method is based on the use of particular mechanical and electric setups of piezoelectric patches. These 3 methods are numerically and experimentally implemented allowing us to compare and discuss on results.

12:00

2aSAc4. Characterisation of structure borne sound sources from measurement in-situ. Andrew Elliott (University of Salford, Acoustic Research Centre, Newton Building, M5 4WT Salford, UK, a.s.elliott@pgr.salford.ac.uk), Andy T. Moorhouse (University of Salford, Acoustic Research Centre, Newton Building, M5 4WT Salford, UK, a.t.moorhouse@salford.ac.uk)

In-situ source characterisation methods are those which involve measurements made whilst source and receiver are coupled as they would be in a real installation. Potentially in-situ source characterisation may account for the physical reality lost in the ″black box″ approach. There are other potential benefits such as ease of measurement. In this work a structure borne sound source is characterised using in-situ measurements of blocked force and coupled mobility. Promising results from the method have been presented previously. Further to this, an extension of the method allowing the use remote measurement positions has been developed. Using reciprocity, the extended method will further ease measurement for situations where access poses a problem. The extended method is outlined and some preliminary validation results are presented.

12:20

2aSAc5. General model of a structure-borne sound source and its application to shock vibration. Yuri Bobrovnitskii (Mechanical Engineering Research Institute, 4, M. Kharitonievky Str., 101990 Moscow, Russian Federation, yuri@imash.ac.ru), Tatiana Tomilina (Mechanical Engineering Research Institute, 4, M. Kharitonievky Str., 101990 Moscow, Russian Federation, tatiana@imash.ac.ru)

One of the well-known models of structure-borne sound sources is described by the quantities defined at the source-receiving structure interface: by the matrix of the source impedances or mobilities and by the vector of the blocked forces or free velocities. Together with the receiver impedances or mobilities it allows one to predict the vibration field in the receiver and to develop and examine candidates for source characterization. In this paper, it is shown analytically that this model is valid when the sound is generated inside the source by a kinematic excitation as well as by force excitation or their combination and, thus, it represents a rather general model with wide range of applications. Most attention is paid to its peculiarities and limitations when applied to shock sources. Results are illustrated by the data measured in the shock tests of some scientific space instruments.

TUESDAY MORNING, 1 JULY 2008 ROOM 240, 8:20 TO 10:40 A.M.

Session 2aSCa

Speech Communication: Speech Technology II

Indranil Dutta, Chair
Rice University, Center for the Study of Languages, MS #36, 6100 Main St., Houston, TX 77005, USA

Contributed Papers

8:20

2aSCa1. Evaluation of acoustic environments using deteriorated speech sound. Yoshiki Nagatani (Nara Medical University, 840 Shijo-cho, 634-8522 Kashihara, Japan, naramed-u@nagatani.ne.jp), Takefumi Sakaguchi (Nara Medical University, 840 Shijo-cho, 634-8522 Kashihara, Japan, t-saka@naramed-u.ac.jp), Hiroshi Hosoi (Nara Medical University, 840 Shijo-cho, 634-8522 Kashihara, Japan, hosoi@naramed-u.ac.jp)

Aged or hearing-impaired people require better acoustical environments for higher QOL. Many methods of evaluating the quality of speech in acoustical environments have been developed. However, since they mainly focus on quality of speech in bad conditions (e.g. environments with huge noise or long reverberation), they are not suited for evaluations in generic environments such as normal houses or public facilities for aged people. For instance, the scores of D value (deutlichkeit) or speech transmission index (STI) are too high and not sensitive enough to refer to such environments. The intelligibility test using normal speech sounds cannot clarify the differences between different room environments because the intelligibility scores reach almost 100 percent in ordinary room environments. Therefore, we proposed a new evaluation method using deteriorated speech sounds. In this method, signal-processed speech sounds are presented to trial subjects under target sound environments. In this study, Japanese monosyllabic speech sounds convoluted by the impulse responses of room reverberations were presented through a headphone in order to simulate the architectural acoustic environments. As a result, it was shown that this new method could detect the small difference of sound environments, which the conventional methods could hardly evaluate.

8:40

2aSCa2. Measurement of Speech Intelligibility Using Low Level Output - Threshold Efficient S/N Ratios. Christos Nestoras (London South Bank University, FESBE, Borough Road, SE1 0AA London, UK, nestorasc@gmail.com), Stephen Dance (London South Bank University, FESBE, Borough Road, SE1 0AA London, UK, dances@lsbu.ac.uk)

Excessive background noise levels or perhaps annoyance due to a high test signal level are the main reasons for having a low signal to noise ratio (S/N) or low level output during a measurement session. A significant error, in this sense, can be introduced in the resulting data since the measurement technique requires, among others, a minimum S/N ratio for an accurate result. A validation is presented of low level output intelligibility measurements in an attempt to establish a point of reference for the verification of data accuracy for a given space. An indication of the functions taking place in this respect could be obtained through practical experimentation. Results are reported here.

9:00

2aSCa3. Vowels recognition using mellin transform and plp-based feature extraction. Mahdi Jamaati (Technical University of Shahrood, 12345 Shahrood, Iran, mahdi.jamaati@gmail.com), Hossein Marvi (Technical University of Shahrood, 12345 Shahrood, Iran, h_marvi@shahroodut.ac.ir), Milad Lankarany (Technical University of Shahrood, 12345 Shahrood, Iran, milad.lankarany@gmail.com)

Feature extraction for speech recognition is a subject of majer interest today. Different feature have been investigated in speech recognition systems. The scale transform is a particular restriction of the Mellin transform. The key property of the scale transform is the scale invariance .The mel frequency(MFCC) and perceptual linear predicate (PLP) have usually reported to have yielded good performance. In this paper, a new method is presented which combines feature extracted from mellin transform with the plp features. To evaluate the performance of the proposed features the comparative tests with the lpc features, MFCC and PLP were performed on the vowels based speech recognition system. Preliminary experiment show that this approach posses promissing result.

9:20

2aSCa4. A system for automatic detection and correction of detuned singing. Michał Lech (Gdansk University of Technology, Multimedia Systems Department, 11/12 Gabriela Narutowicza Street, 80-952 Gdansk, Poland, mlech@sound.eti.pg.gda.pl), Bozena Kostek (Gdansk University of Technology, Multimedia Systems Department, 11/12 Gabriela Narutowicza Street, 80-952 Gdansk, Poland, bozenka@sound.eti.pg.gda.pl)

The aim of the paper is to show a system engineered for automatic detection and correction of detuned singing. For this purpose, existing methods of fundamental frequency detection and pitch correction are reviewed. In addition, main characteristics of some existing detuning systems are presented. As algorithms for fundamental frequencies detection and pitch correction, the fast autocorrelation and HPS (Harmonic Product Spectrum), and the modified phase vocoder and PSOLA (Pitch-Synchronous Overlap-Add) are chosen and examined. Four possible combinations of the algorithms are reviewed not only in the context of fundamental frequency detection and pitch shifting correctness but also with regard to the quality of the resulting singing signal. Experiments are performed on both male and female singing samples consisting of a variety of tones and various articulations. Basing on the obtained results, it is concluded that the HPS and PSOLA algorithms are the optimum choice as means to correct detuned singing. In addition, listening tests are performed in order to confirm objective measurements of pitch detection and correction. The system is implemented in JAVA. Conclusions are drawn and proposals of improvements are provided.

9:40

2aSCa5. Spectrotemporal and f0 evidence for a theory of enhancement. Indranil Dutta (Rice University, Center for the Study of Languages, MS #36, 6100 Main St., Houston, TX 77005, USA, indranil.dutta@rice.edu)

According to Keyser & Stevens, 2006, apart from universal features that are utilized to calculate motoric instructions, there is a second parallel and language-specific process called enhancement, where additional motoric instructions are processed for the enhancement of the saliency of features that are in danger of being obliterated. The ″Standard View″ on the distinction between voiced stops (VS) and voiced aspirated stops (VAS) in Hindi proposes that the breathy-murmured release following VAS is sufficient to make the contrast between the VS and VAS (Ladefoged & Maddieson, 1996). Hence, audible distinctions between VAS and VS during closure (CD/VLT) are not relevant in maintaining the contrast. Results from our studies show that CD is a relevant cue in making a distinction between aspirated and unaspirated stops. Spectral intensity measures show that speakers employ different glottal configurations to obtain the breathy/modal voicing contrast. VLT durations are inversely correlated with f0, such that longer VLT for VAS leads to further lowering of f_0. This evidence suggests that f_0 perturbations, differences in the durations of closure, and nature and extent of aspiration, all contribute in making the four-way stop contrast possible in Hindi, in support of a theory of enhancement.

10:00

2aSCa6. A Statistical Prosodic Model for Voice Conversion. Jan Schwarz (Institute for Circuit and System Theory, Christian-Albrechts-University of Kiel, Kaiserstrasse 2, 24143 Kiel, Germany, js@tf.uni-kiel.de), Ulrich Heute (Institute for Circuit and System Theory, Christian-Albrechts-University of Kiel, Kaiserstrasse 2, 24143 Kiel, Germany, uh@tf.uni-kiel.de)

Nowadays Text-to-Speech (TTS) systems adapt the output voice to the user and the corresponding application. The aim is a personalisation. Thereby, the user is set into familiar surroundings, increasing the TTS acceptance. For example, an e-mail client that may read the incoming messages with the synthesised voice sounding like that of the sender. Such a personalised TTS system is costly; so, voice-conversion (VC) techniques are used to save resources. VC transforms the voice of a ''source speaker'' in such a way that the converted voice sounds like that of another ''target speaker''. This voice sounds only natural, if it includes all features relevant for the true target voice. Here, a main problem is the mapping of the prosody which is one of the essential features. This contribution introduces a statistical prosodic model for voice conversion. It is based on Gaussian-Mixture Models (GMM), trained for the pitch and the duration of diphones. To ensure sufficient data for the GMM training, seven diphone classes are separated as related to the international phonetic alphabet. The suitability for VC as well as limitations, necessary extensions (stress) and problems are pointed out.

10:20

2aSCa7. Modelling acoustic parameters of prosody for read and acted-speech synthesis. Milan Rusko (Institute of Informatics of the Slovak Academy of Sciences, Dubravska cesta 9, 845 07 Bratislava, Slovakia, milan.rusko@savba.sk), Marián Trnka (Institute of Informatics of the Slovak Academy of Sciences, Dubravska cesta 9, 845 07 Bratislava, Slovakia, trnka@savba.sk), Sakhia Darjaa (Institute of Informatics of the Slovak Academy of Sciences, Dubravska cesta 9, 845 07 Bratislava, Slovakia, utrrsach@savba.sk), Richard Kováč (Institute of Informatics of the Slovak Academy of Sciences, Dubravska cesta 9, 845 07 Bratislava, Slovakia, richard.kovac@savba.sk), Juraj Hamar (Philosophical Faculty, Comenius University, Gondova 2, 818 01 Bratislava, Slovakia, juraj.hamar@chello.sk)

The prosody model is one of the most important parts of every speech synthesizer, influencing mainly its naturalness. The intonation contour and phoneme lengths (together with speech quality) bear a great deal of extralinguistic and paralinguistic information contained in the synthesized speech. The features reflecting personality, mood and emotions of the speaker are in strong interaction with those reflecting speech styles. Anyway the appropriate choose of prosody model and training material can make it possible to create special model for every speaking style. The paper presents our approach to modelling of acoustic parameters of prosody in two different speech styles in Slovak. Our model is based on Classification and regression trees (CARTs). It uses independent CART for phoneme lengths and three CARTs for fundamental frequency (F0) at the beginning, centre, and end of every syllable. Two hours of read speech were used for training a model of read speech. The recordings of a puppet player were used to train a model of acted speech. The models were implemented in the Kempelen 2.2 unit selection Slovak speech synthesizer. The listening tests have shown that the models are capable of modelling significant amount of the differences of the two speaking styles.

Session 2aSCb

Speech Communication: How do Physical and Motor Knowledge Matter to Speech Perception?

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Invited Papers

8:20

2aSCb1. Action-based multisensory integration in striking events. Bruno L. Giordano (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, bruno.giordano@music.mcgill.ca), Stephen McAdams (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, smc@music.mcgill.ca), Paolo Crosato (Dept. of Information Engineering, Univ. of Padova, Via G. Gradenigo 6/A, 35131 Padova, Italy, jimbobaus@virgilio.it), Federico Avanzini (Dept. of Information Engineering, Univ. of Padova, Via G. Gradenigo 6/A, 35131 Padova, Italy, avanzini@dei.unipd.it), Carmine Casciato (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, casciato@music.mcgill.ca), Stephen Sinclair (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, sinclair@music.mcgill.ca), Marcelo M. Wanderley (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, marcelo.wanderley@mcgill.ca)

Many theories of multisensory integration focus on action-independent perception. This approach has limited ecological validity, because under everyday conditions we frequently act upon multisensory events through motor behavior. We present a novel methodology for the study of multisensory integration in action-directed perception, focusing on striking events. Observers repeatedly hit a virtual object with a target striking velocity, and are presented with various types of sensory information about the striking event: auditory and/or haptic and/or visual. For each of the experimental trials, they initially receive feedback on the tracking of the target striking velocity. In a second phase, feedback is eliminated. In a third phase, the sensory properties of the striking event are changed. We quantify the extent to which a variation in each of the sensory modalities disrupts performance in the tracking of the target striking velocity. Multisensory dominance hierarchies are inferred from the comparison of the patterns of performance disruption in unimodal and multisensory contexts.

8:40

2aSCb2. Amodal specification of talker-specific motor behavior. Lawrence D. Rosenblum (University of California, Riverside, Department of Psychology, 900 University Ave., Riverside, CA 92521, USA, rosenblu@citrus.ucr.edu)

There is substantial evidence that the speech perception function incorporates information for articulatory motor behavior. Research over the last 15 years has shown that speech perception also uses information for talker-specific motor behavior. Interestingly, this research reveals that talker information can facilitate speech perception whether it is apprehended through auditory or visual speech (lipread) means. The multimodal nature of these effects could mean that the talker-specific properties used by the system are amodal and motoric-or gestural-in nature. Evidence for this proposition-that talker-specific facilitation of speech is based on amodal specification of gestural style-will be discussed. This evidence includes findings that isolated talker-specific phonetic information, available both optically and acoustically, can be informative about talker identity to the degree that it supports cross-modal talker matching. Other evidence shows that the talker familiarity gained through one modality can facilitate perception of speech in the other. Finally, recent evidence reveals that speech perceivers will align to (inadvertently imitate) talker-specific properties of utterances they are asked to shadow, whether that speech is presented auditorily or visually. These results suggest that some of the 'motor knowledge' bearing on speech perception takes the form of talker-specific gestural properties, amodally specified.

9:00

2aSCb3. Cross-modal synchrony perception reveals aspects of categorical perception. Armin Kohlrausch (Philips Research Europe, Digital Signal Processing (MS WO02), High Tech Campus 36, 5656 AE Eindhoven, Netherlands, armin.kohlrausch@philips.com), Rob L. Van Eijk (Technische Universiteit Eindhoven, Human-Technology Interaction, P.O. Box 513, 5600 MB Eindhoven, Netherlands, r.l.j.v.eijk@tue.nl), Steven Van De Par (Philips Research Europe, Digital Signal Processing (MS WO02), High Tech Campus 36, 5656 AE Eindhoven, Netherlands, steven.van.de.par@philips.com), James F. Juola (University of Kansas, Department of Psychology, 1415 Jayhawk Blvd., Lawrence, KS KS 66045, USA, juolas@ku.edu), Michael Vitevich (University of Kansas, Department of Psychology, 1415 Jayhawk Blvd., Lawrence, KS KS 66045, USA, mvitevit@ku.edu)

The phenomenon of ″categorical perception″ has played an important role in speech research. When a specific (combination of) feature(s) of a speech stimulus is varied along a physical dimension, categorical perception is reflected by two observations: (1) the percept of the sound changes abruptly from one category to another, e.g., from ″ba″ to ″da″ to ″ga″ for changes in formant transition frequencies, and (2) physical changes of a given amount lead to more easily perceivable differences for stimuli close to a category boundary, compared to stimuli in the center of a category. In this talk, I present data about audio-visual synchrony perception, which indicate that perceived synchrony also reveals properties of categorical perception. Depending on the physical delay between the auditory and visual component, the percept changes from ″audio first,″ to ″synchronous″ to ″video first.″ When measuring sensitivity to changes in audio-visual delays, we observe that the thresholds are small for stimuli at the transitions between the earlier determined categories, while they are larger for stimuli within a category.

9:20

2aSCb4. Do we perceive articulatory gestures when we listen to speech? Pierre Divenyi (VA Northern Calif. Health Care Syst. and East Bay Inst. for Research and Education, Bldg. R4, 150 Muir Rd., Martinez, CA 94553, USA, pdivenyi@ebire.org), Adam Lammert (VA Northern Calif. Health Care Syst. and East Bay Inst. for Research and Education, Bldg. R4, 150 Muir Rd., Martinez, CA 94553, USA, alammert@ebire.org)

Articulatory synthesis methods, classic and contemporary, have demonstrated that it is possible to generate speech from an ensemble of functions derived from articulatory gestures. Such gesture-to-waveform transforms suggest that, inversely, the speech signal should be also decomposable into the same set of gesture, or gesture-like, functions. These functions vary slowly in time and their association with the speech waveform (words as well as sentences) can be established by machine learning algorithms. In a recent study at our laboratory, listeners were asked to type the word or the sentence they heard, with speech (degraded in diverse ways) as the stimulus. The subjects' responses were synthesized, time-aligned with the stimulus, and decomposed into a set of eight gestures, as specified by the Haskins Laboratories TADA system (http://www.haskins.yale.edu/tada_download/index.html). When the running distance between input and response gesture functions is calculated, results indicate a significant degree of gesture information transmitted even during severely degraded speech segments, suggesting that the perceptual system may track speech via underlying functions similar to gestures. Epochs at which this running distance estimate fails, i.e., exceeds a certain threshold, may be considered to signal periods during which insufficient bottom-up information had to be supplemented using higher-order linguistic knowledge.

9:40

2aSCb5. Hearing the tongue and lips of vowel gestures: A new differential paradigm. Jean-Luc Schwartz (ICP-GIPSA, INPG, 46 Av. Félix Viallet, 38031 Grenoble, France, schwartz@icp.inpg.fr), Nathalie Vallée (ICP-GIPSA, INPG, 46 Av. Félix Viallet, 38031 Grenoble, France, vallee@icp.inpg.fr), Sonia Kandel (LPNC, Université Pierre Mendès-France - BP 47, 38040 Grenoble, France, Sonia.Kandel@upmf-grenoble.fr)

It is an old question to know to what extent a listener can recover the articulatory dimensions of a speaker's gesture. In the case of vocalic configurations, a number of experiments have been done on expert phoneticians, showing that vowel height can be reasonably well estimated from the sound, but the front-back and lip rounding dimensions are much less well recovered. However, almost nothing has been done on naive listeners, due to the difficulty to perform absolute estimations in the lack of explicit phonetic knowledge. In the past years, we have developed an original paradigm, exploiting differential rather than absolute estimations. We show that French listeners, even naïve, are able to discriminate to a certain extent which vowel in a given pair has a higher vs. lower or more front vs. more back tongue position, or more or less rounded lips. From these data, we have elaborated an algorithm enabling to estimate what are the internal representations of vowel height, frontness and rounding, and correlated these representations with acoustic parameters, F1, F2-F1 and F'2 in Barks appearing to play a key role in the auditory recovery of these three motor dimensions.

10:00

2aSCb6. Mathematical Evidence For Motor Theories of Speech Perception. Gordon Ramsay (Haskins Laboratories, 300 George Street, New Haven, CT 06511, USA, ramsay@haskins.yale.edu)

Theories of speech perception have often proposed, controversially, that recovery of phonological information from the acoustic signal necessarily requires implicit knowledge of the physical processes underlying speech production. Most of the evidence supporting this hypothesis has been derived empirically from experiments in speech perception. In this paper, we show that many of the basic ideas underlying motor theories of speech perception can also be derived mathematically from first principles, using classic results in stochastic nonlinear filtering theory. Two key results, the Duncan-Mortenson-Zakai and Fujisaki-Kallianpur-Kunita theorems, show that the optimal state estimator for any partially-observed nonlinear stochastic dynamical system always takes the form of a ″matched filter″, which is itself a nonlinear stochastic dynamical system, the structure of which mimics, and resonates with, the structure of the original. Interpreting this in the context of speech perception, recovery of phonological information from sound produced by a human vocal tract necessarily involves construction of an internal model of the processes implicated in speech production. A key prediction, which we explore, is that any such model need not reproduce all the details of these processes, but is only required to predict the lawful conditional correlation between gesture and sound.

Session 2aSCc

Speech Communication: Speech Recognition in Noisy Environments

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Contributed Paper

10:40

2aSCc1. Speech recognition with body-conducted speech using differential acceleration. Masashi Nakayama (Hiroshima City University / National Institute of Advanced Industrial Science and Technology, 3-4-1 Ozuka-Higashi, Asa-Minami-Ku, 731-3194 Hiroshima, Japan, m.nakayama @aist.go.jp), Shunsuke Ishimitsu (Hiroshima City University / National Institute of Advanced Industrial Science and Technology, 3-4-1 Ozuka-Higashi, Asa-Minami-Ku, 731-3194 Hiroshima, Japan, ishimitu @hiroshima-cu.ac.jp), Seiji Nakagawa (National Institute of Advanced Industrial Science and Technology (AIST), 1-8-31 Midorigaoka, 563-8577 Ikeda, Osaka, Japan, s-nakagawa@aist.go.jp)

Speech-recognition rates decrease in noisy environments. The body-conducted speech, conducted in solids such as body and skins, has a noise-robust characteristics and can be served for recognition systems even in 98 dBSPL-noise (-20 dBSNR) environments. However, the body-conduction could not capture high frequency sounds. Conventional methods for the improvement in sound quality of body-conducted speeches needs both speeches and body-conducted speeches. In this paper, a new body-conducted speech retrieval technique in sound quality without a speech signal itself is proposed. First, high-frequency components in the body-conducted speech were emphasized using differential acceleration. Second, conventional noise reduction method was adopted to make a clear body-conducted speech from a retrieval speech which contains constant noise. The recognition experiments using the proposed method showed that it improved recognition rate in all speakers.

Invited Paper

11:00

2aSCc2. Exploiting confidence measures for missing data speech recognition. Christophe Cerisara (LORIA UMR 7503, Campus Scientifique, 54506 Vandoeuvre-lès-Nancy, France, Christophe.Cerisara@loria.fr)

Automatic speech recognition in highly non-stationary noise, for instance with a competing speaker or background music, is an extremely challenging and still unsolved problem. Missing data recognition is a robust approach that is well adapted to this kind of noise. A standard missing data technique consists in marginalizing out, from the observation likelihoods computed during decoding, the contribution of the spectro-temporal fragments that are dominated by noise. However, such an approach can hardly be applied to advanced parameterization domains that do not separate speech from noise frequencies, such as the cepstrum or ETSI AFE. We propose in this work to extend this technique to such parameterization domains, and not only to spectrographic-like front-ends as it was the case before. This is realized by masking the observations that favor erroneous decoding paths, instead of masking the features that are dominated by noise. These new missing data ″masks″ are now estimated based on speech recognition confidence measures, which can be considered as indicators of the reliability of decoding paths. A first version of this robust algorithm is evaluated on the French broadcast news ESTER corpus.

Contributed Paper

11:20

2aSCc3. An MTF-based blind restoration of temporal power envelopes as a front-end processor for automatic speech recognition systems in reverberant environments. Xugang Lu (Japan Advanced Institute of Science and Technology, 1-1, Asahidai, Nomi, 923-1292 Sendai, Japan, xugang@jaist.ac.jp), Masashi Unoki (JAIST, 1-1 Asahidai, 923-1292 Nomi, Japan, unoki@jaist.ac.jp), Masato Akagi (Japan Advanced Institute of Science and Technology, 1-1, Asahidai, Nomi, 923-1292 Sendai, Japan, akagi@jaist.ac.jp)

To reduce speech degradation in reverberant environments, we previously proposed a modulation transfer function (MTF) based method for speech dereverberation. It is based on the MTF relation that the sub-band temporal power envelope of reverberant speech can be represented as the convolution between temporal power envelopes of clean speech and the room impulse response. Therefore, the sub-band power envelope of clean speech can be estimated using inverse MTF filtering without measuring the room impulse response. We tested the effectiveness of this method as a front-end for automatic speech recognition (ASR) in both artificial and real reverberant environments. Reverberant speech signals were created by simple convolution of clean speech (AURORA-2J) and artificially-produced or real room impulse responses. The relative spectral filtering of the auditory-power-spectrum based method was used as a baseline. Compared with the baseline, our proposed method had 36.64% and 21.68% improvements in error reduction rate for artificial reverberant environments (reverberation times from 0.2 to 2.0 s) and real reverberant environments (43 reverberant impulse responses), respectively. These results indicate that our proposed method can be used as a robust front-end for ASR. [Work supported by a Grant-in-Aid for Science Research from the Japanese Ministry of Education (No. 18680017).]

Invited Paper

11:40

2aSCc4. Front-end processing of a distant-talking speech interface for control of an interactive TV system. Maurizio Omologo (Fondazione Bruno Kessler - IRST, Via Sommarive, 18, Povo, 38050 Trento, Italy, omologo@fbk.eu)

This work addresses a research activity being conducted for the development of a user-friendly interface for the access to a virtual smart assistant enabling the interaction with TV-related digital devices and infotainment services. In the given scenario, the users can speak in a natural and comfortable way, not encumbered by any hand-held or head-mounted microphone. The environment is typically a living room, equipped with digital TV, Hi-Fi audio devices, etc., and populated by a group of people (e.g., family members). Among the most challenging issues involved in this scenario are a multi-microphone front-end for an effective processing of the given acoustic scene, an Acoustic Echo Cancellation (AEC) component to compensate the sound produced by loudspeakers, and a multi-modal distant-talking spoken dialogue system. As far as the front-end is concerned, multiple speaker localization, speech activity detection, speaker identification, and speech recognition will have to perform accurately even when AEC is applied to the given microphone array. The paper aims to present preliminary results of this research, which is being conducted under the European Project DICIT.

Contributed Paper

12:00

2aSCc5. Speaker adaptation combined with missing data reconstruction. Ulpu Remes (Adaptive Informatics Research Centre, Helsinki University of Technology, P.O. Box 5400, 02015 Espoo, Finland, ulpu.remes@tkk.fi), Kalle J. Palomäki (Adaptive Informatics Research Centre, Helsinki University of Technology, P.O. Box 5400, 02015 Espoo, Finland, kalle.palomaki@tkk.fi), Mikko Kurimo (Adaptive Informatics Research Centre, Helsinki University of Technology, P.O. Box 5400, 02015 Espoo, Finland, mikko.kurimo@tkk.fi)

Missing data methods offer an attractive framework for automatic speech recognition (ASR) under time-varying and unpredictable noise conditions. The motivation for the methods comes from the human speech perception and auditory scene analysis, studies on which have suggested that the more intense sound components in an auditory scene dominate the neural response to the scene. The missing data methods developed for ASR divide the noisy speech signal to speech and noise dominated regions. Speech components in the noise dominated regions are viewed as missing but may be reconstructed based on the observed components. Missing data methods have performed well in previous studies with noise corrupted speech but have not become a standard in ASR. Especially considering ASR on large vocabularies, the methods have a disadvantage: it is not straightforward to combine them with common speaker adaptation methods such as constrained maximum likelihood linear regression (CMLLR). In this work, we investigate efficient ways to combine missing data reconstruction with speaker adaptation in large vocabulary ASR task with speech data recorded in noisy real-world environments. The experiments show that speaker adaptation combined with the missing data reconstruction can improve the missing data approach in noisy large vocabulary ASR.

Invited Paper

12:20

2aSCc6. An efficient frame selection approach to variable frame rate analysis for noise robust speech recognition. Zheng-Hua Tan (Department of Electronic Systems, Aalborg University, Niels Jernes Vej 12, 9220 Aalborg, Denmark, zt@es.aau.dk), Børge Lindberg (Department of Electronic Systems, Aalborg University, Niels Jernes Vej 12, 9220 Aalborg, Denmark, bli@es.aau.dk)

This paper presents a low-complexity, effective variable frame rate (VFR) analysis method that conducts frame selection on the basis of a posteriori signal-to-noise ratio (SNR) weighted energy distance. It has two characteristics. First, energy distance (instead of cepstral distance) is used to make it computationally efficient and thus enable a finer granularity in search as compared with cepstral distance criterion. Secondly, SNR weighting is used to emphasize the reliable regions in noisy speech signals. In terms of frame selection, it is experimentally found that the method is able to assign a higher frame rate to fast changing events such as consonants, a lower frame rate to steady regions like vowels and no frames to silence, even for very low SNR signals. The VFR method is applied to speech recognition in noisy environments to improve noise robustness. Being a method that takes effect in the time-domain, it is moreover combined with spectral- and cepstral-domain techniques to gain further improvement. Experiments are conducted on the Aurora 2 database, which is the TI digits database artificially distorted by adding different noises, and very encouraging results are obtained.

Session 2aSCd

Speech Communication: Speech Perception I

Mirjam Broersma, Chair
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Contributed Papers

11:00

2aSCd1. Phonological specificity of vowels and consonants in 20-month-olds' word representations. Anders Højen (Univ. of Southern Denmark, Dept. Lang. Communication, Campus Vej 55, 5230 Odense, Denmark, anders@andershojen.dk), Thomas O. Madsen (Univ. of Southern Denmark, Dept. Lang. Communication, Campus Vej 55, 5230 Odense, Denmark, tom@language.sdu.dk), Werner Vach (Univ. of Southern Denmark, Dept. Lang. Communication, Campus Vej 55, 5230 Odense, Denmark, werner.v@stat.sdu.dk), Torkil Østerbye (Univ. of Southern Denmark, Dept. Lang. Communication, Campus Vej 55, 5230 Odense, Denmark, oesterbye@sdu.dk), Karina F. Christensen (Univ. of Southern Denmark, Dept. Lang. Communication, Campus Vej 55, 5230 Odense, Denmark, kf@language.sdu.dk), Hans Basbøll (Univ. of Southern Denmark, Dept. Lang. Communication, Campus Vej 55, 5230 Odense, Denmark, hba@language.sdu.dk), Sueli Caporali (Univ. of Southern Denmark, Dept. Lang. Communication, Campus Vej 55, 5230 Odense, Denmark, s.caporali@widex.com), Dorthe Bleses (Univ. of Southern Denmark, Dept. Lang. Communication, Campus Vej 55, 5230 Odense, Denmark, bleses @sdu.dk)

At the onset of word learning around the age of 12 months, infants are highly capable of perceiving native phonological distinctions. However, required to make word-object associations, infants sometimes fail to perceive certain phonological distinctions. The present study used an inter-modal preferential looking technique to examine Danish 20-month-olds' sensitivity to mispronunciations involving either a vowel or a consonant in familiar words. Each mispronounced word could be ″reconstructed″ to either of two familiar Danish words. By way of example, the English nonword ″bock″ can be turned into either ″rock″ or ″book″ depending on whether the mispronunciation is perceived to be in the first consonant or in the vowel. The results so far indicate that the infants were not differentially sensitive to vowel vs. consonant mispronunciations. This result, in turn, suggests that vowels and consonants do not differ in constraining lexical access to familiar words in Danish 20-month-olds.

11:20

2aSCd2. Perceptual asymmetries induced by category learning. Laurent Bonnasse-Gahot (CAMS-EHESS, 54 bd Raspail, 75270 Paris Cedex 06, France, lbg@ehess.fr)

This paper takes an information-theoretic approach to study the perceptual consequences of the neural encoding of categories (e.g. vowels). We focus on two well-known psychophysical phenomena: *categorical perception*, characterized by greater cross-category than within-category discrimination, and *perceptual magnet effect*, stating that perceptual space is wrapped around prototypical instances of a given category, leading to a better ability to discriminate stimuli near non-prototypical exemplars of a category than near prototypical ones. Introducing a perceptual distance based on the Kullback-Leibler distance between the patterns of activity evoked by two stimuli, we show that both categorical perception and prototypical effects emerge from maximizing information contained in the neural representation about a set of categories. We discuss the relations between these two psychophysical phenomena and show that they go in hand with another kind of asymmetric effect. The discriminability between a category prototype and a non-prototype is not symmetric, depending on which stimuli serves as a referent. Non-prototypical stimuli are judged closer to prototypical exemplars than the reverse. Quantitative and qualitative comparisons with experimental data and previous theoretical work are presented and discussed.

11:40

2aSCd3. Order effects and peripherality: a cross-linguistic perceptual study using an [i]-[e] articulatory continuum. Charalampos Karypidis (UMR 7018, CNRS / Univ. Paris III, 19, rue des Bernardins, 75005 Paris, France, ch_karypidis@yahoo.com), Antonia Colazo-Simon (UMR 7018, CNRS / Univ. Paris III, 19, rue des Bernardins, 75005 Paris, France, simonantonia@hotmail.com), Angelica V. Costagliola (UMR 7018, CNRS / Univ. Paris III, 19, rue des Bernardins, 75005 Paris, France, angelicacostagliola@yahoo.it), Cirineu Cecote Stein (UMR 7018, CNRS / Univ. Paris III, 19, rue des Bernardins, 75005 Paris, France, cirineustein@uol.com.br), Gilles Guglielmi (ARP / UFRL Univ. Paris VII - Denis Diderot, 30, rue du Chateau des Rentiers, 75013 Paris, France, gillesgug@yahoo.fr)

This paper attempted to provide further evidence on whether peripherality is a universal bias triggering stimulus order effects. Participants from four linguistic systems - Brazilian Portuguese, Salentinian Italian, French and Spanish - participated in the study. In Experiment 1, listeners identified as /i/ or /e/ the ten stimuli of a continuum prepared with an articulatory model. Experiment 2 consisted in an AX 'same-different' discrimination task including pairs of stimuli differing in one or two steps along the continuum. Discrimination was better when the more peripheral stimulus was presented second in a pair. However, these order effects were found mainly within the /i/ category and only for Italian and French. Experiment 3 was a replication of Experiment 2, except that two modifications were introduced: a. identical pairs were added, and b. one- and two-step pairs were separately examined. Order effects were again found but, this time, they covered a wider area of the continuum. Peripherality alone was not able to account for order effects. The role of focalization was later discussed.

12:00

2aSCd4. Perception of an infrequent assimilation: Labial-to-alveolar assimilation in German. Holger Mitterer (Max Planck Institute for Psycholinguistics, Wundtlaan 1, 6525 XD Nijmegen, Netherlands, holger .mitterer@mpi.nl)

In German (and many languages) the alveolar nasal /n/ can assimilate to [m] if followed by a labial (e.g. 'in Berlin' ← ″i[m] Berlin″). In a German spontaneous speech corpus (KielCorpus), one also finds a few cases in which an /m/ followed by an alveolar consonant surfaces as [n] (e.g., 'samstag' ← ″sa[n]stag″, Engl. Saturday). Four experiments investigated whether there is a similar pre-lexical compensation process for these labial-to-alveolar assimilations as previous research uncovered for alveolar-to-labial assimilations. This turns out to be the case: German and Dutch listeners-the latter potentially unfamiliar with this type of assimilation-tend to perceive the assimilated [n] in ″sanstag″ as /n/ if presented in isolation (″an″), but as /m/ if presented with minimal context (″ansta″). For German listeners, this context effect is larger if they hear the complete word ″sanstag″, introducing an additional lexical bias to perceive the [n] as /m/. Finally, phonetic detail such as transitions of the [s] fricative pole are also important: If the assimilated ″an″ is spliced into another /s/-initial syllable by the same speaker, the context effect is reduced. Perceiving infrequent assimilations seems therefore similar to perceiving frequent assimilations.

12:20

2aSCd5. Nonnative listeners prefer perceptual cues they know from their L1: Dutch listeners use vowel duration less than English listeners for English final /v/-/f/. Mirjam Broersma (Radboud University Nijmegen, PO Box 9104, 6500 HE Nijmegen, Netherlands, mirjam.broersma@mpi.nl)

Two 2AFC experiments investigated Dutch and English listeners' use of preceding vowel duration for the English nonword-final /v/-/f/ contrast. Like English, Dutch has a /v/-/f/ contrast, but unlike English, Dutch has no final /v/. Dutch listeners therefore have no native language experience with the use of preceding vowel duration as a final voicing cue. Previous research showed that Dutch listeners used preceding vowel duration less than English listeners when it was invariable in the experiment (Broersma, 2005, JASA, 117, 3890-3901). The present results show that they also used it less when it was varied. An 11-step fricative voicing continuum from a natural /v/ to a natural /f/ and a 7-step vowel duration continuum from a long (originally /v/-preceding) to a short (originally /f/-preceding) vowel were created. In Experiment 1, all steps of the fricative voicing continuum were combined with one long and one short vowel. In Experiment 2, all steps of the vowel duration continuum were combined with the /v/ and /f/ endpoints and with one ambiguous fricative. In both experiments, Dutch listeners used fricative voicing more and vowel duration less than English listeners did. Thus, the nonnative listeners especially used the perceptual cues they knew from their native language.

TUESDAY MORNING, 1 JULY 2008 ROOM 343, 8:00 TO 10:20 A.M.

Session 2aSPa

Signal Processing in Acoustics, Physical Acoustics, Biomedical Ultrasound/Bioresponse to Vibration, and Underwater Acoustics: Overview of Time Reversal in Acoustics I

David Chambers, Cochair
Lawrence Livermore Natl. Lab., L-333, 7000 East Ave., Livermore, CA 94550, USA

Claire Prada, Cochair
Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, Paris, 75005, France

Invited Papers

8:00

2aSPa1. An overview of time-reversal acoustics. Mathias Fink (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, mathias.fink@espci.fr)

Time-reversal invariance is a very powerful concept in physics. In the field of acoustics where time reversal invariance occurs, time-reversal experiments may be achieved simply with arrays of transmit-receive transducers, allowing an incident wave field to be sampled, recorded, time-reversed and re-emitted. Time reversal mirrors (TRMs) may be used to study random media and chaotic reverberating structures. Common to these complex media is a remarkable robustness exemplified by observations that the more complex the medium between the probe source and the TRM, the sharper the focus. TRMs open the way to new signal processings that interest imaging, detection, telecommunications and therapy. Time reversal mirrors have plenty of applications including ultrasonic therapy and medical imaging, non destructive testing, telecommunications, underwater acoustics, seismology, sound control, home automation. An overview of these fields will be presented.

8:20

2aSPa2. Ocean acoustic time reversal. William A. Kuperman (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, wkuperman@ucsd.edu)

A series of ocean acoustic time experiments have been jointly conducted in the last ten years by the NATO Undersea Research Centre and the Marine Physical Laboratory. The experiments demonstrated a robustness of the ocean acoustic time reversal mirror process similar to that found in more controlled, ultrasonic laboratory experiments. Results from these experiments have pointed to potential applications to SONAR signal processing methods and to acoustics communications. In addition, the experimental procedures themselves have yielded data that provide further understanding of the ocean acoustic environment and the increased stability of time reversal process over one way propagation in a fluctuating medium.

Contributed Paper

8:40

2aSPa3. 2004 M6.0 Parkfield earthquake characterization using Time Reversal. Carene Larmat (EES-11 (Geophysics) - Los Alamos National Laboratory, MS D443, Los Alamos, NM 87545, USA, carene@lanl.gov), Paul A. Johnson (EES-11 (Geophysics) - Los Alamos National Laboratory, MS D443, Los Alamos, NM 87545, USA, paj@lanl.gov), Lianjie Huang (EES-11 (Geophysics) - Los Alamos National Laboratory, MS D443, Los Alamos, NM 87545, USA, ljh@lanl.gov)

Time reversal has proved to be a robust source location method in acoustics and is now being developed for a number of seismic applications. One problem of particular interest is locating sources where the signal-to-noise ratio is small. These include small earthquakes (<M5.5) or atypical seismic sources with a small seismic energy radiation (e.g., tremor, slow earthquakes). Time reversal has been shown to be very robust and work in the presence of poor data, low signal to noise ratio, etc. We present a prototype study showing the power of time reversal, using seismic data from the 2004 M6.0 Parkfield earthquake, which is the world's best recorded event to date and thus one of the most studied. The back-propagation of recorded seismic data in a 3D Earth velocity model is numerically carried out. We show that the reconstructed reverse wave-field exhibits clear focusing at the source point but also displays a four-lobe radiation pattern for each type of rebroadcast waves (body, surface), which is consistent with the known source mechanism: a right-lateral strike slip along the almost-vertical San Andrea fault.

Invited Paper

9:00

2aSPa4. Time reversal and subwalength focal spot. Julien De Rosny (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, julien.derosny@espci.fr), Arnaud Tourin (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, arnaud.tourin@espci.fr), Geoffroy Lerosey (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, geoffroy.lerosey@espci.fr), Mathias Fink (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, mathias.fink@espci.fr)

When a wave is time-reversed inside a homogeneous medium, the focal spot width at best equals half a wavelength. This limit comes from the loss of evanescent wave during propagation. An analysis of the time reversed field in terms of the Green's function formalism shows that in order to get a finer spot, not only the field has to be time reversed but also the initial source. In such a case, an acoustic sink is obtained. Experimental results are presented. Then we show two methods to obtain subwavelength focal spot without time-reversed source. The first method consists of setting the time reversal mirror in the near field of the initial source. Despite the evanescent wave transmission, we will see that subwavelength focusing is only observed for a special time-reversal mirror. The second method consists of surrounding the initial source by many scatterers. In such a case, the evanescent waves emitted by the initial source are converted into propagating ones. During the time reversal step, back-conversion occurs that leads to a sub wavelength focal spot. Thanks to this principle, a focal spot of a thirtieth of a wavelength has been reported in the case an electromagnetic experiment.

Contributed Papers

9:20

2aSPa5. Model-based time reversal method for photoacoustic imaging of heterogeneous media. Peter Burgholzer (Upper Austrian Research, Hafenstr. 47, 4020 Linz, Austria, peter.burgholzer@uar.at), Hubert Gruen (Upper Austrian Research, Hafenstr. 47, 4020 Linz, Austria, huber.gruen@uar.at), Robert Nuster (Karl-Franzens-Universität Graz, Universitätsplatz 5, 8010 Graz, Austria, ro.nuster@uni-graz.at), Günther Paltauf (Karl-Franzens-Universität Graz, Universitätsplatz 5, 8010 Graz, Austria, guenther.paltauf@uni-graz.at), Markus Haltmeier (University of Innsbruck, Technikerst. 21a/2, 6020 Innsbruck, Austria, markus.haltmeier @uibk.ac.at)

In photoacoustic (also called optoacoustic or thermoacoustic) tomography acoustic pressure waves are generated by illumination of a semitransparent sample with pulsed electromagnetic radiation. Subsequently the waves propagate towards the detection surface enclosing the sample. The inverse problem consists of reconstructing the initial pressure sources from those measurements. In certain applications of photoacoustic imaging one has to deal with media with spatially varying sound velocity, e.g. bones in soft tissue. Image reconstruction without any compensation of this effect leads to a poor image quality. It is therefore essential to develop reconstruction algorithms that take spatially varying sound velocity into account and are able to reveal small structures in acoustically heterogeneous media. A model-based time reversal reconstruction method is presented that is capable of reconstructing the initial pressure distribution despite variations of sound speed. This reconstruction method calculates the time reversed field directly with a second order embedded boundary method by retransmitting the measured pressure on the detector positions in reversed temporal order. Numerical simulations and experiments with phantoms consisting of areas with spatially varying sound velocity are presented.

9:40

2aSPa6. The effects of transducers on the time reversal process in solids. Brian E. Anderson (EES-11 (Geophysics) - Los Alamos National Laboratory, MS D443, Los Alamos, NM 87545, USA, bea@lanl.gov), Michele Griffa (EES-11 (Geophysics) - Los Alamos National Laboratory, MS D443, Los Alamos, NM 87545, USA, mgriffa@lanl.gov), Paul A. Johnson (EES-11 (Geophysics) - Los Alamos National Laboratory, MS D443, Los Alamos, NM 87545, USA, paj@lanl .gov)

Every experimental implementation of Time Reversal (TR) involves the use of transducers to convert wave motion, whether mechanical or acoustical, into electrical signals, and vice versa. Practical considerations of transducers are not included in the basic theory of time reversal, which is based on idealized point-like sources. These considerations include temporal ring down at a narrowband transducer resonance, the finite size of the transducer giving rise to directivity, and the impedance contrast between the transducer and the medium. The effects of these considerations on the TR process will be characterized by presenting data from various TR experiments.

10:00

2aSPa7. Estimation of guided waves from cross-correlations of diffuse Wavefields for passive Structural Health Monitoring. Adelaide Duroux (Georgia Institute of Technology, School of Mechanical Engineering, 771 Ferst Drive, NW, Atlanta, GA 30332-0405, USA, adelaide.duroux@gatech.edu), Karim G. Sabra (Georgia Institute of Technology, School of Mechanical Engineering, 771 Ferst Drive, NW, Atlanta, GA 30332-0405, USA, karim.sabra@me.gatech.edu), Massimo

Ruzzene(Georgia Institute of Technology, School of Aerospace Engineering, 270 Ferst Drive, Atlanta, GA 30332-0150, USA, massimo .ruzzene@ae.gatech.edu), Vin Sharma (Georgia Institute of Technology, School of Aerospace Engineering, 270 Ferst Drive, Atlanta, GA 30332-0150, USA, vin.sharma@millenniumdynamics.com), James Ayers (Georgia Institute of Technology, School of Aerospace Engineering, 270 Ferst Drive, Atlanta, GA 30332-0150, USA, jayers3@gatech.edu)

Recent theoretical and experimental studies in a wide range of applications (ultrasonics, underwater acoustics, seismicœ) have demonstrated that Green's functions (impulse responses) can be extracted from cross-correlation of diffuse fields using only passive sensors. The technique, whose validity is supported by a physical argument based on time-reversal invariance, effectively uses a correlation process between the point source and points located in the focal zone. Indeed, the coherent noise source distributions can be considered as a time-reversal mirror and the cross-correlation operations gives the field measured at one receiver after refocusing on the other receiver. Passive-only reconstruction of coherent Lamb waves (80-200 kHz) in an aluminum plate and thickness comparable to aircraft fuselage and wing panels will be presented. In particular, the influence of the noise source characteristics (location, frequency spectrum) on the signal-to-noise ratio the emerging coherent waveform will be investigated using a scanning laser Doppler velocimeter. This study suggests the potential for a structural health monitoring method for aircraft panels based on passive ultrasound imaging reconstructed from diffuse fields.

Session 2aSPb

Signal Processing in Acoustics, Biomedical Ultrasound/Bioresponse to Vibration, and Underwater Acoustics: Time Reversal Methods for Array Imaging and Signal Processing I

David Chambers, Cochair
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Claire Prada, Cochair
Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, Paris, 75005, France

Invited Papers

10:40

2aSPb1. The detection and characterization of buried resonant targets by iterative, single-channel time reversal. Ronald A. Roy (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, ronroy@bu.edu), Zachary J. Waters (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, zjwaters@bu.edu), Benjamin R. Dzikowicz (Naval Surface Warfare Center, Panama City Division, Code HS-11, 100 Vernon Ave., Panama City, FL 32407, USA, benjamin.dzikowicz@navy.mil), R. Glynn Holt (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, rgholt@bu.edu)

A technique for detecting buried resonant targets is described in which enhanced signal-to-noise ratio and convergence to a narrowband signal is achieved using the iterative time reversal of backscattered echo returns. [Waters et al., J. Acoust. Soc. Am. 122, 3023 (2007).] The center frequency of the converged-upon signal is typically a resonance frequency of the target and thus could be used for classification and identification purposes. This technique could offer a straightforward means for enhancing target return levels in a noisy or cluttered environment using existing SONAR systems. The procedure consists of exciting the transducer with a broadband pulse, digitizing the echo return windowed about the target, reversing the data stream in time, and using this signal as the source waveform for the next interrogation pulse. We report results derived from a multi-pronged investigation that includes numerical modeling, high-frequency (100 kHz - 2 MHz) scaled tank experiments employing both free-field and buried targets, and mid-frequency (20 kHz -200 kHz) buried target experiments run in a shallow pond. [Work supported by The Office of Naval Research and the Center for Subsurface Sensing and Imaging Systems (NSF ERC Award No. EEC-9986821).]

11:00

2aSPb2. Green's function estimation in speckle using the FDORT method. Jean-Luc Robert (Philips Research, 345 Scarborough Road, Briarcliff Manor, NY 10510, USA, jean-luc.robert@philips.com), Mathias Fink (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, mathias.fink@espci.fr)

The FDORT method (French acronym for Decomposition Of the Time Reversal Operator using Focused beams) is a Time Reversal based method that can detect point-scatterers in a heterogeneous medium and extract their Green's function. It is particularly useful when focusing in a heterogeneous medium. In this presentation, the theory of the FDORT method is generalized to random media (speckle), and it is shown that it is possible to extract Green's functions from speckle signal using this method. Therefore it is possible to achieve a good focusing even if no point scatterers are present. Moreover, a link is made between FDORT and the Van Cittert Zernike theorem. We deduce from this interpretation that the normalized first eigenvalue of the Focused Time Reversal Operator is a well-known

focusing criterion. The concept of an equivalent virtual object is introduced, that allows the random problem to be replaced by an equivalent deterministic problem and leads to an intuitive understanding of FDORT in speckle. Applications to aberration correction are presented. The reduction of the variance of the Green's function estimate is discussed.

Contributed Papers

11:20

2aSPb3. Separation of single and multiple scatterring: Application to the ultrasonic detection of a target embedded in a diffusive medium. Alexandre Aubry (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, alexandre.aubry@espci.fr), Arnaud Derode (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, arnaud.derode@ujf-grenoble.fr)

We present a new imaging technique dedicated to the detection of a target embedded in a strongly scattering medium. Classical imaging techniques such as echography fail in this kind of configuration because of multiply scattered echoes and aberration distorsions. The experimental set up consists in an array of programmable transducers placed in front of a collection of steel rods randomly distributed. The slab thickness is three times the mean free path. Behind this strongly diffusive slab, we set an echogene steel cylinder that we want to detect and localize. The impulse responses between each couple of transducers are measured and form the interelement matrix. Our technique separates the single-scattered echoes from the multiple scattering background. This is possible because of a deterministic coherence of single-scattering signals along the antidiagonals of the array response matrix, whatever the distribution of scatterers. Once this operation is performed, the detection of the target is achieved by applying the DORT method (French acronym for decomposition of the time reversal operator). The quality of detection is assessed theoretically with Random Matrix Theory and shown to be, by far, better than what is obtained with echography and the classical DORT method.

11:40

2aSPb4. Time reversal processing to forward scattering waves of underwater targets. Yoshiaki Tsurugaya (NEC @Corp., 1-10 @Nissin-cho, Fuchu, 183-8501 Tokyo, Japan, y-tsurugaya@bp.jp.nec.com), Toshiaki Kikuchi (National Defence Academy, 39-21 Uhoyabe 4-chome, Yokosuka, 238-0024 Kanagawa, Japan, ADS01881@nifty.com), Koichi Mizutani (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, mizutani@esys.tsukuba.ac.jp)

This paper presents the detection of underwater targets using a time reversal. When a target exists between a sound source and a time reversal array in shallow water, the time reversal array receive the sound waves from the sound source and the waves scattering by the target. If the time reversal processing to them is carried out and they are re-transmitted from the time reversal array, it will be thought that they are converged at the position of the sound source and the target. However, since the waves converging at the sound source have a high level, the waves converging at the target position are usually masked by the high level sounds. Then, we cannot observe the waves converged at the target. We eliminate only the high level sounds from the sound fields. In each array element, the signals in case of non target are subtracted from the signals including the target. As a result of subtraction, the components of the scattering wave by the target are left on the array elements. The time reversal fields of the scattering wave are constructed by radiating the components of the scattered waves from each element again.

12:00

2aSPb5. Super-resolution imaging of active sound and vibrational sources using a time-reversal sink. Eric Bavu (Univ. de Sherbrooke, Mechanical Engineering Depart., 2500 Boulevard de l'Université, Sherbrooke, QC J1K 2R1, Canada, eric.bavu@usherbrooke.ca), Alain Berry (Univ. de Sherbrooke, Mechanical Engineering Depart., 2500 Boulevard de l'Université, Sherbrooke, QC J1K 2R1, Canada, alain.berry @usherbrooke.ca), Jean-Dominique Polack (Institut Jean le Rond d'Alembert, Laboratoire d'Acoustique Musicale, 11, rue de Lourmel, 75015 Paris, France, polack@ccr.jussieu.fr), Vincent Gibiat (Université Paul Sabatier, PHASE, 118, route de Narbonne, 31062 Toulouse cedex 9, France, gibiat@cict.fr), Charles Besnainou (Institut Jean le Rond d'Alembert, Laboratoire d'Acoustique Musicale, 11, rue de Lourmel, 75015 Paris, France, chbesnai@ccr.jussieu.fr)

Theory and experiments of super-resolution focusing using a time-reversal sink have been investigated in high-frequency regime [Rosny and Fink, Phys. Rev. Lett. **89**] and in audible range [Bavu, Besnainou, Gibiat, Rosny and Fink, Act. Acoust., **93**]. This technique, generalized to the case acoustic and vibrational imaging of active sources, allows super-resolution imaging and provides a new method of characterization of active sources in a known background medium. This imaging technique involves a measurement in the background medium using an array, and the simulation of the backpropagating-field in a fictive medium. An ideal numerical time-reversal sink (NumTRAS) is then used to refine results and obtain high-contrast, high-resolution imaging of initial sources. The algorithm has been validated in parallel supercomputer simulations, in both vibrational and acoustics fields and has been used to detect active vibrational sources in a clamped Mindlin plate and active sound sources in an anechoic room. All results show high-resolution imaging capabilities when compared with classical time-reversal backpropagation. NumTRAS provides an alternative to other imaging and source detection techniques, such as acoustic holography and beamforming. Beyond the applications of acoustic and vibrational nondestructive evaluation of industrial structures, NumTRAS has applications in evaluation of musical structures and is being tested to detect and characterize moving sources.

12:20

2aSPb6. Invariants of the time reversal operator for an elastic target in a water waveguide. Franck D. Philippe (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, franck.philippe@espci.fr), Claire Prada (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, claire.prada-julia@espci.fr), Julien De Rosny (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, julien.derosny@espci.fr), Jean-Gabriel Minonzio (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, jean-gabriel.minonzio@espci.fr), Mathias Fink (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, mathias.fink@espci.fr)

Detection and characterization of a target in shallow water is an active field of research. In general, in a waveguide, the dispersion is such that the frequency signature of a target cannot be extracted from a single backscatterd signal unless the waveguide properties as well as the target's position are known (Mignerey et al., JASA 1992 and Yang et al., JASA 1994). We propose to apply the Decomposition of the Time-Reversal Operator method to recover the target's signature in an unknown waveguide. Using a modal theory, we show that provided the target is far from the boundaries of the guide the first singular value of the time reversal operator is proportional to its signature. Using the same approach, the second singular value is shown to be proportional to the second derivative of the angular dependant form function which is a relevant parameter for target identification. Ultrasonic laboratory experiments are presented that confirm these theoretical results.

Session 2aUW

Underwater Acoustics and ECUA: Fifty Years of Progress in Sonar Acoustic Research: The Role of NURC/SACLANTCEN

Henrik Schmidt, Cochair
MIT, 77 Massachusetts Ave, 5-204, Cambridge, MA 02139, USA

Finn B. Jensen, Cochair
NATO Undersea Research Centre, Viale San Bartolomeo 400, La Spezia, 19126, Italy

8:40-9:00 Director's Welcome

Invited Papers

9:00

2aUW1. Saclant ASW Research Centre contributions to underwater acoustics during the first sixteen years of its existence: Personal records. Robert Laval (1 rue Emile Duclaux, 75015 Paris, France, robert-laval@wanadoo.fr)

In less than one year the NATO Undersea Research Centre, previously called Saclant ASW Research Centre, will celebrate its 50th anniversary. I had the great privilege to be one of the first scientists joining the Centre in 1959 and to work there until 1975. The main purpose of the research conducted in underwater acoustics was to reach a physical understanding of the different processes of sound propagation, including multipath effects, transformations introduced by a multilayered bottom and random scattering by the volume inhomogeneities and by the surface and bottom roughness. Experiments at sea first conducted from the research ships Aragonese and then Maria Paolina used explosives charges and active sonars FM pulses as sound sources. The digital analysis equipment, which had been designed to record, process and facilitate the interpretation of the received signals, had no equivalent at the time. The Centre has been a very active platform of intercommunications for most of the civilian and military organizations, which were involved in this discipline. The result of this continuous exchange of ideas has facilitated the harmonization of the research programs in the different NATO countries and has greatly contributed to the creation of a large international research community in underwater acoustics.

9:20

2aUW2. Shallow water environmental acoustics at NURC/SACLANTCEN. William A. Kuperman (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, wkuperman@ucsd.edu)

The Centre has a long tradition of studying shallow water acoustics. The research has been experimental and theoretical, the latter effort often involving either using or developing models. Examples include innovations such as the concept of the optimum frequency of an acoustic channel, first measured and then precisely explained with modeling, to, for a recent example, the utilization of ocean ambient noise based on its special properties. Consequently, the Centre has continued to remain at the leading-edge for a half century. The common goals at the Centre that provided the impetus for the progression of research are presented. Then a brief historical review of a selection of the contributions the Centre has made in shallow water acoustics is given.

9:40

2aUW3. Seafloor studies at NURC/SACLANTCEN: The first 50 years (1959-2008). Michael Richardson (Naval Research Laboratory, Marine Geosciences Division, Stennis Space Center, MS 39529-5004, USA, Mike.Richardson@nrlssc.navy.mil)

Characterization of the seafloor has always been one of the central research themes at NURC/SACLANTCEN. These studies, reviewed in this presentation, generally supported acoustic measurement and modeling programs. Motivation for bottom-interacting acoustic research began with the deep-water ASW programs of the 1960s and 1970s (long-range propagation, reflection, and bottom loss measurements) and slowly evolved into support for high-frequency, shallow-water MCM programs of the 1990s and 2000s (acoustic propagation within sediments, penetration into and scattering from the sediment surface). Seafloor characterization has included scales appropriate for plate tectonics to sub-mm scale sediment microstructure used to statistically characterize sediment heterogeneity for high-frequency acoustic studies. Although collection and analysis of sediment cores has been the dominate methods of seafloor characterization, direct in situ measurements and remote acoustic characterization has often been used. Seafloor studies were first rate, cutting edge research as demonstrated by quality and quantity of peer-reviewed publications by NURC/SACLANTCEN scientists and their post-SACLANTCEN research careers. The excellent engineering department made possible the development of unique acoustic and seafloor sampling equipment. Ship support for seafloor studies has always been a NURC/SACLANTCEN advantage that has attracted many scientists to multi-national, multi-institutional experiments and symposia.

10:00

2aUW4. My personal perspective of sonar research at the NATO Undersea Research Centre. Peter Wille (Kitzeberger Strasse 31, 24226 Heikendorf, Germany, p.c.wille@t-online.de)

As the first German Director of the NATO Undersea Research Centre I was contemporary to the fall of the Berlin Wall in November 1989 and the unification of Germany in October 1990, the breathtaking termination of the Cold War without bloodshed. NATO's Undersea Research Centre, thirty years of age when the Berlin Wall fell, has been a key player and focal point of a most important and demanding field of defense research. It has been exemplary in combining customer oriented applied research on sonar concepts up to system demonstrators with basic research to understand, to model and to predict the variability of the ocean environment, the most difficult challenge of successful surveillance and reconnaissance. Some unique capabilities enable the Centre to serve the nations and the NATO commands: Above all, the ability to attract first class scientists to work together at the Centre for a few years and then return to their home countries, forming an ever growing international network of intense scientific cooperation. The advanced sea trial related technology and data treatment realized by a capable and flexible technical staff is no less decisive. The many high-ranking visitors, scientific, military and political underline the international standing of the Centre.

10:20-10:40 Break

10:40

2aUW5. 19 years in the SCNR for NURC: A recollection. Leif Bjørnø (UltraTech Holding, Stendiget 19, DK-2630 Taastrup, Denmark, prof.lb@mail.dk)

I joined the SCNR for NURC by January 1st 1980 and left it again by the end of 1998. This period, on which I will concentrate my talk, includes the toughest period of the ″Cold War″ up to 1989 and the period after, when money to NURC became sparse and new ways of funding had to be found. This presentation will emphasize some of the highlights in NURC's contributions to Underwater Acoustics, ASW, MCM and to engineering aspects of underwater research. These highlights include for instance towed arrays, underwater acoustic modelling, MILOC, the MCM and the REA programmes, time-reverse underwater acoustics, operations research and engineering of underwater instruments. These important contributions over 19 years were supervised by 10 chairmen of the SCNR and by 6 directors, all putting their special fingerprint, closely related to their personality, on the course of NURC's development. A few humoristic features and experiences from the 19 years will briefly be referred to.

11:00

2aUW6. The Centre: It's Impact on Ocean Science---Past, Present and Future. David Bradley (Pennsylvania State University, Post Office Box 30, State College, PA 16804-0030, USA, dlb25@psu.edu)

The Centre (NATO Undersea Research Centre) was formed in 1959, in response to the (then) clear Soviet Union expansion of submarine operations and consequent threat to the NATO Alliance. It began as an experiment to provide a ″water leveling″ of ASW capability to the Nations of the Alliance. The combination of multi-national staff, on site for 3-5 years for the most part; together with the resident staff, has made this ″experiment″ an incredible success. The contributions to this success come in two parts: Specific work and scientific progress on site at the Centre and equally importantly, continued international collaboration upon return to one's Nation. The primary purpose of this presentation is to provide some ″data″ to illustrate both the impact of the Centre on Ocean Science and it's role within the Alliance. Given the rather specific circumstances that lead to the creation of the Centre, the question should be (and is) asked: ″Should the experiment end?″ The simple answer is: ″NO!″ The Centre is poised to have an even more critical international role in the future: The knowledge and stewardship of vital marine resources.

11:20

2aUW7. NURC/SACLANTCEN contributions to sonar signal processing and ASW systems research. Douglas Abraham (CausaSci LLC, PO Box 5892, Arlington, VA 22205, USA, abrahad@ieee.org)

The NATO Undersea Research Centre (NURC) was established in 1959 as the SACLANT ASW Research Centre and tasked with providing scientific research and technical analysis in the area of antisubmarine warfare (ASW) to NATO nations. This presentation will review NURC's contributions to sonar acoustic research in the areas of sonar signal processing and ASW systems. These contributions span at-sea testing and experimentation of novel sonar system concepts to algorithm development and theoretical advances in sonar signal processing. The diversity of expertise across fields such as oceanography, acoustics, signal processing, systems, and operations research brought by international scientists into the cauldron of a small, focused research centre with world-class technicians, engineers, and at-sea capabilities produced innovative, interdisciplinary research. [Supported by the Office of Naval Research Code 321US under contract number N0001407C0092]

11:40

2aUW8. NURC/SACLANTCEN milestone experiments toward solving inverse problems in ocean acoustics. Jean-Pierre Hermand (Université libre de Bruxelles (U.L.B.) - Environmental hydroacoustics lab, av. Franklin D. Roosevelt 50, CP 194/5, 1050 Bruxelles, Belgium, jhermand@ulb.ac.be)

This paper reviews milestone experiments conducted by Saclant Undersea Research Centre to support the development and validation of techniques for obtaining and taking into account environmental information in sonar. The experiments brought together underwater acousticians, geophysicists and oceanographers with the aim to collect comprehensive acoustic and environmental ground-truth data. Environmental-adaptive signal processing was first demonstrated in a deep water area (WESTSARDINIA'89&90). From ducted propagation measurements, a model-based matched filter (MBMF) receiver that fully incorporates the physics of wave propagation determined a source range, depth and Doppler. For environmental inversion the broadband extension of matched-field processing

was investigated in a shallow water area south of Elba island (YELLOWSHARK'94&95). Geoacoustic properties of the sea bottom were determined by finding the best fit between predicted and observed sound fields at multiple frequencies using genetic search algorithms. The same inversion results were obtained by MBMF processing of broadband linearly-frequency-modulated signals enabling the use of an array of a few hydrophones instead of a dense and large vertical array. This motivated the development of drifting acoustic buoys which were tested successfully over the southern continental shelf off Marettimo island, Sicily (ENVERSE'97&98). A recent experiment demonstrated an integrated concept of Rapid Environmental Assessment (MREA/BP'07) using sparse arrays of hydrophones and pressure/temperature sensors, hand-deployed from small vessels.

Session 2pAAa

Architectural Acoustics and ASA Committee on Standards: Comparison of US and European Standards in Building/Room Acoustics I

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Invited Papers

2:00

2pAAa1. Building acoustics standards in the USA - ASTM and ISO. Sean D. Browne (Armstrong World Industries, 2500 Columbia Avenue, Lancaster, PA 17604, USA, sdbrowne@armstrong.com), Kenneth P. Roy (Armstrong Building Products, 2500 Columbia Ave, Lancaster, PA 17603, USA, kproy@armstrong.com)

In the US most architectural design and acoustical performance specifications are based on the ASTM International standards for Construction/Building & Environmental Acoustics. Those standards are developed as a responsibility of ASTM International committee E 33 on Building & Environmental Acoustics. The US is of course a participating member in the development of ISO standards through the American National Standards Institute (ANSI), and this participation relative to standards in architectural acoustics is within ISO TC 43 SC2 which deals with building acoustics. The actual ISO standard development and voting responsibilities in the US rests with the ANSI Technical Advisory Group (TAG) to ISO TC43 SC2 which is delegated to ASTM International committee E 33.06 International Standards. This paper will present the process whereby international participation is involved in the development of both ASTM and ISO standards from the US perspective.

2:20

2pAAa2. Prediction models for building performance - European need and world wide use. Eddy Gerretsen (TNO Science and Industry, Stieljesweg 1, 2628CK Delft, Netherlands, eddy.gerretsen@tno.nl)

The development of the unified European market made it necessary to create CE-marking to indicate a certain performance of products. For building products the performance had to enable the fulfilment of essential requirements by buildings, constructed with those products. This made it essential for acoustic requirements to standardize predictions models: the link between acoustic product performance and building performance. In the mean time all six parts of that standard (EN 12354) have been published and are used. Some parts have also been published as ISO standard (ISO 15712) indicating the wider interest in the subject. To be of use in Europe and elsewhere it is important that all types of building structures are covered and indeed work is going on to extend the models to lighter building elements, even more common outside Europe than within. Furthermore, a very important aspect of prediction models is the input data, hence an increased need of standards to determine product performance in an appropriate way. This is even more an item of world wide interest. So the existing standards, the current developments for improvement and the identified need for product standards will be addressed.

2:40

2pAAa3. Comparison of ASTM and ISO sound absorption test methods. Marsha S. Bischel (Armstrong Building Products, 2500 Columbia Ave, Lancaster, PA 17603, USA, msbischel@armstrong.com), Kenneth P. Roy (Armstrong Building Products, 2500 Columbia Ave, Lancaster, PA 17603, USA, kproy@armstrong.com), Joan V. Greenslade (Armstrong Building Products, 2500 Columbia Ave, Lancaster, PA 17603, USA, jvgreenslade@armstrong.com)

A series of planned experiments has been conducted in which the differences between ASTM C423 and ISO 354 have been compared. These experiments focused on the differences in the two test methods, with the goal of identifying and understanding differences in the ultimate test results. A variety of acoustical ceiling tiles were tested in the same NVLAP-certified facility, generating data that cover a broad range of acoustical performance. The factors examined were: the effect of sample size on absorption; the effect of