Contributed Papers

12:20

5aUWi3. Radar and sonar interferometry. René Garello (ENST-Bretagne, Dpt ITI, CS 83818, 29238 Brest Cedex 03, France, rene.garello@telecom-bretagne.eu), Christophe Sintes (ENST-Bretagne, Dpt ITI, CS 83818, 29238 Brest Cedex 03, France, christophe.sintes @enst-bretagne.fr), Didier Gueriot (Telecom Bretagne, Dept iTi - Technopole Brest-Iroise, CS 83818, 29238 Brest, France, didier.gueriot @telecom-bretagne.eu), Jean-Marie Nicolas (Ecole Nationale Supérieure des Télécommunication de Paris, Telecom ParisTech, Département TSI, 46 rue Barrault, 75634 Paris Cedex 13, France, nicolas@enst.fr)

This paper is an attempt to compare two interferometric processings. The first one is applied to traditional space-borne radar (SAR) and the second on recent interferometric sonar data. Few comparisons between those tech-niques have already been made, despite the fact that they share many similar principles, only a. Thus, the key idea of this article is to present both techniques with assets, drawbacks and specific “tricks” used in dataprocessing. The first part introduces briefly both sensors and compares signal and processing techniques used for both of them. The second part deals with interferometry, and more precisely with underwater and satellite interferometry. Then a noise-pollution analysis is performed on both techniques followed by bias removal methods for getting interferometric information. The conclusion summarizes the similarities between sonar and radar processing, pointing at the techniques that can applied to both.

12:40

5aUWi4. Interferometric synthetic aperture processing: a comparison of sonar and radar. Michael Hayes (University of Canterbury, Private Bag 4800, 8022 Christchurch, New Zealand, michael.hayes @canterbury.ac.nz), Peter T. Gough (University of Canterbury, Private Bag 4800, 8022 Christchurch, New Zealand, peter.gough@canterbury.ac.nz)

Interferometric aperture synthesis is an inverse problem that attempts to form an elevation map of the earth (in the case of radar) or a bathymetric map of the seafloor (in the case of sonar). In both cases, a pair of (nominally) vertically displaced transducers is configured as an interferometer. After aperture synthesis is performed to produce a pair of images, the height of each resolvable scatterer can be estimated using time delay estimation between the image pairs and knowledge of the system geometry. While interferometric synthetic aperture sonar (InSAS) seems like an obvious extension of the methods of interferometric synthetic aperture radar (InSAR), the height estimation algorithms are surprisingly different. In this paper we start with the principle of generalised correlation for optimal time delay estimation. This filters the signals to maximise their coherence since the accuracy of the time delay estimates, and thus the height estimates, strongly depends upon the signal coherence. We then consider the fundamental differences between InSAR and InSAS; namely the relative signal bandwidth, aperture sampling rate, and geometry and show how application of generalised correlation time delay estimation leads to the differences in how InSAS and InSAR signals are processed.

FRIDAY AFTERNOON, 4 JULY 2008 — ROOM 242B, 3:20 TO 6:20 P.M.

Session 5pAAa

Architectural Acoustics and Musical Acoustics: New Measurement Parameters in Performing Arts Spaces II

Lily M. Wang, Cochair
University of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0681, USA

Brian F. Katz, Cochair
LIMSI-CNRS, B.P. 133, Orsay, 91403, France

Invited Paper

3:20

5pAAa1. Determining acoustical parameters using cochlear modeling and auditory masking. Jasper Van Dorp Schuitman (University of Technology Delft, Lorentzweg 1, 2628 CJ Delft, Netherlands, J.vanDorpSchuitman@tudelft.nl), Diemer De Vries (University of Technology Delft, Lorentzweg 1, 2628 CJ Delft, Netherlands, d.devries@tudelft.nl)

The acoustical qualities of a concert hall or any other room are generally expressed using acoustical parameters determined from impulse responses. From microphone array measurements it turned out that these parameters can fluctuate severely over small distances, whereas the perceptual cues for which these parameters are supposed to be a measure remain constant. This means that a local parameter value has a very low predictive value for acoustic quality. In this research, cochlear modeling techniques and simulations of auditory masking effects have been applied to model human hearing. These techniques together model various stages in the auditory path, like the movement of the basilar membrane inside the cochlea and mechanisms inside the brains. It turns out that determining acoustical parameters using this representation leads to results which show much less spatial fluctuations, and are closer to human perception.

Contributed Papers

3:40

5pAAa2. Sound strength and reverberation time in small concert halls. Marc Aretz (RWTH Aachen University, Institute for Technical Acoustics, Neustraße 50, 52066 Aachen, Germany, Marc.Aretz@akustik.rwth-aachen.de), Raf Orlowski (Arup Acoustics, St. Giles Hall Pound Hill, CB3 0AE Cambridge, UK, Raf.Orlowski@arup.com)

Many small concert halls are being built in music departments in schools and universities and these have to cater for a wide variety of musicalensembles ranging from orchestras to solo performers. Such diverse musical forces require different acoustic conditions in terms of reverberation and loudness and so variable acoustics are frequently provided. However, introducing absorption decreases reverberation and loudness and so a careful balance needs to be struck between controlling loudness and maintaining reverberation. In the course of this study a series of measurements was carried out in six small concert halls in Cambridge, UK, which accommodate a range of sizes of musical ensembles from quartets to orchestras, in order to determine the range of reverberation times and sound strengths, including changes due to variable absorption. The measured strength levels were compared to values derived from traditional and revised theory on strength calculations in order to assess the accuracy of the theories for small chamber music halls. The measured values of strength levels were observed to be mostly lower than the predicted ones. In order to account for this difference (particularly in spaces with added absorption) a combination of Barron's revised theory and the Vorländer correction factor is proposed.

4:00

5pAAa3. Proposition of new acoustical parameters to analyze the 3D spatial composition of sound in music spaces. Alban A. Bassuet (Arup Acoustics, 155 Avenue of the Americas, New York, NY 10013, USA, alban.bassuet@arup.com)

From acoustic measurements conducted in more than 100 renowned historical music spaces, for the Constellation Project, this paper proposes new acoustical parameters created to better describe the 3D spatial composition of sound in music spaces. Using B-format recordings, the author is proposing a visualization algorithm to plot the acoustic intensity at specific time frames and ranges. The acoustic energy is decomposed into relevant space segments and energy ratio parameters LH (lateral frontal high versus lateral frontal low), and FR (lateral rear high versus lateral rear low) are deduced and proposed for analyzing the distribution of sound and the envelopment characteristics of the room. Examples of intensity plots and of the proposed 3D acoustical parameters are given for various room types ranging from small to large concert halls, small to large opera houses, famous organ churches and Roman basilicas.

4:20-4:40 Break

Invited Paper

4:40

5pAAa4. Early reflection surfaces in Concert Halls - a new quantitative criterion. Yann Jurkiewicz (Kahle Acoustics, 188 avenue Molière, 1050 Brussels, Belgium, yjurkiewicz@kahle.be), Eckhard Kahle (Kahle Acoustics, 188 avenue Molière, 1050 Brussels, Belgium, kahle@kahle.be)

A new acoustic parameter has been defined for the acoustic brief of the Philharmonie de Paris Concert Hall. With a seating capacity of 2400 and the audience enveloping the performers on all sides, the new hall will be at the upper limit of the ideal range for symphonic music, and an efficient acoustic design was called for. In order to relate architectural design to acoustic efficiency, and based on quantitative study of existing halls, an early efficiency parameter was developed. For the Paris Philharmonie the brief requested a total area of 1400 m^2 of surfaces being able to create early reflections, with 500 m^2 being less than 15 m from the stage. The studies leading to the definition and the justificantion of the parameter will be presented. Another, more accurate definition expresses the early efficiency parameter in terms of the solid angle for a source on stage, allowing generalization of the new criterion for all hall sizes.

Contributed Paper

5:00

5pAAa5. Diffuseness and intensity analysis of spatial impulse responses. Tapio Lokki (Helsinki University of Technology, P.O. Box 5400, 02015 TKK, Finland, Tapio.Lokki@tkk.fi)

Spatial impulse responses, meaning responses measured with a microphone grid, were measured in seven concert halls. The microphone array consisted of 12 omni-microphones, enabling a construction of three intensity pairs (in x, y, and z directions) with 1 cm spacing and three intensity pairs with 10 cm spacing. In each hall impulse responses were measured with at least three loudspeaker and four microphone positions. They were analyzed with directional audio coding methodology, which enables analysis of diffuseness and instantaneous intensity as a function of time and frequency. In other words, with this analysis it is possible to analyze the directions of early reflections and to estimate diffuseness of sound field in a measurement position. Preliminary results indicate that diffuseness is quite similar in different positions in one hall, but it varies more between halls. The directions of early reflections are hard to visualize, however some example videos are shown to get an idea about the possibilities of such an analysis technique.

Invited Paper

5:20

5pAAa6. Experiments with the orchestral impulse response. Gary W. Siebein (Univ. of Florida, 231 Arch, PO Box 115702, Gainesville, FL 32611, USA, gsiebein@siebeinacoustic.com), Robert M. Lilkendey (Siebein Associates, Inc., 625 NW 60th Street, Suite C, Gainesville, FL 32607, USA, rlilkendey@siebeinacoustic.com), Hyun Paek (Siebein Associates, Inc., 625 NW 60th Street, Suite C, Gainesville, FL 32607, USA, hpaek@siebeinacoustic.com), Chris Jones (Siebein Associates, Inc., 625 NW 60th Street, Suite C, Gainesville, FL 32607, USA, cjones@siebeinacoustic.com), Joshua Fisher (Siebein Associates, Inc., 625 NW 60th Street, Suite C, Gainesville, FL 32607, USA, jfisher@siebeinacoustic.com), Youngmin Kwon (Univ. of Florida, 231 Arch, PO Box 115702, Gainesville, FL 32611, USA, ymkwon@hotmail.com)

Experiments were conducted in a large multipurpose performance hall to examine the differences in listening quality and physical acoustical measurements that resulted from using a single dodecahedral loudspeaker as the sound source and an array of multiple directional loudspeakers spread across the stage to simulate the various sections of an orchestra. Significant differences in listening qualities were recorded by listeners at three locations in the hall. The impulse responses recorded using the two systems varied dramatically in structure and in some of the typical acoustic metrics measured. However, there were also similarities among many measurements using the two systems. The research indicates the need for new measurement parameters to express the significant architectural features of the room and the physical acoustical difference that caused the perceived qualities of the sound field to vary. Preliminary analysis of the existing and proposed parameters will be presented.

Contributed Papers

5:40

5pAAa7. Relevance of acoustic parameters for musician communication. Anne Guthrie (1926 5th Ave, Troy, NY 12180, USA, guthra2@rpi.edu), Jonas Braasch (Rensselaer Polytechnic Institute, Greene Bldg., 110 8th St., Troy, NY 12180, USA)

In situations of indeterminate musical performance (particularly in telepresence, where acoustic degradation is a frequent concern), autonomous musical communication, both practical and artistic, forms the crux of the musical material. The relevance of stage acoustic and psychoacoustic parameters to contemporary performance situations must be re-examined with regards to the heightened importance of communication. Parameters developed by A. C. Gade, J. Meyer, and D. Brungart are starting points for this examination. Experiments are conducted with four instrumentalists playing excerpts from a composition by Christian Wolff (open notation allows for measurable variations depending on communication quality), communicating telematically between two virtual environments. Parameters determined by questionnaire to have the strongest effect on quality and efficiency of communication are varied at intervals and evaluated by the performers. Five parameters are tested: Self-to-Others Ratio, Commonality of Aural Space, Masking of Individual Voices, Visual-Audio Synchrony, and Position/Directivity. The performances are recorded and analyzed for variations in musical content, such as dynamics, rhythm, register, and density. The three sets of data (objective parameters, performer evaluation, and musical analysis) are compared to determine the effects of the selected parameters on musical communication.

6:00

5pAAa8. A qualitative and quantitative analysis of impulse responses from balloon bursts. Dominique J. Cheenne (Columbia College Chicago, Department of Audio Arts & Acoustics, 33 East Congress, Suite 601, Chicago, IL 60605, USA, dcheenne@colum.edu), Mauricio Ardila (Columbia College Chicago, Department of Audio Arts & Acoustics, 33 East Congress, Suite 601, Chicago, IL 60605, USA, mardila@colum.edu), Connie G. Lee (Columbia College Chicago, Department of Audio Arts & Acoustics, 33 East Congress, Suite 601, Chicago, IL 60605, USA, miss.connie.lee@gmail.com), Ben Bridgewater (Columbia College Chicago, Department of Audio Arts & Acoustics, 33 East Congress, Suite 601, Chicago, IL 60605, USA, captain.cranium@gmail.com)

Anechoic recordings of balloon bursts were systematically acquired for various conditions of balloon diameters, puncture location, and inflation pressure. The recordings were analyzed to derive the average frequency spectrum over the effective duration of the acoustic impulse. Although the data show the well-known limitations for the impulse responses (in terms of repeatability and directional behavior) when viewed at high resolution, the results are quite consistent when averaged over one-third octave bands and reveal that the diameter factor (the ratio between the diameter of the inflated balloon to that of its stated maximum), rather than the overall diameter of the balloon, is a good indicator of the sound pressure level, especially above 200 Hz. The study proposes some simple empirical formulas to predict the quantitative sound pressure level and the qualitative spectral response (using the spectral centroid and skewness) from balloon bursts, based on the inflation factor as a variable. The study also offers suggestions to maximize the value of the balloon-burst methodology in building acoustics measurements by describing an effective way to measure reverberation time while simultaneously acquiring useful directional information associated with the reflected sound.

Session 5pAAb

Architectural Acoustics: Coupled Volume Acoustics II

Jason E. Summers, Cochair
U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA

Alexis Billon, Cochair
Universite de Liege, INTELSIG group - Département E.E.I., B28 Sart-Tilman, Liege, 4000, Belgium

Invited Papers

3:40

5pAAb1. Soundfields in coupled rooms: A theoretical and phenomenological synopsis. Jason E. Summers (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, jason.summers@nrl.navy.mil)

In systems of acoustically coupled rooms, energy in the reverberant field is exchanged between constituent rooms via transondentboundaries. Energy in the direct field can be distributed between rooms by the same mechanism. These exchanges of energy have been able to explain the phenomenon of multiple-slope decay curves. Likewise, they result in spatial and spectral variations in steady-state SPL and decay-curve shape. The basic form of the decay curve is governed by the gross locations of the source(s) and receiver: which room(s) they occupy, in addition to properties of the rooms themselves: volume, surface area, and absorption. Historically, these basic dependencies have been well explained by statistical-acoustics (SA) models. More subtle variations in decay-curve shape result from the fine-scale locations of source(s) and receivers relative to one another and boundary regions through which energy is exchanged (e.g., apertures). By accounting for radiation from boundaries, and propagation delays within and between rooms, more sophisticated SA models can reproduce these effects. Even so, these models fail when SA assumptions are violated or energy transfer becomes so great (e.g., large aperture areas) that room boundaries are ambiguous. In these cases, newer computational models yield accurate predictions and physical insight. [Work supported by ONR.]

4:00

5pAAb2. On the use of diffusion equations to model the acoustics of coupled rooms. Alexis Billon (Universite de Liege, INTELSIG group - Département E.E.I., B28 Sart-Tilman, 4000 Liege, Belgium, abillon@ulg.ac.be), Vincent Valeau (Laboratoire d'Etudes Aérodynamiques (LEA), Université de Poitiers - ENSMA - CNRS, Bâtiment K, 40 Avenue du Recteur Pineau, F-86022 Poitiers, France, vincent.valeau@lea.univ-poitiers.fr), Judicaël Picaut (Lab. Central des Ponts et Chaussées, Division Entretien, Sécurité et Acoustique des Routes, Route de Bouaye - BP 4129, 44341 Bouguenais Cedex, France, Judicael.Picaut@lcpc.fr), Cédric Foy (CEBTP-SOLEN, 12 Avenue Gay Lussac, ZAC La Clef Saint Pierre, 78990 Elancourt, France, c.foy@cebtp.fr), Anas Sakout (LEPTIAB Université de La Rochelle, Avenue Michel Crépeau, 17042 La Rochelle Cedex 01, France, asakout@univ-lr.fr)

The acoustics of coupled rooms are characterized by energy exchanges through apertures and/or partition walls. The use of systems of diffusion equations allows to predict the temporal and spatial energy distributions in these configurations quite accurately. In this presentation, the diffusion formalism for room acoustics-prediction is summarized. The systems of equations to be solved in the case of coupling through an aperture and through a partition wall are presented. For two rooms coupled through an aperture (two classrooms connected through an open door), the results obtained with the diffusion model are compared to experimental data, in terms of sound pressure levels and sound decays. On the other hand, for the case of two classrooms connected through a partition wall, the diffusion model is compared to experimental data in terms of sound pressure level difference only. Finally, an engineering application is presented in the configuration involving a workroom including multiple sound sources (e.g., machines) connected to offices through open and closed doors.

4:20

5pAAb3. Modeling and analysis of acoustically coupled spaces using a diffusion equation model. Yun Jing (Rensselaer Polytechnic Institute, Greene Building, School of Architecture, 110 8th Street, Troy, NY 12180, USA, jingy@rpi.edu), Ning Xiang (Rensselaer Polytechnic Institute, Greene Building, School of Architecture, 110 8th Street, Troy, NY 12180, USA, xiangn@rpi.edu)

Acoustically coupled spaces have been studied and applied to concert halls due to a number of interesting phenomena inside the spaces, including nonexponential energy decays, which are believed to benefit both desired clarity and reverberance. A diffusion equation model has been recently applied to acoustically coupled spaces to predict both steady-state and time-dependent sound field [A. Billon, et. al., J. Acoust. Soc. Am., **120**, 2006, pp. 2043-2054], good agreements between simulations and experimental measurements have been found. In this paper, the diffusion equation along with a recently proposed modified boundary condition [Y. Jing and N. Xiang, J. Acoust. Soc. Am., **123**, 145-153 (2008)] is used, to reveal intriguing characteristics of coupled spaces, including the sound pressure level distribution along the aperture, energy flow in both rooms, and location dependence of the acoustic source on energy decay characteristics. Experimental results are employed to verify the model, and show the capability of the diffusion equation model for guiding the design of coupled spaces.

4:40

5pAAb4. Low frequency evaluation of steady-state pressure distribution and reverberation time in two-room coupled system. Miroslaw Meissner (Institute of Fundamental Technological Research PAS, Swietokrzyska 21, 00-049 Warsaw, Poland, mmeissn@ippt.gov.pl)

A modal expansion method supported by a numerical implementation has been used for studying acoustic properties of coupled room system composed of two connected rectangular enclosures. In a theoretical model a low frequency limit was considered, where modes are lightly damped, thus they were approximated by eigenmodes of a hard-walled room. Eigenfunctions and eigenfrequencies were computed numerically via an application of the forced oscillator method. Calculation results have shown a great influence of absorbing material location and sound source position on the distribution of acoustic pressure and sound decay inside enclosures. As was shown it is the result of a modal localization caused by a generation of modes with eigenfrequencies very close to frequencies of modes in rectangular prisms having the same dimensions as enclosures. When one of enclosures was much more absorbent than the other one, calculation data have demonstrated an interaction of modes during a sound decay that produces reverberant curves with a rapid initial decay and a shallow late decay slope. As was found a "sagging" appearance of decay curves occurs when a late sound decay is dominated by a decay of eigenmodes localized in an enclosure with a weak sound damping.

Contributed Paper

5:00

5pAAb5. The application of acoustically coupled spaces in concert hall design. Christopher Jaffe (167 East Rocks Road, Norwalk, CT 06851, USA, ADK117@GMAIL.COM)

At mid-twentieth century, a hall having a reverberation decay curve with a late arriving extended second slope was considered an acoustical failure. It was assumed that that the level of the extended reverberation in the hall would interfere with the ongoing running music of the ensemble and reduce orchestral clarity, definition and transparency. To my knowledge, only two practitioners pioneered the utilization of physical acoustic coupling in concert shell and concert hall design during those early years. One was Russell Johnson, when he was with Bolt Beranek and Newman and later with his own firm Artec. The other was me with both my firms Stagecraft Corporation and Jaffe Acoustics. This paper discusses how the need to improve the concert hall environment of American multi-use theatres led to the application of coupling techniques in these halls and how both Mr. Johnson and I later applied physical acoustic coupling in single room concert spaces.

Invited Paper

5:20

5pAAb6. Multivariable analysis of energy decay in coupled volume rooms: How can we objectively describe what we perceive in coupled volume performance spaces? Todd L. Brooks (Artec Consultants Inc, 114 W 26th ST FL 12, New York, NY 10001, USA, tlb@artecconsultants.com), Ted Pyper (Artec Consultants Inc, 114 W 26th ST FL 12, New York, NY 10001, USA, tap@artecconsultants.com), Kelly A. Aston (Artec Consultants Inc, 114 W 26th ST FL 12, New York, NY 10001, USA, ka@artecconsultants.com), Damian J. Doria (Artec Consultants Inc, 114 W 26th ST FL 12, New York, NY 10001, USA, dd@artecconsultants.com)

The behavior of coupled volumes in room acoustics is commonly characterized by multiple slopes observed in sound energy decay curves, as measured (or modeled) using an omnidirectional receiver. This characterization is often limiting and can not adequately explain the change in character of sound energy decay with time that we perceive as listeners in music performance spaces that employ coupled volumes. By investigating energy decay as a function of several independent variables, including time, frequency, and direction of energy arrival, we seek to forge new objective measures that relate to what a listener actually hears in a coupled volume concert hall. We will discuss relevant geometry and materials of several coupled volume performance spaces designed by Artec Consultants Inc, describe subjective phenomena we have observed in these halls, and present results of our ongoing investigation of new objective metrics geared to better characterize coupled volume music performance spaces.

Contributed Paper

5:40

5pAAb7. Theoretical considerations in the prediction of decay times for the Philharmonie de Paris. Thomas Scelo (Marshall Day Acoustics LTD, P O Box 5811, Wellesley St., 1000 Auckland, New Zealand, thomas.scelo@marshallday.co.nz), Harold Marshall (Marshall Day Acoustics LTD, P O Box 5811, Wellesley St., 1000 Auckland, New Zealand, harold.marshall@marshallday.co.nz), Joanne O. Valentine (Marshall Day Acoustics LTD, P O Box 5811, Wellesley St., 1000 Auckland, New Zealand, joanne.valentine@marshallday.co.nz)

The brief for the Philharmonie de Paris includes the requirement that the hall should combine great clarity with high reverberation. The proposed solution, which won the design competition, consists of two nested chambers: an inner space producing acoustical intimacy and an outer space with its own architectural and acoustical presence. The interaction between these two spaces gives the possibility for the full range of acoustical adaptability required in the acoustical brief. This paper reports on some theoretical modelling work for the hall where the geometry considered is first described in the context of coupled space modelling. The predicted range of the variability achieved in the design by closure of the coupling openings is then presented. Finally, the paper discusses the appropriateness of these models when predicting the decay times in such a complex geometry.

Invited Paper

6:00

5pAAb8. Adjustable acoustics --- Coupled volumes in Artec concert halls: an extravagance or necessity? Tateo Nakajima (Artec Consultants Inc, 114 W 26th ST FL 12, New York, NY 10001, USA, tn@artecconsultants.com), Damian J. Doria (Artec Consultants Inc, 114 W 26th ST FL 12, New York, NY 10001, USA, dd@artecconsultants.com), Edward Arenius (Artec Consultants Inc, 114 W 26th ST FL 12, New York, NY 10001, USA, ea@artecconsultants.com), Andrew Morgan (Artec Consultants Inc, 114 W 26th ST FL 12, New York, NY 10001, USA, ajm@artecconsultants.com)

Led by its founder, Russell Johnson, Artec has developed an unequalled body of experience in the practical application of coupled volumes in the design and construction of concert halls. This paper will present a survey of selected past and future Artec projects from the point of view of the artists and venue managers that perform and work in these halls on a daily basis. Are adjustable acoustics an extravagance or a necessity? What practical problems have been encountered and are they inherent in the concept of adjustability or can they be avoided? How have the musicians reacted? And what is the future of adjustable acoustics?

FRIDAY AFTERNOON, 4 JULY 2008 ROOM 343, 1:40 TO 4:20 P.M.

Session 5pAO

Acoustical Oceanography and ECUA: Acoustical Tomography and Long Range Propagation

Timothy Duda, Cochair
Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 107, MS-12, Woods Hole, MA 02543, USA

Yann Stephan, Cochair
SHOM, 13 rue du Chatellier, CS 92803, Brest cedex 2, 29228, France

Contributed Papers

1:40

5pAO1. Ocean acoustic tomography using a double-beamforming algorithm. Ion Iturbe (GIPSA-lab, dep. DIS, 961, rue de la Houille Blanche, 38402 St Martin d'Hères, France, ion.iturbe@gipsa-lab.inpg.fr), Philippe Roux (LGIT - CNRS - Université Joseph Fourier, Maison des Géosciences, 1381 rue de la Piscine, BP 53, 38041 Grenoble, France, philippe.roux@obs.ujf-grenoble.fr), Barbara Nicolas (GIPSA-lab, dep. DIS, 961, rue de la Houille Blanche, 38402 St Martin d'Hères, France, barbara.nicolas@gipsa-lab.inpg.fr), Jérôme I. Mars (GIPSA-lab, dep. DIS, 961, rue de la Houille Blanche, 38402 St Martin d'Hères, France, jerome.mars@gipsa-lab.inpg.fr)

Since Munk and Wunsch proposed the basis for ocean acoustic tomography, many experiments have been performed to estimate sound speed fluctuations in the ocean, using ray identification and measurement of their travel times. However, technical limitations appeared such as the precision of the arrival time measurements or the number of ray arrivals that can be extracted from the signal. Recently, technical improvements allowed more complete experiments using two vertical arrays of sensors (source array and hydrophone array). In this configuration, the signals between each source and receiver are recorded which greatly improve the available information to identify the acoustic rays. One way to increase the number of rays in the tomography algorithm is to perform double-beamforming on the source and receive arrays. With double-beamforming, ray arrivals are separated by emission angle, reception angle and arrival time. Thus, we solve more ray arrivals than with a single beamforming or with a point-to-point approach. In order to avoid previous limitations and to explore acoustical limitations, we study two simple cases through the double-beamforming algorithm: with simulated data and with ultrasonic small-scale experimental data.

2:00

5pAO2. A simulation study of shallow water tomography for coastal monitoring. Olivier Carrière (Université libre de Bruxelles (U.L.B.) - Environmental hydroacoustics lab, av. Franklin D. Roosevelt 50, CP 194/5, 1050 Bruxelles, Belgium, ocarrier@ulb.ac.be), 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), Yann Stephan (SHOM, 13 rue du Chatellier, CS 92803, 29228 Brest cedex 2, France, yann.stephan@shom.fr)

Developing operational oceanographic models for coastal environment is an exciting challenge for the next decades. The typical sparsity of assimilated *in-situ* observations often creates biases in the model predictions reducing the overall accuracy of the forecasting. In such a highly dynamic environment, acoustic tomography can be a good candidate to provide synoptic measurements over wide areas while a range-dependent inversion scheme allows to achieve a reasonable spatial resolution. In this work, we present simulation results of a Kalman-based assimilation of ocean-acoustic data for a basic model of the Ushant front west off Britanny. In a first part, a single vertical slice tomography experiment is simulated for a static front model to study in which way the modal propagation of a multifrequency acoustic signal is affected by the characteristics of the front (position, intensity). In a second part, the problem of assimilating full-field acoustic data into a dynamic model and tracking of the range-dependent sound-speed field is addressed.

2:20

5pAO3. Sound-speed estimation from RAFOS transmissions. Emmanuel Skarsoulis (FORTH / IACM, N. Plastira 100, Vasilika Voutes, GR-70013 Heraklion, Greece, eskars@iacm.forth.gr), George Piperakis (FORTH / IACM, N. Plastira 100, Vasilika Voutes, GR-70013 Heraklion, Greece, piperak@iacm.forth.gr)

Acoustic navigation of Lagrangian (moving) floats is carried out by measuring travel times from a number of fixed stations/moorings. A minimum of two fixed stations are needed for location estimation in the horizontal, whereas an additional fixed station is commonly used to remove left-right ambiguity. The signals (RAFOS) and sampling schemes used in ocean acoustic navigation are characterized by limited time resolution (order of 200 msec), much smaller than the resolution used in travel-time tomography (order of 1-10 msec). The possibility of combining navigation signals (travel

times) from three fixed stations to multiple moving floats for simultaneous sound-speed estimation and float localization is examined here. The redundant travel-time information in this case offers a significant advantage and makes the accurate estimation of the speed of sound feasible. It is shown that the estimation error for the sound speed decreases with the number of floats, and thus the estimation accuracy improves as the set of floats grows larger. This procedure also leads to improved location estimates for the individual floats. A number of numerical experiments are used to demonstrate the performance of the method. [Work supported by EU/FP6 Damocles project.]

2:40

5pAO4. Using Seagliders for acoustic receiving and communication. Bruce M. Howe (Applied Physics Lab., Univ. of Washington, 1013 Northeast 40th St., Seattle, WA 98105, USA, howe@apl.washington.edu), Michael L. Boyd (Applied Physics Lab., Univ. of Washington, 1013 Northeast 40th St., Seattle, WA 98105, USA, mike@apl.washington.edu)

Underwater gliders are beginning to be used as tools in ocean acoustics and acoustical oceanography. Results from several experiments conducted in summer 2006 with Seagliders equipped with acoustic modems and receivers are described. Off Kauai, a glider received signals from the Acoustic Thermometry of Ocean Climate/North Pacific Acoustic Laboratory 75 Hz source; subsequent coherent processing showed close to theoretical gain for 12 min records while moving away from the source at ranges >100 km with velocity 20 cm/s (measured by travel time, Doppler, and dead reckoning). In the Monterey Bay MB06 experiment, two-way communications between other subsea platforms and shore via the acoustic modem-equipped glider was demonstrated (albeit with latency). The results support the future use of gliders as precision navigated platforms, communication and time distribution nodes, and thermometry/tomography mobile receivers. Work supported by the Office of Naval Research.

3:00

5pAO5. A decade of acoustic thermometry in the North Pacific Ocean: Using long-range acoustic travel times to test gyre-scale temperature variability derived from other observations and ocean models. Peter Worcester (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, pworcester@ucsd.edu), Brian D. Dushaw (Applied Physics Lab., Univ. of Washington, 1013 Northeast 40th St., Seattle, WA 98105, USA, dushaw@apl.washington.edu), Rex K. Andrew (Applied Physics Lab., Univ. of Washington, 1013 Northeast 40th St., Seattle, WA 98105, USA, randrew@apl.washington.edu), Bruce M. Howe (Applied Physics Lab., Univ. of Washington, 1013 Northeast 40th St., Seattle, WA 98105, USA, howe@apl.washington.edu), James A. Mercer (Applied Physics Lab., Univ. of Washington, 1013 Northeast 40th St., Seattle, WA 98105, USA, mercer@apl.washington.edu), Robert C. Spindel (Applied Physics Lab., Univ. of Washington, 1013 Northeast 40th St., Seattle, WA 98105, USA, spindel@apl.washington.edu), Bruce Cornuelle (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, bdc@ucsd.edu), Matthew Dzieciuch (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, mad@ucsd.edu), Theodore G. Birdsall (Univ. of Michigan, Electrical Engineering and Computer Science Dept., 1301 Beal Ave., Ann Arbor, MI 48109-2122, USA, birdsall@umich.edu), Kurt Metzger (Univ. of Michigan, Electrical Engineering and Computer Science Dept., 1301 Beal Ave., Ann Arbor, MI 48109-2122, USA, metzger@umich.edu), Dimitris Menemenlis (Jet Propulsion Laboratory, California Institute of Technology, 4800 Oak Grove Dr., Pasadena, CA 91109, USA, menemenlis@jpl.nasa.gov)

Large-scale, range- and depth-averaged temperatures in the North Pacific Ocean were measured by long-range acoustic transmissions over the decade 1996-2006. Acoustic sources off central California and north of Kauai transmitted to receivers throughout the North Pacific. Even though acoustic travel times are spatially integrating, suppressing mesoscale variability and providing a precise measure of large-scale temperature, the travel times sometimes vary significantly on time scales of only a few weeks. The interannual variability is large, with no consistent warming or cooling trends. Comparison of the measured travel times with travel times derived from (i) the World Ocean Atlas 2005 (WOA05), (ii) an upper ocean temperature estimate derived from satellite altimetry and in situ profiles, (iii) an analysis provided by the Estimating the Circulation and Climate of the Ocean (ECCO) project, and (iv) simulation results from a high-resolution configuration of the Parallel Ocean Program (POP) show similarities, but also reveal substantial differences. The differences suggest that the data can provide significant additional constraints for numerical ocean simulations. The acoustic data show that WOA05 is a much better estimate of the time-mean hydrography than either the ECCO or POP estimates and provide significantly better time resolution for large-scale ocean variability than can be derived from satellite altimetry and in situ profiles.

3:20

5pAO6. Coherence of tracked arrivals in SPICEX. Matthew Dzieciuch (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, mad@ucsd.edu), Bruce Cornuelle (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, bdc@ucsd.edu), Peter Worcester (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, pworcester@ucsd.edu)

In the fall of 2004, 250 Hz broadband signals were received at 500 km and 1000 km ranges on a near full water-column vertical array in the North Pacific ocean. Individual ray arrivals of very high SNR could easily be identified and tracked using a turning-point filter (time-delay beamforming accounting for channel structure), thus providing accurate vertical coherence estimates. The observations can be compared to standard Monte Carlo estimates of coherence made using accurate parabolic-equation acoustic propagation in an ensemble of ocean states consistent with the standard Garrett-Munk ocean internal-wave spectrum. Acoustic coherence can also be expressed as a depth- dependent structure function. This is naturally estimated by the full-wave travel-time sensitivity kernel (TSK) which provides a linearized transformation from the internal wave spectrum to the structure function. Environmental measurements were conducted almost concurrently with the acoustic trials, constraining the acceptable ocean variability. Allowances must be made for scattering by sound-speed variability along isopycnals (spiciness) in the upper mixed layer. The most important conclusion to date is that the vertical coherence is depth-dependent but this statement must be qualified by the ability of the beamformer to separate arrivals.

3:40

5pAO7. Travel-time sensitivity kernels in long-range propagation. Emmanuel Skarsoulis (FORTH/IACM, N. Plastira 100, Vasilika Voutes, GR-70013 Heraklion, Greece, eskars@iacm.forth.gr), Bruce Cornuelle (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, bdc@ucsd.edu), Matthew Dzieciuch (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, mad@ucsd.edu)

The effect of increasing range on the sensitivity of finite-frequency travel-time observables to sound-speed perturbations is studied using the notion of wave-theoretic sensitivity kernels based on the first Born approximation of the Green's function and the notion of peak arrivals. Travel-time sensitivity kernels are examined in a range-independent background ocean environment. While at medium ranges the kernels exhibit zero-sensitivity cores on the eigenrays, these cores shrink and disappear at long ranges due to refraction and the kernels converge in the vertical towards the corresponding eigenrays; this behavior is not observed in free space. On the other hand the kernels expand in the horizontal cross-range direction and attain their maximum extent at the midpoint between source and receiver, similar to the behavior of the Fresnel volume in free space. Thus, stratification affects the shape of the sensitivity kernel in the vertical preventing expansion with increasing range but not in the horizontal such that the sensitivity kernel takes the form of a folded rug expanding the eigenray in the horizontal cross-range direction. [Work supported by ONR.]

4:00

5pAO8. High-frequency broadband acoustic current tomography in shallow water. Jing Luo (University of Delaware, College of Marine and Earth Studies, S. College Street, Newark, DE 19716, USA, luojing@udel.edu), Entin A. Karjadi (College of Marine and Earth Studies, University of Delaware, Newark, DE 19716, USA, karjadi@udel.edu), Mohsen Badiey (University of Delaware, College of Marine and Earth Studies, S. College Street, Newark, DE 19716, USA, badiey@udel.edu)

To study current tomography in very shallow water regions a simultaneous oceanographic and broadband (1-25 kHz) acoustic experiment was conducted in the Delaware Bay. The mean water depth was 15 m and the source-receiver range was 760 m. In this paper, we discuss the feasibility of using reciprocal acoustic transmission for current tomography applications. A beamforming technique is used to resolve the arrival time of direct and surface-bounced rays since in shallow water the received acoustic signals are more complicated due to multiple interactions with bottom and sea surface. Using the experimental data, the accuracy of travel time measurements for variable environmental conditions is examined for different center frequencies and bandwidths. The current velocity prediction results are compared with ADCP measurements to determine the feasibility of current tomography in shallow water.

FRIDAY AFTERNOON, 4 JULY 2008 ROOM 353, 2:00 TO 5:20 P.M.

Session 5pBB

Biomedical Ultrasound/Bioresponse to Vibration: General Topics in Biomedical Ultrasound/Bioresponse to Vibration II

Michael R. Bailey, Cochair
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Oleg A. Sapozhnikov, Cochair
Center for Industrial and Medical Ultrasound, Applied Physics Lab., University of Washington, 1013 NE 40th St., Seattle, WA 98105, USA

Contributed Papers

2:00

5pBB1. Dual confocal ultrasound system for shear wave elastography. Michael D. Gray (Georgia Institute of Technology, Mechanical Engineering, 771 Ferst Drive, Atlanta, GA 30332-0405, USA, michael.gray@me.gatech.edu), James S. Martin (Georgia Institute of Technology, Mechanical Engineering, 771 Ferst Drive, Atlanta, GA 30332-0405, USA, james.martin@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)

A dual confocal transducer system for ultrasound-based elastography is presented. The system is intended to noninvasively measure the complex shear speed in cetacean head tissues, including brain, jaw fat, and melon. The system instrumentation features a pair of dual-element confocal ultrasound transducers, one of which is used to remotely generate low frequency (100-1000 Hz) shear waves in soft tissues via radiation force, and the other is used to measure the resulting shear wave displacements using Doppler techniques. One transducer is configured as an open ring into which the other transducer can be placed and translated. The relative positions of the transducers are mechanically manipulated in order to measure short-path propagation and estimate shear wave speed and loss. Work supported by ONR.

2:20

5pBB2. Simulated response of the human lung to low-frequency underwater sound using a finite-element-based thoracic model. Mark S. Wochner (Applied Research Laboratories, The University of Texas, P.O. Box 8029, Austin, TX 78713-8029, USA, mwochner@mail.utexas.edu), Yurii A. Ilinskii (Applied Research Laboratories, The University of Texas, P.O. Box 8029, Austin, TX 78713-8029, USA, yura@arlut.utexas.edu), Mark F. Hamilton (Applied Research Laboratories, The University of Texas, P.O. Box 8029, Austin, TX 78713-8029, USA, hamilton@mail.utexas.edu), Evgenia A. Zabolotskaya (Applied Research Laboratories, The University of Texas, P.O. Box 8029, Austin, TX 78713-8029, USA, zhenia@arlut.utexas.edu)

In a previous paper, inhomogeneity within the lungs and its influence on the lung's response to low-frequency underwater sound using a finite-element-based model was discussed [Wochner, *et al*., J. Acoust. Soc. Am. **122**, 2957 (2007)]. Here we report an extension of the previous work that adds surrounding organs to the finite element model. It is hypothesized that the significant damage that can occur at relatively low amplitudes when the lung is in resonance is due primarily to the shear stresses that can occur in the lung through its interactions with surrounding organs. In particular, the heart, diaphragm, and ribs, all of which have considerably different material properties compared to lung, may have a sizable effect on the lung's response to low-frequency underwater sound. This paper will discuss the resonance frequency, motion, and stress and strain fields produced using this new finite-element-based thoracic model. [Work supported by ONR and ARL:UT IR&D.]

2:40

5pBB3. Dual apodization technique for improved contrast. Jesse Yen (University of Southern California, 1042 Downey Way, Los Angeles, CA 90089, USA, jesseyen@usc.edu), Chi Seo (University of Southern California, 1042 Downey Way, Los Angeles, CA 90089, USA, chiseo@usc.edu)

We propose a method to use dual apertures or dual apodization functions to reduce side lobes and clutter for ultrasound imaging. Using a common transmit aperture but different receive apodizations or apertures, we create two point spread functions with very similar main lobes and different side-lobe and clutter signals. Main lobe signals can be distinguished from clutter signals using normalized cross-correlation of the raw radio-frequency data. The normalized cross-correlation coefficient is used as a pixel-by-pixel weighting to pass main-lobe signals and suppress side lobe and clutter

signals. Main lobe signals will have a high cross-correlation coefficient near 1 and clutter signals will have coefficients between 0 and -1. Point target simulations show a narrowing of the main beam compared to conventional beamforming at beamwidths -20 dB and lower. Using a 5 MHz 128-element linear array, improvements of contrast-to-noise ratio (CNR) of an anechoic cyst compared to delay and sum beamforming exceed 130% in both simulations and experiments. We also evaluated this method for robustness in the presence of phase aberration. Aberrators ranging from 25-45 ns rms with correlation lengths of 3-5 mm were used. Here, improvements in CNR also exceed 100% in many cases.

3:00

5pBB4. Nonlinear ultrasonic imagery of high contrast objects. Régine Guillermin (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, guillermin@lma.cnrs-mrs.fr), Philippe Lasaygues (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, lasaygues@lma.cnrs-mrs.fr)

This study is concerned whith the ultrasonic imagery of elastic materials like cylinders or tubes by diffraction tomography technic. The aim of this work is then to solve a nonlinear inverse scattering problem. Various methods can be employed, generally involving a minimization of the differences between modeling data and measurements. The Distorted Born Iterative (DBI) method belongs to the class of algebraic reconstruction algorithms and have therefore been investigated in literature. Very promising results have been obtained both on synthetic and experimental data especially for electromagnetic inverse diffraction problems, but as far as the authors know few ultrasonic experimental results are available. This method was developed to increase the domain of application of the Born approximation to high contrast targets. Iterations are performed numerically solving a forward and an inverse problems at every iteration. This yields quantitative information about the scatterer, such as the speed of sound. Inversions of both numerical and experimental data are presented.

3:20

5pBB5. Inverse scattering in modern ultrasound imaging. Francesco Simonetti (Imperial College London, Department of Mechanical Engineering, South Kensington Campus, SW7 2AZ London, UK, f.simonetti@imperial.ac.uk)

Progress in solid state electronics and sensor manufacturing has led to the rapid development of ultrasound arrays over the last decade resulting in prototypes with thousands of transreceivers. Ultrasound scanners that use this technology are widely used in medical imaging and are based on beamforming techniques. In a similar fashion to an optical lens, the array forms an aperture which can focus and steer an ultrasound beam in space as it is done by microscopes and telescopes. The beamforming process can be seen as an inverse scattering problem whereby the scattering measurements are used to reconstruct the structure of the object being probed. To achieve this, a model that describes the interaction of the probing wave with the object is required. Beamforming assumes that scattering events occurring at different locations within the object are independent of each other, thus neglecting multiple scattering. Here, it is argue that accounting for more accurate wave-matter interaction models in the inverse scattering problem leads to greater image quality than that obtained with conventional beamforming. Experimental images with unprecedented resolution beyond the classical diffraction limit are presented along with tomographic reconstructions of a complex 3D breast phantom that show striking similarities with x-ray CT.

3:40

5pBB6. Deconvolution of freehand 3D ultrasound data using improved reconstruction techniques in consideration of ultrasound point spread functions. Holger J. Hewener (Fraunhofer IBMT, Ensheimer Strasse 48, 66386 Sankt Ingbert, Germany, holger.hewener@ibmt.fraunhofer.de), Robert M. Lemor (Fraunhofer IBMT, Ensheimer Strasse 48, 66386 Sankt Ingbert, Germany, robert.lemor@ibmt.fhg.de)

Medical ultrasound data suffers from blur caused by the volume expansion of the pressure field of the mechanical wave. This blur is dependent on the used excitation pulse and focusing of the ultrasonic wave and can therefore be examined. In order to improve the overall system resolution for 3D ultrasound reconstructions we have to know this signal degeneration to compensate it using deconvolution techniques or multicode compounding during the volume reconstruction step. Looking at the ultrasound transfer function we can focus on the simulation and measurement of the "point spread function" especially in the lateral and elevational direction. To understand its effects on a 3D reconstruction we compute a simulation of freehand-ultrasound slices based on synthetic phantom structures and given US parameters. Computing a 3D reconstruction of these simulated slices we are able to optimize the reconstruction algorithm itself to archive better resolution in the volume data sets considering ultrasound parameters like beamforming and the excitation pulses.

4:00-4:20 Break

4:20

5pBB7. Noninvasive monitoring of mesenchymal stem cells by 1.2 GHz acoustic microscopy. Moritz Von Buttlar (Institute of Experimental Physics II, University of Leipzig, Linnéstr. 5, 04103 Leipzig, Germany, vbuttlar@physik.uni-leipzig.de), Evgeny Twerdowski (Institute of Experimental Physics II, University of Leipzig, Linnéstr. 5, 04103 Leipzig, Germany, twerdowski@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)

Cell-based therapies can benefit from noninvasive and marker-free monitoring techniques for living cells. For this purpose a phase-sensitive scanning acoustic microscope operating at a frequency of 1.2 GHz was combined with a commercial confocal laser scanning microscope. The system is equipped with a live-support system for the long-term observation of living cells. Confocal acoustic imaging with phase and magnitude contrast and confocal laser scanning microscopy can be performed simultaneously. Both techniques are used in reflection from opposing sides of the object. Time-lapsed acoustic microscope images of ovine mesenchymal stem cells are presented. For this purpose, a pseudo-3D representation is generated by encoding the unwrapped phase in the height and the magnitude in the brightness. In the case of highly reflective substrates and sufficiently low reflection from the interface between the cells and the surrounding fluid the echo from the top of the cells can be neglected and the phase contrast image can be transformed to a time-of-flight image. In the same approximation the magnitude image provides information about the gradual extinction of the echo signal due to absorption in the cells. The two images can be combined to generate a new form of contrast representing the product of the absorption coefficient and the velocity of sound inside the observed cells.

4:40

5pBB8. Transverse vibration of prestressed beams: An experimental technique for the determination of dynamic viscoelastic material properties of tissue mimicking materials. Yigit Yazicioglu (Middle East Technical University, Orta Dogu Teknik Universitesi, Makine Muhendisligi Bolumu B-313, 06531 Ankara, Turkey, yigit@metu.edu.tr), Bryn A. Martin (University of Illinois at Chicago, 2923 W. 71st Street, Woodridge, IL 60517, USA, flux@ebryn.com), Karen Navarro-Castillo (University of Illinois at Chicago, 842 W. Taylor St. ERF 1072, Chicago, IL 60607, USA, knc_001@yahoo.com.mx), Umit Kutluay (Tubitak-Sage, Samsun Yolu 25 .Km, Tilkicak Tepe Mevkii P.K.16, 06261 Ankara, Turkey, ukutluay@sage.tubitak.gov.tr), Thomas J. Royston (University of Illinois at Chicago, 842 W. Taylor St. ERF 1072, Chicago, IL 60607, USA, troyston@uic.edu)

An experimental dynamic material property identification technique is presented that is based on the theoretical formulations of a vibrating prestressed beam. The technique determines the viscoelastic material properties of tissue mimicking materials that govern their dynamic behavior. Results are presented for silicone-based materials (Sylgard 184, Dow Corning, Midland, MI) that are formed in the lab using a range of mixing ratios and cast in the form of a thick string held between fixed supports under tensile prestress. The specimens are excited through the transverse harmonic displacement of the boundary. Transverse vibration at an arbitrary location is measured and compared with theory to identify material viscoelastic moduli valid up to at least several hundred Hertz. The presented technique can aid in providing accurate viscoelastic parameter values for phantoms that are used in the development of a range of medical diagnostic techniques that attempt to identify pathology or tissue differentiation via changes in mechanical stiffness and viscosity.

5:00

5pBB9. Air-borne and tissue-borne sensitivity of skin-radiation acoustic sensors. Matias Zanartu (School of Electrical and Computer Engineering, Purdue University, 206 South Martin Jischke Drive, West Lafayette, IN 47907, USA, mzanartu@purdue.edu), Julio C. Ho (Weldon School of Biomedical Engineering, Purdue University, 206 South Martin Jischke Drive, West Lafayette, IN 47907, USA, hoj@purdue.edu), Steve Kraman (Department of Internal Medicine, University of Kentucky, 740 South Limestone Street, Lexington, KY 40536, USA, sskram01@email.uky.edu), Hans Pasterkamp (Department of Pediatrics and Child Health, University of Manitoba, CS516-840 Sherbrook St, Winnipeg, MB R3A 1S1, Canada, pasterkamp@umanitoba.ca), Jessica E. Huber (Department of Speech, Language, and Hearing Sciences, Purdue University, 500 Oval Drive, West Lafayette, IN 47907, USA, jhuber@purdue.edu), George R. Wodicka (Weldon School of Biomedical Engineering, Purdue University, 206 South Martin Jischke Drive, West Lafayette, IN 47907, USA, wodicka@purdue.edu)

Measurements of body sounds on the skin surface have been widely used in the medical field and continue to be a topic of current research, ranging from the diagnosis of the respiratory and cardiovascular diseases to the monitoring of voice dosimetry. These measurements are normally acquired by means of light-weight accelerometers and/or air-coupled microphones attached to the skin. Such recordings can be corrupted by air-borne sounds that are generated by the subject or by other sources of background noise. In this project, the sensitivity of various commonly used bioacoustic sensors to airborne sounds was evaluated and compared with their sensitivity to tissue-borne body sounds. To delineate the sensitivity to each pathway, the sensors were first tested in-vitro, and then on human subjects. The results indicated that in many cases the air-borne sensitivity is sufficiently high to significantly corrupt body sound signals. The effectiveness of different air-borne insulation devices was also evaluated. Spectral analysis showed that simple acoustic insulators (e.g., passive hearing protectors) provide significant attenuation within the range of frequencies of interest for most applications, particularly when using air-coupled microphones.

FRIDAY AFTERNOON, 4 JULY 2008 — ROOM 252A, 2:00 TO 6:20 P.M.

Session 5pNSa

Noise, ASA Committee on Standards, Architectural Acoustics, and EURONOISE: Classroom Acoustics II

Louis C. Sutherland, Cochair
Consultant in Acoustics, 27803 Longhill Dr., Rancho Palos Verdes, CA 90275-3908, USA

Luigi Maffei, Cochair
Built Environment Control Laboratory Ri.A.S., Second University of Naples, Abazia di S. Lorenzo, Aversa, 81031, Italy

David Lubman, Cochair
DL Acoustics, 14301 Middletown Lane, Westminster, CA 92683, USA

Invited Paper

2:00

5pNSa1. Are classrooms in historical buildings compatible with good acoustics standards? Luigi Maffei (Built Environment Control Laboratory Ri.A.S., Second University of Naples, Abazia di S. Lorenzo, 81031 Aversa, Italy, luigi.maffei@unina2.it), Gino Iannace (Built Environment Control Laboratory Ri.A.S., Second University of Naples, Abazia di S. Lorenzo, 81031 Aversa, Italy, gino.iannace@unina2.it), Massimiliano Masullo (Built Environment Control Laboratory Ri.A.S., Second University of Naples, Abazia di S. Lorenzo, 81031 Aversa, Italy, ing.masullo@libero.it)

Many surveys and researches have underlined that the acoustic characteristics of classrooms are strictly connected to performances of students and to the stress of the teachers during lessons. In standard classrooms sound quality can be easily reached without sound amplification but introducing appropriate sound absorbing/scattering materials at the ceiling and/or at the vertical walls. Nevertheless in historical buildings with vaults or trusses, high walls and many architectural restrictions imposed by district superintendent, it could be very difficult to achieve good acoustics standards with widespread solutions. In this paper the acoustic performances of different classrooms in an historical Monastery actually center of the Faculty of Architecture of the Second University of Naples are analysed. After these analysis, compatible architectural and acoustic solutions to improve the sound quality were developed and tested in laboratory measurements and then applied in some classrooms to verify the benefits.

Contributed Paper

5pNSa2. Results of acoustical treatments in existing classrooms. Arianna Astolfi (Politecnico di Torino, Department of Energetics, Corso Duca degli Abruzzi, 24, 10129 Turin, Italy, arianna.astolfi@polito.it), Alessia Griginis (Onleco srl, Via Pigafetta, 3, 10129 Turin, Italy, griginis@onleco.com)

In 2001, 2002, and 2005, the Department of Energetics of the Politecnico di Torino has carried out in-field objective and subjective surveys with the aim of evaluating the acoustical quality in secondary-school classrooms of the Province of Turin (Italy). From the results the following main problems emerged: high reverberation times, high background noise levels caused mainly by low sound insulation between classrooms and corridors and between adjacent classrooms, low façade sound insulation, excessive teachers' vocal efforts. In this work results of acoustical treatments in some of these existing classrooms are reported. Measurements are made before and after the restorations. The case studies are chosen with reference to different building typologies and urban contexts, and consisted of improvements in sound absorption, sound insulation of internal partitions, doors and façades, and acoustic bridges elimination. They are divided in "light" and "heavy" types. The first, less expensive, with the aim to obtain good acoustical conditions, the latter, more expansive, focused to obtain optimal conditions. In fact, mainly in restoration of public schools, a good level of acoustical quality with contained costs is requested. The treatments are carried out with the aim to constitute a repertory of solutions to apply primarily in school restoration.

Invited Papers

2:40

5pNSa3. Speech perception in classroom noise and reverberation by children with typical and impaired hearing - Preliminary results. Frank Iglehart (Clarke School for the Deaf, 47 Round Hill Road, Northampton, MA 01060, USA, figlehart@clarkeschool.org)

A child's ability to perceive speech in the classroom influences academic progress. In this study, students with a various degrees of hearing loss perceive spoken words in sentences in a classroom. The room contains multiple noise levels and reverberation times (RTs). Classroom reverberation is measured using standardized procedures (with slight modifications to ASTM C423-02a: X2). RTs are controlled by quantities of acoustic panels in the room. Using the BKB-SIN Test, the speech-to-noise ratio at which the students perceive 50% words correctly (i.e., SNR-50) are measured in three reverberant conditions. Data collected to date indicate that students with severe-to-profound hearing loss ($n=15$; ages 8-16 years) demonstrated average SNR-50s of $+12$ dB ($SD=4$), $+13$ dB ($SD=4$), and $+17$ dB ($SD=4$) for conditions of 0.3, 0.6 and 0.9 s RT, respectively. Students with typical hearing ($n=14$; ages 8-16 years) had SNR-50s that averaged -4 dB ($SD=2$), -3 dB ($SD=2$) and -2 dB ($SD=2$), respectively. Performance/intensity curves are also calculated in order to estimate minimal optimal listening conditions for each RT. This is an ongoing study. The session presentation will include emerging data on children with other degrees of hearing loss.

3:00

5pNSa4. Vocal symptoms in preschool teachers and the working environment. Valdis Inigbjörg Jonsdottir (Tad er Malid, Furuvellir 13, 601 Akureyri, Iceland, valdisj@ismennt.is)

Research has shown that teachers experience a high risk of developing voice problems. Noise levels in classrooms with young children are higher than in classrooms with older students, indicating preschool teachers' voices may be at greater risk. Questionnaires were sent to 88 preschool teachers in five preschools in Akureyri, and the parents of the 424 children in daycare. The study was aimed to obtain information about (1) teachers: experienced vocal symptoms, occurrence of vocal symptoms, opinions on working environment; air quality, noise, heat, acoustics. (2) parents: children's hearing problems. The study also aimed to ascertain noise levels and reverberation time in pre-school classrooms, and to compare teachers' reported ages, teaching experience, vocal symptoms and frequency with findings in other studies. Sound and reverberation-time measurements were taken by the Icelandic Department of Occupational Health and Safety. Reported vocal symptoms were more common among preschool teachers than other teachers, even though their youth and lack of teaching experience was marked. They appeared to be workrelated. 40% of the children had a history of hearing problems. Noise levels measured were very high. Correlation was found between voice fatigue and high temperature, bad indoor air, and poor acoustics.

3:20

5pNSa5. Acoustical requirements of classrooms and new concepts of teaching. Kurt Eggenschwiler (Empa, Laboratory of Acoustics, Ueberlandstrasse 129, CH-8600 Duebendorf, Switzerland, kurt.eggenschwiler@empa.ch), Markus Cslovejcsek (School of Teacher Education at the University of Applied Sciences FHNW, Kuettigerstrasse 42, CH-5000 Aarau, Switzerland, markus.cslovjecsek@fhnw.ch)

In recent years acoustic requirements for classrooms were published in various guidelines. Of course it is important that the present knowledge will be implemented, but how should this be done? And what is the correspondence of acoustic requirements and new concepts of learning? These questions are the basis of a cooperation of the Laboratory of Acoustics at Empa, Dübendorf and the School of Teacher Education, Aarau in Switzerland. The cooperation itself is one possibility to transfer knowledge. In addition it was planned but not yet realized to develop concepts of knowledge transfer by student work (diploma thesis). Classroom acoustics were discussed in seminars and there were completed some diploma theses. One project dealt with the acoustic problems of a new teaching concept. Part of the concept is that teaching of four groups of pupils takes place in three classrooms and in the corridor. The doors of the classrooms remain open and at the same time pupils work individually, in groups, and a small group is instructed by a teacher. There were found suggestions for improving the acoustics, but for different reason the realization was not possible.

3:40

5pNSa6. An Ambiophonics system for the study of intelligibility in the virtual classrooms. Nicola Prodi (Engineering Dept. - Univ. of Ferrara, Via Saragat, 1, 44100 Ferrara, Italy, nicola.prodi@unife.it), Andrea Farnetani (Engineering Dept. - Univ. of Ferrara, Via Saragat, 1, 44100 Ferrara, Italy, andrea.farnetani@unife.it), Patrizio Fausti (Engineering Dept. - Univ. of Ferrara, Via Saragat, 1, 44100 Ferrara, Italy, patrizio.fausti@unife.it), Roberto Pompoli (Engineering Dept. - Univ. of Ferrara, Via Saragat, 1, 44100 Ferrara, Italy, roberto.pompoli@unife.it)

In recent years the investigations on the intelligibility has profited by the increasing efficiency of the simulation and rendering technologies of virtual acoustics. In particular the latest systems are well suited to reproduce sound fields whose tridimensional characteristics are quite close to the real ones both objectively and subjectively. In this framework the Ambiophonics system, consisting of the merging of a double stereo-dipole and of an first order Ambisonics system, has gained particular attention. In the configuration used in this work the system employs twelve loudspeakers and is fed either with measured binaural and B-format data or by simulated ones. Based on this apparatus listening tests have been prepared in the Italian language to investigate the acoustics of classrooms and to focus on the effect of the directional characteristics of noise (i.e., fan-coils or tapping from the upper floor) on the intelligibility of words. The whole apparatus is driven by a Labview application which has also the aim of collecting the responses by the testers.

4:00-4:20 Break

4:20

5pNSa7. Noise - A stress factor? Acoustic ergonomics at schools. Gerhart Tiesler (University of Bremen/ISF, Grazer Str. 4, D-28359 Bremen, Germany, tiesler@uni-bremen.de), Markus Oberdoerster (Saint-Gobain Ecophon GmbH, Taschenmacherstr. 8, D-23556 Luebeck, Germany, markus.oberdoerster@ecophon.de)

Description of teaching reality is the main topic of this interdisciplinary examination about “acoustic ergonomics of schools.” In the first step we analyse how different teaching methods affect basic noiselevel and working noise level. Which effect has an altered room acoustics on the sound levels in the context of each teaching method. The data records available as timeseries allows for the first time analysing single phases of lessons, which are characterized by certain pedagogical characteristics or individual instruction phases. Is it correct to speak of “noise stress” or is this stress an emotional reaction to the kind of? Based on recordings of teachers heartrate we analyse in the third step the effects of noise level on the workload of the teachers as a stress reaction. A distinction is made on the subject of different teaching methods, and on the basis the different room acoustic conditions. In one primary school we had four classes with a RT$>$0.5 s and four further classes with a RT$<$0.5s. At a second primary school we analysed the effect of the room acoustic refurbishment. Finally we investigate factors of fatigue. Which effect has CO_2 in the classroom on all people, and the working noise level?

4:40

5pNSa8. Acoustical evaluation of nonclassroom university learning spaces. Murray Hodgson (The University of British Columbia, Department of Electrical and Computer Engineering, 2332 Main Mall, Vancouver, BC V6T 1Z4, Canada, mhodgson@interchange.ubc.ca), Jorge Garcia Moreno (The University of British Columbia, Department of Electrical and Computer Engineering, 2332 Main Mall, Vancouver, BC V6T 1Z4, Canada, 123calabaza@gmail.com)

This paper reports the results of an acoustical evaluation of non-classroom learning spaces at UBC. In twelve buildings, 25 indoor spaces -- a restaurant, a cafeteria, libraries, dedicated study spaces, building atria, etc. -- used for learning activities by at least 50 people were studied. The evaluation involved physical and acoustical (reverberation time, sound propagation, Speech Intelligibility Index) measurements, and occupant activity and satisfaction questionnaires. Questionnaires were administered three times (morning, lunch-time, and afternoon) on one day. The questionnaires asked about satisfaction with, and the effects of, the acoustical and nonacoustical environments. The acoustical measurement results were compared with established acceptability criteria. Questionnaires were analyzed for differences between times of day, test space, etc. The questionnaire responses and acoustical-measurement results were correlated. Using both as possible predictors, multivariable-regression models for predicting and explaining occupant satisfaction with, and the effects of, the acoustical environment were developed.

5:00

5pNSa9. The effect of amplification on children’s performance in the classroom. Bridget M. Shield (London South Bank University, Borough Road, SE1 0AA London, UK, shieldbm@lsbu.ac.uk), Julie E. Dockrell (Institute of Education, 25 Woburn Square, WC1H 0AA London, UK, j.dockrell@ioe.ac.uk)

The use of amplification systems in the classroom has the potential to reduce the impact of poor classroom acoustics for typically developing pupils and those with special educational needs (SENs). The immediate benefits of amplified acoustic signals on the performance of 253 primary school children, including 24 children with special needs, were examined. All participants were familiar with the use of the amplification systems. Children’s performance was assessed on two verbal measures (spelling and oral comprehension) and one non-verbal measure, using a balanced repeated measures design. It was predicted that the effects of amplification would be evident for spelling and oral language comprehension, but that there would be no discernable effect on speed of processing. Children with SEN were expected to have an added advantage with amplification. The predictions were partially supported. Both gain score analysis and ANOVAs of performance scores with amplification, controlling for performance without amplification, revealed an effect on spelling only for the typically developing children, while children with SENs benefited in both the spelling and the oral comprehension tasks. When installed appropriately, amplification improves children’s ability to decode single words; SEN children also benefit in terms of their ability to process oral input.

5:20

5pNSa10. Soundfield amplification is a poor substitute for good classroom acoustics. David Lubman (DL Acoustics, 14301 Middletown Lane, Westminster, CA 92683, USA, dlubman@dlacoustics.com), Louis C. Sutherland (Consultant in Acoustics, 27803 Longhill Dr., Rancho Palos Verdes, CA 90275-3908, USA, lou-sutherland@juno.com)

Soundfield (amplification) systems are widely and often aggressively marketed for small classrooms. In June 2006, the Acoustical Society of America (ASA) issued a public position statement on the use of sound amplification in typical small classrooms http://asa.aip.org/amplification.pdf. This paper attempts to explain why the ASA found that soundfield systems are poor substitutes for good acoustics. Good acoustics for learning requires unoccupied classroom noise levels of 35 dBA or less and midfrequency reverberation times of 0.6 s or less. At the same time, it is recognized that centralized amplifiers and sound distribution systems can provide valuable educational benefits in small classrooms with good acoustics. Such systems are useful for multimedia presentations and voice reinforcement. The authors urge that this message be brought to the attention of educators and educational facility decision makers.

Contributed Papers

5:40

5pNSa11. Subjective evaluation of acoustical quality of lecture rooms with respect to the quality of the sound reinforcement system and the level of background noise. Sanja Grubesa (Faculty of EE and Computing, Unska 3, Department of Electroacoustics, HR-10000 Zagreb, Croatia, sanja.grubesa@fer.hr), Marko Horvat (Faculty of EE and Computing, Unska 3, Department of Electroacoustics, HR-10000 Zagreb, Croatia, marko.horvat@fer.hr), Hrvoje Domitrovic (Faculty of EE and Computing, Unska 3, Department of Electroacoustics, HR-10000 Zagreb, Croatia, hrvoje.domitrovic@fer.hr)

As a part of a general questionnaire on the quality of lecturing on the Faculty of EE and Computing in Zagreb, Croatia, the students have to grade the acoustical quality of lecture rooms in an indirect way by giving answers to several questions included in that questionnaire. As the sound reinforcement system in two largest lecture halls is in rather bad condition, it is to be replaced in the near future with a new one of significantly better quality. The goal of this paper is to examine whether this change is reflected in the grades of acoustical quality given by the students. In the second part of the paper, the students' grades on acoustical quality of two smaller lecture rooms of identical size and acoustic treatment are compared. The comparison is made with respect to the position of these rooms relative to the nearby street as the source of traffic noise. The overall level and spectral content of noise are measured in those lecture rooms. The correlation is then examined between the results of these measurements and the grades given by the students.

6:00

5pNSa12. Education vs cost - Difficulties in implementing acoustical design standards in classrooms. Patricia M. Scanlon (Cerami & Associates, 404 Fifth Avenue, New York, NY 10018, USA, pscanlon@ceramiassociates.com), James Perry (Cerami & Associates, 404 Fifth Avenue, New York, NY 10018, USA, jperry@ceramiassociates.com), Victoria J. Cerami (Cerami & Associates, 404 Fifth Avenue, New York, NY 10018, USA, vcerami@ceramiassociates.com)

Good acoustical design standards for classrooms are established - quiet background noise levels due to mechancial systems and exterior sources, use of absorptive finishes to control reverberation, walls, doors and floors able to control airborne and impact noise transmission. So why do so many schools fail to follow these guidelines? Are private schools even less likely to adhere to these standards? What are the construction and equipment options? We will discuss various educational standards, as well as the reticence of school planners to comply. We will discuss what schools often build, versus what one might say they should build, together with cost comparison. We will review case studies where guidelines were, and were not, followed - examples that reflect the acoustician's mission to educate educators on the value of acoustical design considerations in classrooms.

Session 5pNSb

Noise, Structural Acoustics and Vibration, Physical Acoustics, and EURONOISE: Sound and Vibration from Explosions in Air II

Roger Waxler, Cochair
University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677, USA

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

Invited Papers

2:00

5pNSb1. High-altitude infrasound calibration experiments. Henry E. Bass (The University of Mississippi - NCPA, 1 Coliseum Drive, University, MS 38677, USA, pabass@olemiss.edu), Eugene T. Herrin (Southern Methodist University, P. O. Box 750395, Dallas, TX 75275, USA, herrin@passion.isem.smu.edu), Paul Golden (Southern Methodist University, P. O. Box 750395, Dallas, TX 75275, USA, pgolden@mail.smu.edu), Robert Woodward (Incorporated Research Institutions for Seismology, 1200 New York Avenue, NW, Suite 800, Washington, DC 20005, USA, woodward@iris.edu), Douglas Drob (Naval Research Laboratory, Space Science Division, 4555 Overlook Avenue, Washington, DC 20375, USA, douglas.drob@nrl.navy.mil), Michael A. Hedlin (University of San Diego California, Scripps Institute of Oceanography, 9500 Gilman Drive, La Jolla, CA 92093, USA, hedlin@ucsd.edu), Catherine De Groot-Hedlin (University of San Diego California, Scripps Institute of Oceanography, 9500 Gilman Drive, La Jolla, CA 92093, USA, cdghedlin@gmail.com), Kris Walker (University of San Diego California, Scripps Institute of Oceanography, 9500 Gilman Drive, La Jolla, CA 92093, USA, walker@ucsd.edu), Milton Garces (Infrasound Laboratory, University of Hawaii, 73-4460 Queen Kaahumanu Highway #119, Kailua-Kona, HI 96740, USA, milton@isla.hawaii.edu), Curt Szuberla (University of Alaska, 903 Koyukuk Drive, Fairbanks, AK 99775, USA, cas@gi.alaska.edu), Rod Whitaker (Los Alamos National Laboratory, EES-2 MS J577, Los Alamos, NM 87545, USA, rww@lanl.gov)

At the 152nd Meeting of the Acoustical Society of America, Andre and Bass reported an infrasound experiment conducted at White Sands Missile Range during the 2005-2006 time frame. The experiment consisted of exploding a 22.4 kg charge at altitudes from 31.3 km to 49.6 km then recording the waveforms at 30 infrasound arrays (not all at the same time) at distances up to 1200 km from the source. The analysis is not yet complete but some preliminary observations have been reported in the most recent issue of Acoustics Today. This talk will summarize the findings published in Acoustics Today and offer suggestions to others who might want to access and analyze the data.

2:20

5pNSb2. Infrasonic and seismic signals from explosions in Northwestern Europe. Laeslo G. Evers (Royal Netherlands Meteorological Institute (KNMI), PO Box 201, 3730 AE De Bilt, Netherlands, evers@knmi.nl), Hein Haak (Royal Netherlands Meteorological Institute (KNMI), PO Box 201, 3730 AE De Bilt, Netherlands, haak@knmi.nl)

Large explosions often generate both seismic and infrasound signals that can be detected over large ranges, i.e., hundreds of kilometers. Ground truth of such explosions is available from direct observations and/or seismic signals and can be used to get insight in the propagation of infrasound. Long range infrasound propagation is controlled by the state of the atmosphere. Tropospheric, stratospheric, and thermospheric ducts might exist and have implications on the signal characteristics and their detectability. In this study, we will show results of studies on explosions in Northwestern Europe using both seismic and infrasound data. These observations are used to derive source characteristics like: location, origin time and yield. Furthermore, the propagation characteristics of infrasound will be addressed.

2:40

5pNSb3. Infrasound propagation in the zone of silence. Paul Golden (Southern Methodist University, P. O. Box 750395, Dallas, TX 75275, USA, pgolden@mail.smu.edu), Petru Negraru (Southern Methodist University, P. O. Box 750395, Dallas, TX 75275, USA, pnegraru@smu.edu), Eugene T. Herrin (Southern Methodist University, P. O. Box 750395, Dallas, TX 75275, USA, herrin@passion .isem.smu.edu)

Two controlled source experiments were conducted in Nevada in 2006 and 2007 to study infrasound signal propagation at distances less than 300 km from the source. In 2006 three temporary infrasound arrays were deployed at distances of 76, 108, and 157 from the source. In 2007 the site at 157 km was reoccupied, and data was also recorded at 288 km from the source. Interesting results were derived from the travel time analysis. In 2006 the site at 76 km recorded both tropospheric and stratospheric arrivals, while at 108 and 157 km only stratospheric arrivals were recorded. In 2007 the site at 157 km recorded both tropospheric and stratospheric arrivals, while at 288 km both stratospheric and thermospheric arrivals were recorded. Atmospheric modeling with the InfraMAP software failed to predict returning rays or pressure levels similar to the observed data. Because of the large amplitude variations we attempt to estimate the yields of the explosions using the predominant frequency content of the signals. The physical basis for such a method is found in an increased acoustic transit time of the explosion blast radius with increased yield. Preliminary results suggest this is possible.

3:00

5pNSb4. Validating upper-wind models using infrasound from active volcanoes. Alexis Le Pichon (CEA-DASE, Arpajon Cedex, 91297 Bruyères-le-Châtel, France, alexis.le-pichon@cea.fr), Karl Antier (CEA-DASE, Arpajon Cedex, 91297 Bruyères-le-Châtel, France, karl.antier@cea.fr), Sylvie Vergniolle (Institut de Physique du Globe, 4 Place Jussieu, 75252 Paris, France, vergniolle@ipgp.jussieu.fr), Michel Lardy (IRD Center, BPA5, Cedex, 98848 Noumea, New Caledonia, lardy@noumea.ird.nc), Douglas Drob (Naval Research Laboratory, Space Science Division, 4555 Overlook Avenue, Washington, DC 20375, USA, douglas.drob@nrl.navy.mil)

Known and quasipermanent infrasonic sources are needed to evaluate and improve upper-wind models. Infrasounds generated by active volcanoes offer a unique opportunity for atmospheric studies. The Yasur volcano in the Vanuatu archipelago is an outstanding source of infrasonic waves due to its regular activity. This volcano is permanently monitored by the IS22 infrasound station located in New Caledonia and by one microbarometer installed close to its crater. A five-year monitoring period of Yasur at short and large propagation range provides new insights on quantitative relationships between infrasonic observables and atmospheric specifications. This experimental setting is proposed to validate consistently the Naval Research Laboratory Ground to Space (NRL-G2S) semi-empirical atmospheric model up to the stratosphere. The propagation modeling results accurately explain seasonal changes as well as small short-timescale variations of the infrasonic observables. This study demonstrates that the use of appropriate propagation tools along with the NRL-G2S specifications provides accurate enough results for most of the long-range observations for the purpose of operational infrasound monitoring.

3:20

5pNSb5. Semianalytical modeling of plate flexural waves generated by laser-initiated air shock waves. Vasil B. Georgiev (Loughborough University, Department of Aeronautical and Automotive Engineering, Ashby Road, LE11 3TU Loughborough, UK, V.Georgiev@lboro.ac.uk), Victor V. Krylov (Loughborough University, Department of Aeronautical and Automotive Engineering, Ashby Road, LE11 3TU Loughborough, UK, V.V.Krylov@lboro.ac.uk), Qin Qin (University of Hull, Department of Engineering, Cottingham Road, HU6 7RX Hull, UK, q.qin@hull.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)

The paper describes the results of the semianalytical modeling of the interaction of laser-initiated air shock waves with an infinite elastic plate. The impact of the incident shock wave on the plate has been approximated by an equivalent cylindrically diverging surface force resulting from the combined surface pressure of the incident and reflected shock waves. This force has been then represented in the wave number-frequency domain by means of Hankel and Fourier transforms which have been carried out numerically - and the problem has been solved using the Green's function method applied to an infinite plate. The resulting frequency spectra and time histories of the generated flexural wave pulses have been calculated for different values of laser pulse energy and for different heights of the laser beam focusing above the plate surface. The obtained theoretical results have been compared with the results of the reduced-scale model experiments on shock wave interaction with the ground in which large plastic and wooden plates have been used to represent the ground surface. The comparison shows that the obtained semi-analytical results are in good agreement with the experimental ones.

3:40

5pNSb6. Ground effects on sound and vibration from explosions. Christian Madshus (NGI, P.O. Box 3930 Ullevaal Stadion, Sognsveien 72, 0806 Oslo, Norway, cm@ngi.no), Finn Løvholt (NGI, P.O. Box 3930 Ullevaal Stadion, Sognsveien 72, 0806 Oslo, Norway, flo@ngi.no), Ra Cleave (NGI, P.O. Box 3930 Ullevaal Stadion, Sognsveien 72, 0806 Oslo, Norway, rc@ngi.no), Karin Rothschild (NGI, P.O. Box 3930 Ullevaal Stadion, Sognsveien 72, 0806 Oslo, Norway, kmr@ngi.no), Zenon Cetina-Medina (NGI, P.O. Box 3930 Ullevaal Stadion, Sognsveien 72, 0806 Oslo, Norway, zmc@ngi.no)

Low frequency sound from military activity and explosions do propagate over large distances. The sound pressure may induce substantial vibration in the ground and particularly in buildings. Such vibrations turn out to be a major cause of complaints among neighbouring communities around training fields and blast sites. We will present investigations on long range propagation of low frequency sound and sound-induced vibration, based on a substantial amount of data collected during a series of full scale tests performed in Norway over the last 14 years. All data are assembled in the NORTRIAL database, which is now publicly available. Meteorology and ground interaction largely influence the sound propagation and vibration response. At low frequency meteorological- and ground effects may intricately interact and lead to large, apparently random variability in sound pressure at large distances. Statistically based investigations on the sound propagation and its variability will be presented. Particular focus will be on the ground interaction effects, and a new and more extensive system for ground classification, based on cartographic data, empirical and numerical modeling will be introduced. Findings on building response to outdoor low frequency sound and transfer mechanisms from outdoor pressure to indoor sound and vibration will be presented.

4:00-4:20 Break

4:20

5pNSb7. Deducing ground structure using seismic pulses originating from an outdoor explosive source. 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)

Near-surface layering of ground soil can influence propagation of acoustic and seismic pulses originating from above-surface sources. Simultaneous recording of acoustic air pressure and seismic radial and vertical particle velocities resulting from a small, above ground explosion is used to obtain information about soil structure near the surface. Assuming nonlinear effects to be small at the ranges

of interest here, a numerical model called Fast Field Program for Layered Air Ground Systems (FFLAGS), developed originally for continuous sound sources above a porous elastic ground is used to model a porous elastic layered ground system. Suitable optimisation methods were used to predict a set of best fit parameters for the near-surface ground structure. It is shown that the model can explain multiple seismic arrivals and give a reasonable prediction of wave speeds and layer depths while the pressure pulse can predict permeability of the surface soil.

4:40

5pNSb8. Predictions for the influence of the nocturnal jet on the long range propagation of impulsive signals. Roger Waxler (University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677, USA, rwax@olemiss.edu), Kenneth E Gilbert (National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA, kgilbert@olemiss.edu), Carrick L. Talmadge (The University of Mississippi - NCPA, 1 Coliseum Drive, University, MS 38677, USA, clt@olemiss.edu), Xiao Di (National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA, xiaodi@olemiss.edu)

On clear nights, inland over flat ground, one generally finds a temperature inversion in the first hundred meters or so of the atmosphere. Near the ground, winds tend to be light, increasing with altitude. Above the temperature inversion one finds a stiff geostrophic wind known as the nocturnal jet. Theoretical predictions, based on ray theory and expansions in vertical modes, for the effect of the nocturnal jet on the long range propagation of impulsive signals are presented. For sufficiently short ranges, less than 3 km or so, the nocturnal jet has no effect. At these ranges only the temperature inversion and the light winds in the inversion play a role. At longer ranges the nocturnal jet significantly alters the arrival structure of the propagated signal. It is predicted that, due to coincident arrivals from the inversion and from the nocturnal jet, there is a segment of ranges about 1 km long, beginning at about 4 km, in which the amplitude of the first arrival becomes anomalously large and then splits into two distinct arrivals.

5:00

5pNSb9. The physics of pulse propagation in the nocturnal atmospheric boundary layer: measurement and theory. Carrick L. Talmadge (The University of Mississippi - NCPA, 1 Coliseum Drive, University, MS 38677, USA, clt@olemiss.edu), Roger Waxler (University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677, USA, rwax@olemiss.edu), Kenneth E Gilbert (National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA, kgilbert@olemiss.edu), Jin So (National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA, jso@olemiss.edu), Rommel Stribling (National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA, eercs@olemiss.edu), Xiao Di (National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA, xiaodi@olemiss.edu)

A series of experiments designed to probe the effect of the nocturnal atmosphere on low-frequency (10- 500 Hz) sound propagation will be discussed, and their ramifications explored. These experiments involve detecting arrivals from a propane cannon on a vertical array of microphones 1-3 km from the source, which were collected simultaneously with meteorological experiments designed to measure the vertical temperature and horizontal wind velocity profiles. Chief among the results of these experiments is the observation of a nocturnal model structure that has a significant surface wave component at frequencies below 150 Hz. At higher frequencies and longer propagation distances ($>$1.5 km), the surface wave is not observed due to attenuation from its interaction with the ground. At higher frequencies, the model structure displays a characteristic “quiet height” first described in Waxler et al. (2006). The potential application of these results for remote sensing the atmospheric boundary layer are discussed, and contrasted with other methods of measurement of the atmospheric profile.

Contributed Paper

5:20

5pNSb10. Nonlinear parabolic equation model for finite-amplitude sound propagation in an inhomogeneous medium over a nonflat, finite-impedance ground surface. Thomas Leissing (CSTB, 24 rue Joseph Fourier, 38400 Saint-Martin-d’Hères, France, thomas.leissing@cstb.fr), Philippe A. Jean (CSTB, 24 rue Joseph Fourier, 38400 Saint-Martin-d’Hères, France, philippe.jean@cstb.fr), Jérôme Defrance (CSTB, 24 rue Joseph Fourier, 38400 Saint-Martin-d’Hères, France, jerome.defrance@cstb.fr), Christian Soize (Université de Marne la Vallée, 5, Boulevard Descartes, 77454 Marne la Vallée, France, soize@univ-mlv.fr)

A nonlinear parabolic equation (NPE) model for weakly nonlinear sound propagation in an inhomogeneous medium is described. The model being formulated in the time domain, complex impedances cannot be used to simulate ground surfaces. A second NPE model is thus derived to include the medium in the computational system. Based on a nonlinear extension of the Zwikker-Kosten model for rigidly-framed porous media, it allows to include Forchheimer’s nonlinearities. Both models are then adapted to terrain-following coordinates, and used together with an interface condition, allow to simulate finite-amplitude sound propagation over a nonflat, finite-impedance ground surface. Numerical examples show that the NPE model is in good agreement with the solutions of the frequency domain boundary element method. Applications of this model to the simulation of sound propagation from explosions in air are then discussed.

Invited Papers

5:40

5pNSb11. Mean vs event sound-level prediction: obtaining consistency between atmospheric data inputs, propagation models, and the predictand. David K. Wilson (U.S. Army Engineer Research and Development Center, 72 Lyme Rd., Hanover, NH 03755-1290, USA, D.Keith.Wilson@usace.army.mil), Chris L. Pettit (U.S. Naval Academy, Aerospace Engineering Dept., 590 Holloway Rd., MS 11-B, Annapolis, MD 21402, USA, petitcl@usna.edu), Vladimir E. Ostashev (NOAA/Earth System Research Laboratory, 325 Broadway, Boulder, CO 80305, USA, vladimir.ostashev@noaa.gov), Matthew S. Lewis (U.S. Army Engineer Research and Development Center, 72 Lyme Rd., Hanover, NH 03755-1290, USA, Matthew.S.Lewis@usace.army.mil)

The following, deceptively challenging, questions are addressed: What are the most suitable atmospheric data resources and propagation models for predicting event (explosion and other short duration) sound-exposure levels? Do these differ from those most suitable for predicting mean sound levels? Atmospheric data typically consist either of single, "snapshot" samples of the vertical profiles, as from a weather balloon, or average vertical profiles, as from climatology or a numerical weather model. Recent statistical studies, based on high-resolution atmospheric simulation, demonstrate the superiority of mean profiles for prediction of *both mean and event* sound levels, even when single-sample profiles are synchronized to and collected along the path of the propagation event. Running propagation models "blind" to the nature of the atmospheric inputs is shown to be hazardous: predictions from mean profiles lack turbulent scattering, thus underestimating sound levels near the ground, whereas predictions from single-sample profiles implicitly assume the turbulence has infinite horizontal extent, thus overestimating sound levels. Some desirable consistency results from numerically solving parabolic equations (PEs) for statistical *moments* of the sound pressure, rather than conventional deterministic PEs. The moment PEs directly predict mean sound levels or the expected value and variability of event sound-exposure levels.

6:00

5pNSb12. Beam-forming and dispersion measurements at the edge of a pine forest. Michael J. White (US Army Engineer Research and Development Center, 2902 Newmark Drive, Champaign, IL 61826, USA, Michael.J.White@usace.army.mil), Michelle E. Swearingen (Norwegian Defense Research Establishment/US Army, Postboks 25, 2027 Kjeller, Norway, michelle.swearingen@ffi.no)

Beam-forming analyses were performed on four-microphone linear arrays placed just within the interior of a pine forest to separate signal arrivals by direction from an impulsive sound source in an open field. The arrays were deployed in three configurations: transverse, longitudinal, and vertical. The arrays spacings were organized in a pattern that provides six baselines with four microphones. Longitudinal and transverse configurations show arrivals scattered from trunks, and the vertical configurations indicate some refraction and scattering by the canopy. Signal dispersion curves developed using the four-microphone arrays had greater variation in trace velocity at higher frequencies in each configuration.

FRIDAY AFTERNOON, 4 JULY 2008 ROOM 251, 4:20 TO 6:20 P.M.

Session 5pNSc

Noise and EURONOISE: Propagation and Urban Noise II

Jérôme Defrance, Chair
CSTB, 24 rue Joseph Fourier, Saint-Martin-d'Hères, 38400, France

Contributed Papers

4:20

5pNSc1. Road traffic noise from viaducts in mountainous areas. Jérôme Defrance (CSTB, 24 rue Joseph Fourier, 38400 Saint-Martin-d'Hères, France, jerome.defrance@cstb.fr), Matthieu Beyret (CSTB, 24 rue Joseph Fourier, 38400 Saint-Martin-d'Hères, France, matthieu.beyret@cstb.fr), Marine Baulac (CSTB, 24 rue Joseph Fourier, 38400 Saint-Martin-d'Hères, France, marine.baulac@cstb.fr), Philippe A. Jean (CSTB, 24 rue Joseph Fourier, 38400 Saint-Martin-d'Hères, France, philippe.jean@cstb.fr)

Mountainous zones in Europe, such as the Alps, represent huge areas where many viaducts are built, most of them for motorways. The way the sound grazes the asphalt surface from the low and high traffic equivalent sources up to the road edges, and how it then diffracts towards dwellings is a complex mechanism. The standard approaches are suited to plain situations but fail in predicting finely sound propagation behaviour for such geometries. In this paper, one gives the main trends of received noise levels from viaducts as a function of both their geometry and the receiver location. A 2D Boundary Element Method is used for predictions since meteorological effects can be neglected for the short propagation (a few hundreds meters). This assumption makes sense since the viaduct considered in this work is sufficiently high (20 m) and the ground effect is weakly affected by refraction. Different configurations are then simulated in order to address and discuss several geometrical effects, such as: platform elevation, low height barriers addition, complex shape barriers and presence of a central gap in the platform.

4:40

5pNSc2. Characteristics of road traffic noise level statistics for shielded areas. Jens Forssén (Division of Applied Acoustics, Chalmers University of Technology, 41296 Göteborg, Sweden, jens.forssen@chalmers.se), Maarten Hornikx (Applied Acoustics, Chalmers University of Technology, Sven Hultins Gata 8a, SE-41296 Gothenburg, Sweden, maarten.hornikx@chalmers.se)

For noise immission, it is of interest to study other noise level statistics besides the long-term equivalent levels and maximum levels. By further analysis of the time variations of the noise level, an improved description of the negative effects of the noise may be achieved, for instance concerning perceived annoyance. Here, noise level histograms, i.e., probability density functions of sound pressure levels, from controlled recordings have been investigated. This has been made for a situation of special interest, which is a courtyard shielded from a dominating road traffic noise source. It has been reported previously that many shielded urban areas show levels that are considerably higher than the equivalent level, described as an upward tail of the histogram, which is not a usual characteristic of directly exposed areas. From the analysis made here, it is shown that the upward tail, i.e., the higher levels, of the shielded area is caused by locally occurring, unshielded road traffic events. It is concluded that the upward tail as a common characteristic of shielded urban areas may well be due to locally occurring noise events, for instance due to local road traffic.

5:00

5pNSc3. Use of the transmission line matrix method for the sound propagation modeling in urban area. Gwenaël Guillaume (Lab. Central des Ponts et Chaussées, Division Entretien, Sécurité et Acoustique des Routes, Route de Bouaye - BP 4129, 44341 Bouguenais Cedex, France, Gwenael.Guillaume@lcpc.fr), Judicaël Picaut (Lab. Central des Ponts et Chaussées, Division Entretien, Sécurité et Acoustique des Routes, Route de Bouaye - BP 4129, 44341 Bouguenais Cedex, France, Judicael.Picaut@lcpc.fr), Guillaume Dutilleux (Lab. Régional des Ponts et Chaussées, 11, rue Jean Mentelin, BP 9, 67035 Strasbourg Cedex 2, France, Guillaume.Dutilleux@equipement.gouv.fr)

This paper deals with the sound propagation modeling in urban area. This problematic requires to take into account many phenomena that can have a substantial impact as well in semienclosed spaces as on long-range outdoor sound propagation, such as reflections and absorption on the frontages and on the ground, atmospheric attenuation, sound velocity variations related with wind and temperature vertical gradients, atmospheric turbulences. The numerical method used is the TLM (transmission line modeling), which has been originated in electromagnetism and adapted for acoustics applications. It consists in a physical rendering of the waves propagation based on the Huygens' principle. It is established on a spatiotemporal discretization of the domain using an iterative temporal process for sound pressure propagation, instead of the resolution of mathematical equations. The TLM model has then been developed in two and three dimensions allowing to combine all the phenomena affecting the sound propagation in urban area. Numerical simulations are given for canyon streets.

5:20

5pNSc4. Predictions of sound pressure levels in streets using a diffusion model: numerical validations and experimental comparisons. Alexis Billon (Universite de Liege, INTELSIG group - Département E.E.I., B28 Sart-Tilman, 4000 Liege, Belgium, abillon@ulg.ac.be), Judicaël Picaut (Lab. Central des Ponts et Chaussées, Division Entretien, Sécurité et Acoustique des Routes, Route de Bouaye - BP 4129, 44341 Bouguenais Cedex, France, Judicael.Picaut@lcpc.fr)

Predictions of sound propagation in urban areas have attracted a considerable over the years. If the sound energy is assimilated to particles with a constant energy, their movement can be described by a transport equation. In canyon streets, this transport equation can be reduced to a diffusion equation whose expression is more simple. In this presentation, sound absorption at the boundaries (buildings facades and ground), as well as atmospheric sound attenuation are introduced. The problem is then solved numerically using a finite elements method for the configuration of a canyon street. A systematic validation of the obtained model is carried out in terms of sound pressure level by comparison to numerical simulations taken from the literature. Comparisons with experimental data are then conducted. Finally, applications in more complex configurations are presented.

5:40

5pNSc5. A cellular automaton for urban traffic noise. Erik Salomons (TNO Science and Industry, Stieljesweg 1, 2628CK Delft, Netherlands, erik.salomons@tno.nl)

Propagation of traffic noise in a city is a complex phenomenon, due to multiple reflection, diffraction, and scattering at irregular facades of buildings. These effects may be calculated with computer models based on numerical integration of the basic acoustic equations, but in practice these models can be applied only to small urban regions due to limited computer power. Here we propose a new approach for simulating urban traffic noise: a cellular automaton (CA) based on simple update rules for the configuration of cars in a city, and simple rules for propagation of sound to receivers. An example is presented for a square model city of 25 km^2, consisting of 10^6 square cells. The CA employs a time integration step of 0.3 s, and includes noise contributions from all cars in the city. The fluctuating sound level is computed for a period of 24 h, both for a receiver along a street and for a receiver that is screened by buildings. While the sound level at the first receiver shows sharp peaks corresponding to passages of cars, the sound level fluctuations at the screened receiver are much smaller as most of the sound energy comes from distant cars in this case.

6:00

5pNSc6. Numerical prediction of the effect of traffic lights on the vehicle noise at urban street intersections. Jorge Parrondo (Universidad de Oviedo, Dep. de Energía, Campus de Viesques s/n, 33203 Gijón, Spain, parrondo@uniovi.es), Ruben Zurita (Universidad de Oviedo, Dep. de Energía, Campus de Viesques s/n, 33203 Gijón, Spain, ZURITA@lsi.uniovi.es), Jose A. Corrales (Universidad de Oviedo, Dep. de Energía, Campus de Viesques s/n, 33203 Gijón, Spain, JA@lsi.uniovi.es), Joaquin Fernandez (Univ. de Extremadura, Escuela de Ing. Industriales, 06071 Badajoz, Spain, ffrancos@unex.es)

Intersections of urban streets are particularly noisy locations due to the addition of the noise from vehicles at different streets, a long period of noise emission from queuing vehicles with traffic lights in red and the noise from accelerating vehicles. Besides, the traffic lights impose a modulation in the passage of the vehicles, so that the subsequent noise variability along time contributes to increase the annoyance degree. This paper presents a code especially developed to simulate both the spatial and temporal distribution of the sound pressure levels induced by the vehicle traffic in street intersections. The algorithm combines a traffic model with vehicles in dynamic motion through the domain with a model for sound propagation, based on the method of virtual images in which the determination of the location of the successive image sources was reduced to a reference horizontal plane. Sound emission from each vehicle was established according to the Harmonoise guidelines. After verifying the code predictions by comparison with measurements at several street intersections, the method was applied systematically to investigate the effect of varying the regulation parameters of the traffic lights on the Leq and L10 distributions for some particular cases.

Session 5pPAa

Physical Acoustics: Ducts and Waveguides II

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

Contributed Papers

2:00

5pPAa1. A higher order parabolic equation for predicting in-duct propagation in high frequencies. Patrice Malbéqui (ONERA, 29 avenue de la Division Leclerc, 92322 Châtilllon, France, patrice.malbequi@onera.fr)

Complementary methods are required to predict the in-duct propagation in a large frequency range including the linear absorption and the flow effects. Numerical methods solving the Euler's equations are pertinent for rotational flow but limited to the low frequencies. The Boundary Element Method is applicable in a large frequency range assuming a homogeneous flow. The ray-model is valid in high frequencies but scattering effects are difficult to implement. This paper presents the capabilities of a Higher-Order Parabolic Equation (HOPE) to handle duct propagation in the high frequency range. In contrast with the so-called standard PE and the wide-angle PE, the HOPE improves the accuracy of the solution due to its wider propagation aperture angle, especially close to the cutoff frequency. Several duct propagation configurations including flow and liner are considered. Using a marching algorithm, the HOPE computes in a very short CPU time the sound propagation and represents an attractive alternative to the ray-model in the high frequency range. [Work supported by Airbus-France.]

2:20

5pPAa2. An integrated multimodal acoustic particle manipulator and optical evanescent field waveguide. Peter Glynne-Jones (University of Southampton, School of Engineering Sciences, University Road, SO17 1BJ Southampton, UK, p.glynne-jones@soton.ac.uk), Martyn Hill (University of Southampton, School of Engineering Sciences, University Road, SO17 1BJ Southampton, UK, m.hill@soton.ac.uk), Rosemary J. Townsend (University of Southampton, School of Engineering Sciences, University Road, SO17 1BJ Southampton, UK, R.J.Townsend@soton.ac.uk), Nicholas R. Harris (University of Southampton, Electronics and Computer Science, SO17 1BJ Southampton, UK, nrh@ecs.soton.ac.uk), James S. Wilkinson (University of Southampton, Optoelectronics Research Centre, SO17 1BJ Southampton, UK, jsw@ecs.soton.ac.uk), Fan Zhang (University of Southampton, Optoelectronics Research Centre, SO17 1BJ Southampton, UK, faz@orc.soton.ac.uk), Tracy Melvin (University of Southampton, Optoelectronics Research Centre, SO17 1BJ Southampton, UK, tm@ecs.soton.ac.uk)

A new acoustic/optical/microfluidic system is presented for the manipulation of bead-tagged DNA molecules. Acoustic radiation forces are used to manipulate microspheres into and away from the evanescent field of a laser coupled waveguide that is integrated into the reflector of the acoustic chamber. With suitable fluorophores the presence of the target DNA can be detected with a fluorescence microscope enabling large populations of beads to be examined simultaneously. The integrated waveguide and multimodal acoustic chamber are presented here, with results showing that the microspheres can be successfully detected as they are brought into the evanescent field using a quarter-wave acoustic configuration. It is also shown that by measuring the time of flight of a microsphere between the half- and quarter-wave nodal planes the bead size can be determined, providing a means of multiplexing the detection (detecting a range of different target DNA sequences).

2:40

5pPAa3. Trapped wave in plan waveguides including gaussian varying section domain. Patrick Marical (LOMC FRE-3102 CNRS, Groupe Ondes Acoustiques, University of Le Havre, Place Robert Schuman, BP 4006, 76610 Le Havre, France, patrick.marical368@univ-lehavre.fr), Mounsif Ech-Cherif El-Kettani (LOMC FRE-3102 CNRS, Groupe Ondes Acoustiques, University of Le Havre, Place Robert Schuman, BP 4006, 76610 Le Havre, France, elkettani@univ-lehavre.fr), Zahia Hamitouche (LOMC FRE-3102 CNRS, Groupe Ondes Acoustiques, University of Le Havre, Place Robert Schuman, BP 4006, 76610 Le Havre, France, zahia.hamitouche@univ-lehavre.fr), Mihai Valentin M. Predoi (University Politechnica of Bucharest, Department of Mechanics, 060032 Bucharest, Romania, predoi@cat.mec.pub.ro)

In previous works on plan waveguides including an area of varying section of Gaussian shape, we have observed experimentally and numerically the existence of a trapped wave in the Gaussian varying section domain in the case of the A1 incident Lamb mode. The purpose of this work is to highlight the existence of this trapped wave, depending on the equation of the Gaussian profile of the varying section domain. This study is carried out numerically using FEM, as our numerical model has been previously validated. The results obtained show that the phenomenon of trapped wave is strongly linked to the Lamb wave conversion phenomenon: when the incident Lamb wave gives rise to a trapped wave, it is systematically converted into other Lamb waves transmitted outside the area of varying section. Otherwise, it is totally transmitted without any conversion and in this case, any trapped wave exists. The conversion phenomenon is quantified and we show that it is important and depend on the breaking symmetry of the incident Lamb mode by the varying section area.

3:00

5pPAa4. Intermodal coupling in a dissipative fluid filling a rough-walled waveguide. Tony Valier-Brasier (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, tony.valier-brasier.etu@univ-lemans.fr), Catherine Potel (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, catherine.potel@univ-lemans.fr), Michel Bruneau (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, michel.bruneau@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)

The present study follows recent works dealing with the analytical model of an acoustic field in fluid-filled waveguides with rough walls. In these works, the acoustic field is obtained from the coupling between Neumann eigenmodes of the regularly shaped waveguide which bounds outwardly the rough walls of the waveguide considered, using integral formulation with suitable Green function. The effect of the roughness is expressed in such a way that two intermodal coupling mechanisms are highlighted: a bulk coupling and a surface coupling, the first one depending on the depth of the roughness and the second one depending in addition on the local slope. Moreover, a phonon relation is involved when the rough profile is periodic. The aim of the present study is to account for the thermo-viscous boundary layer effects through eigenmodes which satisfy appropriate mixed boundary conditions, leading to a better understanding of the physical mechanisms when resonances and phonon relationship are involved.

3:20

5pPAa5. Evaluation of the lined duct performances based on a 3D two port scattering matrix. Mohamed Taktak (Université de Technologie de Compiègne, Centre de Recherche Royallieu, BP20529, 60205 Compiègne, France, mohamed.taktak@utc.fr), Jean Michel Ville (Université de Technologie de Compiègne, Centre de Recherche Royallieu, BP20529, 60205 Compiègne, France, jean-michel.ville@utc.fr), Mohamed Haddar (Unité de Modélisation, Mécanique et de Production (U2MP), Ecole Nationale d'Ingénieurs de Sfax, BP 3038, 3038 Sfax, Tunisia, mohamed.haddar@enis.rnu.tn), Félix Foucart (Université de Technologie de Compiègne, Centre de Recherche Royallieu, BP20529, 60205 Compiègne, France, felix.foucart@utc.fr)

The scattering matrix constitutes a good tool to characterize a lined duct. In fact, this matrix relates the incoming modal pressures to the out coming ones and contains detailed information and per mode about the transmission, reflection, and conversion properties of the duct. It depends only on the geometric and acoustic properties of the duct. The two port acoustic dissipation and attenuation interest the designers of lined duct like aircraft engine manufacturers to evaluate the duct performances. These values can be deduced from the two port scattering matrix and from the vector of incoming modal pressures. In this work, a study about the two port acoustic dissipation and attenuation computed from the scattering matrix and for different cases of incoming modal pressures are presented. Scattering matrices used in this study are measured by an experimental procedure developed at the University of Technology of Compiègne based upon the experimental setup realized during the European Project DUCAT. The experimental acoustic power dissipation and attenuation are computed for different cases of modal structure on the both side of the duct. Then, these results were confronted with ones given by a theorical study of the problem based on the finite element method.

3:40-4:00 Break

4:00

5pPAa6. Elaboration of a sacttering matrix measurement procedure using the p-v probe. Yamen Kchaou (Université de Technologie de Compiègne, Centre de Recherche Royallieu, BP20529, 60205 Compiègne, France, yamen.kchaou@yahoo.fr), Mohamed Taktak (Université de Technologie de Compiègne, Centre de Recherche Royallieu, BP20529, 60205 Compiègne, France, mohamed.taktak@utc.fr), Jean Michel Ville (Université de Technologie de Compiègne, Centre de Recherche Royallieu, BP20529, 60205 Compiègne, France, jean-michel.ville@utc.fr), Mohamed Haddar (Unité de Modélisation, Mécanique et de Production (U2MP), Ecole Nationale d'Ingénieurs de Sfax, BP 3038, 3038 Sfax, Tunisia, mohamed.haddar@enis.rnu.tn), Félix Foucart (Université de Technologie de Compiègne, Centre de Recherche Royallieu, BP20529, 60205 Compiègne, France, felix.foucart@utc.fr)

The scattering matrix which relates traveling waves amplitudes as state variables has been shown to be more attractive than transfer or mobility matrices since it reflects the fundamental duct nature: it gives a more complete description of the transmission, reflection, and conversion properties of the duct. In the University of Technology of Compiègne, an experimental procedure was developed to measure this matrix: a p-p probe mounted on a setup designed during DUCAT project is used to measure pressures at two cross sections on the both side of the test lined duct, then by using a modal decomposition and separation techniques, the scattering matrix is deduced. In this paper, a method to measure the multimodal scattering matrix based on the use of a p-v probe getting simultaneously the acoustic pressure and velocity at one section on the both side of the test duct is developed. A comparison of some acoustics values (scattering matrix coefficients, acoustic powersœ) of a hard wall duct straight duct obtained by each technique with the theory is presented to evaluate its advantages and limitations.

4:20

5pPAa7. Axisymmetrical and nonaxisymmetrical guided waves propagating in a solid elastic cylinder embedded in a solid medium. Slah Yaacoubi (LCPC, Lab. Central des Ponts et chaussées (LCPC), Route de Bouaye-BP 4129, 44341 Bouguenais, France, slah.yaacoubi@lcpc.fr), Laurent Laguerre (LCPC, Lab. Central des Ponts et chaussées (LCPC), Route de Bouaye-BP 4129, 44341 Bouguenais, France, laurent.laguerre@lcpc.fr), Eric Ducasse (LMP, Lab. de Mécanique et Physique (LMP), 351, Place de la Libération, 33405 Talence, France, e.ducasse@lmp.u-bordeaux1.fr), Marc Deschamps (LMP, Lab. de Mécanique et Physique (LMP), 351, Place de la Libération, 33405 Talence, France, m.deschamps@lmp.u-bordeaux1.fr)

For NDT of rods and pipes, fundamental characteristics of guided waves are to be known, especially dispersion relations between frequency and wave number. A necessary step before detecting defects is to be able to calculate the propagated elastodynamical field in healthy waveguides. Thus, the goal of this work is the calculation of this field propagating in a cylindrical stratified waveguide. The incident field is generated at the end of the cylinder by a force or velocity source which is off-axis and Gaussian distributed. First, Vector Hankel transform and Fourier series are combined to decompose this field with respect to the angular position. Second, each component is decomposed into an infinite sum of rays, i.e., elementary generalized conical waves. These waves undergo multiple reflections with the guiding surface of the waveguide. Third, we use Generalized Debye Series (GDS) for calculating the global reflection coefficients resulting from these multiple reflections. Finally, the total field is synthesized by the summation of the incident and reflected rays. Many outputs of this code can be exploited like velocity field, stress field, energy field in 2D or 3D spatiotemporal or frequential simulations. Diagrams obtained by this code are compared with results from DISPERSE software.

4:40

5pPAa8. Coupling transfer matrix method to finite element method for the analysis of hollow body networks with passive or reactive elements. Fabien Chevillotte (Groupe d'Acoustique de l'Université de Sherbrooke, 2500, Boul. de l'Université, Département de génie mécanique, Sherbrooke, QC J1K-2R1, Canada, fabien.chevillotte@usherbrooke.ca), Raymond Panneton (Groupe d'Acoustique de l'Université de Sherbrooke, 2500, Boul. de l'Université, Département de génie mécanique, Sherbrooke, QC J1K-2R1, Canada, raymond.panneton@usherbrooke.ca), Hakim Bougrab (Groupe d'Acoustique de l'Université de Sherbrooke, 2500, Boul. de l'Université, Département de génie mécanique, Sherbrooke, QC J1K-2R1, Canada, Hakim.Bougrab@USherbrooke.ca), Christophe Chaut (Henkel Technologies, Acoustics Center, 58203 Cosne sur Loire, France, Christophe.Chaut@fr.Henkel.com), Jean-Luc Wojtowicki (Henkel Technologies, Acoustics Center, 58203 Cosne sur Loire, France, Jean-Luc .Wojtowicki@fr.Henkel.com)

This work shows how to couple transfer matrix method to finite element method with a view to analyze the acoustic response of hollow body structures with a minimum of memory requirements and computational time. An hollow body structure is made up from a series of elongated rigid-walled fluid partitions (i.e., waveguides). These fluid partitions are separated by any passive (e.g., multilayered sound barrier) or reactive elements (e.g., expansion chamber). In the proposed hybrid model, the elongated fluid partitions are modeled using 1D fluid finite elements, and the passive or reactive elements using transfer matrices. From the weak integral formulation of the acoustic problem, it is shown how the coupling with the transfer matrix is taken into account through a mixed boundary condition. After discretization of the acoustic pressure and application of the variational principle, the finite element matrix system is obtained, where only the nodal pressures in the fluid partitions remain. The transfer matrix has been converted into a kind of admittance matrix, where no additional degrees of freedom are necessary to account for the passive or reactive elements. The method is used to predict the acoustic response of a real hollow body structure. Good correlations are obtained with experimentations.

Session 5pPAb

Physical Acoustics: Ultrasonics Under Extreme Conditions II

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Frédéric Decremps, Cochair
IMPMC, Université Paris VI, 140, rue de Lourmel, Paris, 75015, France

Invited Papers

3:20

5pPAb1. Gigahertz ultrasonic interferometry at high pressure and temperature: Geophysical implications. Anastasia P. Kantor (Bayerisches Geoinstitut, University Bayreuth, 95440 Bayreuth, Germany, Anastasia.Kantor@Uni-Bayreuth.de), Steven D. Jacobsen (Department of Earth and Planetary Sciences, Northwestern University, Evansto, IL 60208-2150, USA, steven@earth.northwestern.edu), Innokenty Y. Kantor (Bayerisches Geoinstitut, University Bayreuth, 95440 Bayreuth, Germany, Innokenty.Kantor@uni-bayreuth.de), Leonid S. Dubrovinsky (Bayerisches Geoinstitut, University Bayreuth, 95440 Bayreuth, Germany, Leonid.Dubrovinsky@uni-bayreuth.de), Hans Josef Reichmann (Geoforschungszentrum Potsdam, Telegrafenberg, Division 4, 14473 Potsdam, Germany, hanni@gfz-potsdam.de)

High-frequency acoustic interferometry is widely used to penetrate a medium and measure the reflection signature, which can reveal details about the inner structure of the medium. It is a very helpful and one of the most accurate techniques for determination elastic properties of different materials being capable to measure sound wave velocities in very small samples under high pressures. The ultrasonic interferometry system operating at 0.6-2.1 gigahertz (GHz) frequencies was developed in the Bavarian Geoinstitute of the University of Bayreuth for in situ high pressure and temperature experiments. High pressures are reached by using diamond anvil cell, and a Pt-resistive heater allows reaching high temperatures. The experimental setup is equipped with a laser system, which allows measuring a shift of ruby fluorescence line at every given temperature.

3:40

5pPAb2. Estimation of thermophysical properties of fluids under high pressure from speed of sound measurements. Jean Luc Daridon (Université de Pau et des Pays de l'Adour, Laboratoire des Fluides Complexes, UMR 5150, BP 1155, 64000 Pau, France, jean-luc.daridon@univ-pau.fr)

The thermophysical properties of pure substances in fluid state as functions of temperature and pressure are of great interest not only for industrial applications (for example in the oil and gas industry), but also for fundamental aspects in view of developing models for an accurate representation of dense fluids. Now these measurements are difficult to perform under pressure, particularly for non single-phase systems, at atmospheric pressure. An interesting alternative consists in using the ultrasonic velocity which can be determined experimentally with a high degree of accuracy including at high pressures and high temperatures, and which presents the advantage of giving access to various derived properties. This potential, which is specific to ultrasonic velocity in fluids, has been the starting point for the investigation of a large number of pure liquids and gases as well as of several types of mixtures by ultrasonic measurements. In this work we will review the procedures used to evaluate thermophysical properties as a function of pressure from speed of sound measurements. The validity of the different approaches will be checked by comparison to several thermophysical properties measured in an extended pressure range. The accuracy reached for each property by the different procedures will be bring out.

Contributed Papers

4:00

5pPAb3. New results on the sound velocity measurements under extreme conditions using time-resolved picosecond acoustic technique. Frédéric Decremps (IMPMC, Université Paris VI, 140, rue de Lourmel, 75015 Paris, France, frederic.decremps@impmc.jussieu.fr), Laurent Belliard (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, lbelliar@ccr.jussieu.fr), Bernard Perrin (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, bernard.perrin@insp.jussieu.fr), Michel Gauthier (IMPMC, Université Paris VI, 140, rue de Lourmel, 75015 Paris, France, michel.gauthier@impmc.jussieu.fr)

In this presentation, recent works on the pressure and temperature dependence of the sound velocity will be discussed. We have used a newly developed method combining the time-resolved picosecond optical technique and a diamond anvil cell [1]. This setup makes possible accurate measurements of the attenuation and velocity of longitudinal waves in the Ghz range, and opens the elastic investigations of all materials (opaque, transparent, single- or polycrystal, liquids) up to several Mbar and thousands of K. The experimental method will be first described, with a discussion of the factors limiting the possibilities and the technique accuracy. [1] F. Decremps, L. Belliard, B. Perrin, and M. Gauthier, Phys. Rev. Lett, to be pubished in January 2008.

4:20-4:40 Break

4:40

5pPAb4. Elastic moduli at high temperatures with two different ultrasonic methods. Ludivine Bourgeois (Commissariat à l'Energie Atomique (CEA), Centre de Valduc, 21120 Is sur Tille, France, ludivine.bourgeois@cea.fr)

"The elastic moduli and specially the shear modulus G is one of the parameters needed to implement the constitutive relations. These mechanical

properties can be measured for a solid or a liquid with different ultrasonic methods. This paper deals with the determination of G(T) with two different methods: a contact delay-line ultrasonic device which has been developed to measure the shear and compression waves velocities up to 1000 K on cylindrical specimens and a contactless (laser) ultrasonic device which can measure surface acoustic waves velocities in metals up to the melting point. Results on metals such as Al without phase transition and for Co through the phase transition will be presented."

5:00

5pPAb5. Measurements under high pressure of ultrasonic velocity in glycerol. Hassina Khelladi (Faculty of Physics, University of Science and Technology Houari Boumedienne, BP 32 El Allia, Bab-Ezzouar, 16000 Algiers, Algeria, hassinakhelladi@yahoo.fr), Frédéric Plantier (Université de Pau et des Pays de l'Adour, Laboratoire des Fluides Complexes, UMR 5150, BP 1155, 64000 Pau, France, frederic.plantier@univ-pau.fr), Jean Luc Daridon (Université de Pau et des Pays de l'Adour, Laboratoire des Fluides Complexes, UMR 5150, BP 1155, 64000 Pau, France, jean-luc.daridon@univ-pau.fr), Hakim Djelouah (Faculty of Physics, University of Science and Technology Houari Boumedienne, BP 32 El Allia, Bab-Ezzouar, 16000 Algiers, Algeria, djelouah_hakim@yahoo.fr)

Glycerol has been the subject of significant scientist interest. Indeed, glycerol is a polyalcohol and the presence of three hydroxyl groups per molecule makes glycerol a complex system to explore. The purpose of this investigation is to measure under high pressure the ultrasonic wave velocity in glycerol, from which a number of important thermodynamic properties could be derived and determined as a function of pressure and temperature. Pressure and temperature ranges exploited in this experimental investigation of various glycerol properties, are extended respectively from 0.1 MPa to 100 MPa and from 10 °C to 100 °C. A high pressure measurement cell equipped with temperature and pressure monitoring and control instrumentation was used. A time of flight method was exploited to measure, under high pressure, the ultrasonic wave velocity at different temperatures. The resulting experimental data of ultrasonic wave velocity in glycerol combined with measurements at atmospheric pressure, of density, specific heat and thermal expansion coefficient were used to derive density at elevated pressures. As isentropic compressibility is linked to ultrasonic wave velocity and density by means of the Newton-Laplace equation, this intrinsic physical property is easily deduced. These results led to the behavior of each physical property as a function of temperature and pressure.

5:20

5pPAb6. Laser optoacoustic study of near-critical states and phase transitions in metals. Alexander Y. Ivochkin (Moscow State University International Laser Centre, Leninskie gory, 1, 119992 Moscow, Russian Federation, ivochkin@yandex.ru), Alexander G. Kaptilniy (Joint Institute for High Temperatures, Russian Academy of Science, Izhorskaya str., 13/19, 125412 Moscow, Russian Federation, drc@pochta.ru), Alexander Karabutov (Moscow State University, MSU, 1, building 2, GSP-2, Leninskiye Gory, 119992 Moscow, Russian Federation, akarabutov@gmail.com)

Pulsed laser optoacoustic technique is used for generation and study of near-critical states and phase transitions in metals. Metal surface is confined by a layer of transparent dielectric. In this case the efficiency of pressure generation is much greater than in case of the free surface so it is possible to achieve states of metals with relatively high thermodynamic parameters: P $\sim 10^4$ atm and T ~ 10^4 K with a table-top laser system. The experimental setup for simultaneous measurements of pressure, temperature, and reflectivity of metal surface with nanosecond temporal resolution was assembled. Q-switched Nd:YAG laser with pulse duration ~ $\sim$10 ns and pulse energy $\sim$1 J was used. Pressure was measured using $LiNbO_3$ piezotransducer. Temperature was obtained with optical pyrometer. Lead and mercury were chosen as test metals. Pressure pulses up to 1 kbar in lead and up to 7 kbar in mercury (with T $\sim$2400 K - super-critical area of the phase diagram for Hg) were obtained. The curve of laser heating process in P-T coordinates was plotted. The measurements of optical properties showed considerable decrease of surface reflectivity both for lead and mercury at high laser fluences due to increase of temperature and density decrease.

5:40

5pPAb7. Effect of intense neutron dose radiation on piezoceramics. Franck P. Augereau (IES/Université Montpellier II, Université Montpellier II, Place Eugène Bataillon, 34095 Montpellier, France, Franck.Augereau@ies.univ-montp2.fr), Jean-Yves Ferrandis (Radio Application Division, NEC Corporation, Université Montpellier II, Place Eugène Bataillon, 34095 Montpellier, France, ferrandi@lain.univ-montp2.fr), Jean-François Villard (CEA Saclay, 91191 Gif sur Yvette Cedex, France, jean-francois.villard@cea.fr), Damien Fourmentel (CEA Saclay, 91191 Gif sur Yvette Cedex, France, damien.fourmentel@cea.fr), Mark Dierckx (SCK-CEN, Boeretang 200, B-2400 Mol, Belgium, mdierckx@sckcen.be), Jan Wagemans (SCK-CEN, Boeretang 200, B-2400 Mol, Belgium, jwageman@SCKCEN.BE)

Four grades of commercial PZT materials have been exposed to nuclear radiation during five months in an irradiation channel of the BR1 research reactor at SCK·CEN (Belgium). This experimental study was performed in the framework of the Joint Instrumentation Laboratory with the CEA French Commission of Atomic Energy to validate these materials for future applications in severe conditions such as online measurements in irradiation experiments performed in research reactors. For this purpose, thin piezoelectric discs were irradiated while a remote network analyser continuously monitored the frequency response of their electrical impedance. The total neutron dose has reached a level of 1.5.1017 neutrons/cm^2. Positive and negative shifts of the peak resonance frequency have been recorded but in any case with a variation lower than 1%. On the other hand, the amplitude of the electrical impedance at resonance frequency has largely decreased with even a reduction by factor two or three for some piezoelectric cells. Transitory effects have also been detected for these two parameters as function of the reactor activity. Additional thermal and gamma radiation effect have been investigated. Similarly, some piezoelectric cells glued on glass delay line have been tested with satisfactory results to these stresses.

6:00

5pPAb8. Ultrasonic exploration at extreme shallow underground in submerged soil. Kunihiko Seo (Toin University of Yokohama, 1614 Kurogane-cho, Aoba-ku, 225-8502 Yokohama, Japan, seo.ylk@gmail.com), Takashi Shirakawa (Toin University of Yokohama, 1614 Kurogane-cho, Aoba-ku, 225-8502 Yokohama, Japan, t_shira7@yahoo.co.jp), Tsuneyoshi Sugimoto (Toin University of Yokohama, 1614 Kurogane-cho, Aoba-ku, 225-8502 Yokohama, Japan, tsugimot @cc.toin.ac.jp)

Now a lot of land mines remain buried in the world, so that the clearance of them is required. As a tool of removing land mines, equipments using electromagnetic radiation are often employed. However, there is a problem that the land mines in the flooded soil such as in Southeast Asia cannot be detected in the rainy season. Therefore, the new way using sound waves will be profitable to detect the land mines in the flooded soil. In this research, the acoustic exploration at very shallow area in submerged sand is examined at a water tank in the lab. First we measure the propagation property of ultrasound of 120 kHz in the shallow submerged sand, and examine underground imaging. As a result, the acoustic velocity is measured at about 1500 m/s and the attenuation is measured at about -19 dB/m. And next, shallow underground exploring by using acoustic shielding boards is carried out. As a result, underground images in the water tank simulating the submerged soil are obtained. Then acoustic shielding boards can block wave which propagates specific route. This will make another exploration method possible.

Session 5pPAc

Physical Acoustics: General Topics in Physical Acoustics II

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Andi Petsculescu, Cochair
University of Louisiana, Department of Physics, P.O. Box 44210, Lafayette, LA 70504, USA

Contributed Papers

4:00

5pPAc1. An automated 3 dimensional scanning system for validation of acoustical simulation results. Simo-Pekka Simonaho (University of Kuopio, P.O.Box 1627, 70211 Kuopio, Finland, simo-pekka.simonaho@uku.fi)

To validate acoustical simulation results, a great number of measurement points are needed especially in 3 dimensional cases. These measurements can be extremely laborious when done manually. Also, the spatial information of the measurement points has to be accurate. In this work, an automated 3-D scanning system with high spatial resolution for validation of acoustical simulation results is introduced. The system consists of a multi channel data acquisition hardware, a microphone array and a 3-D scanning system. The movement of the microphone array is controlled by the data acquisition hardware. The performance of the automated 3-D scanning system is demonstrated and the experimentally measured pressure fields are compared to simulation results.

4:20

5pPAc2. Signal processing of impedance spectrum for speed of sound and pressure measurement in plane or radial resonators. Eric Rosenkrantz (Radio Application Division, NEC Corporation, Université Montpellier II, Place Eugène Bataillon, 34095 Montpellier, France, eric.rosenkrantz@ies.univ-montp2.fr), Jean-Yves Ferrandis (Radio Application Division, NEC Corporation, Université Montpellier II, Place Eugène Bataillon, 34095 Montpellier, France, ferrandi@lain.univ-montp2.fr), Gerard Leveque (Institut d'Electronique du Sud UMR-CNRS 5214, Université Montpellier II, Place Eugène Bataillon, 34095 Montpellier, France, Gerard.leveque@ies.univ-montp2.fr)

The impedance of gas contained between two plane walls is a periodic function of the frequency. The frequency interval between two resonances of the gas is equal to c/2D. Furthermore, the amplitude of the resonances is proportional to the pressure. We describe a signal processing to easily deduce the speed of sound and the pressure of the gas from the impedance spectrum. We show that the modulus of the Fourier transform of the modulus of impedance, called "Tempograph," contains all information about the gas. In some industrial cases the container is a cylinder or a sphere which can be used as a radial resonator exited by a radial wave [1, 2]. The frequency response of such resonator at high frequencies is quasi-periodic and thus the same signal processing can be used. [1] M. F. Narbey, et al., "Determination of the composition of a gas mixture in a nuclear fuel rod by an acoustic method." INSIGHT, **42**(9), 603-605 (2000). [2] A. Olson, "Helium bottle pressure measurement by portable ultrasonic technique." 1989, Rapport de Boeing n°AD-A208 994, http://stinet.dtic.mil/str/index.html.

4:40

5pPAc3. Absorption and velocity of acoustical waves in binary solutions of poly (ethylene glycol) and water. Rajendra Kumar Singh (Department of Physics, Banaras Hindu University, 221005 Varanasi, India, rksingh_17@rediffmail.com), Manish Pratap Singh (Banaras Hindu University, 221005 Varanasi, India, mps_bhu@yahoo.co.in),Rishi Pal Singh (Banaras Hindu University, 221005 Varanasi, India, rishisingh80@rediffmail.com)

A resonator technique has been devoleped to measure ultrasonic velocity and absorption for aqueous solutions of PEG of different molecular weights. The velocity has been measured at different frequencies and the concentrations (by weight) ranged from 1% to 10% of poly (ethylene glycol) in water. Adiabatic compressibility has been obtained at different temperatures, using experimental value of velocity and density. Viscocity has also been measured in wide temperature range . Ultrasonic absorption has been measured in the frequency range 400 KHz-50 MHz, using resonator technique and pulse technique in temperature range 400-650 °C. Observations showed that the ultrasonic absorption decreases with increasing temperature at a given concentration and also increases with concentration at a given temperature. The velocity increases with increasing temperature and concentration. Shear viscosity has been found to decrease with temperature but increases with concentration. Velocity studies show that as the polymer concentration increases a more rigid molecular structure is formed by bonding between the large polymer molecules.

5:00

5pPAc4. Acoustic field in a spherical resonator: effect of modal coupling due to small perturbations. Cécile Guianvarc'H (Institut National de Métrologie (LNE-INM/Cnam), 61 rue du Landy, 93210 La Plaine Saint Denis, France, cecile.guianvarch@cnam.fr), Laurent Pitre (Institut National de Métrologie (LNE-INM/Cnam), 61 rue du Landy, 93210 La Plaine Saint Denis, France, pitre@cnam.fr), Arnaud Guillou (Institut National de Métrologie (LNE-INM/Cnam), 61 rue du Landy, 93210 La Plaine Saint Denis, France, guillou.arnaud@gmail.com), Michel Bruneau (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, michel.bruneau@univ-lemans.fr), Anne-Marie Bruneau (Laboratoire d'Acoustique de l'Université du Maine (LAUM, UMR CNRS 6613), Avenue Olivier Messiaen, 72085 Le Mans, France, anne-marie.bruneau@univ-lemans.fr)

The international community recently recommended a re-definition of the kelvin referring to the value of the Boltzmann constant *k*, which must thus be known with a relative uncertainty of 10^{-6}. The measurement of the acoustic resonance properties of a gas filled spherical or quasi-spherical resonator is an appropriate method to do this with these requirements. Actually, a detailed modeling of the acoustic field in the resonator is required for the determination of *k*. Several phenomena must be taken into account including heat conduction, shear and bulk viscosity of the gas, the real shape of the resonator, the acoustic input impedance of small acoustic elements flush-mounted on the wall (tubes, transducers)... . Significant theoretical studies have already been done in which these perturbations are accounted for separately, the coupling between them being neglected. The scope here is thus to provide a unified model for the acoustic field in the cavity including all these perturbations and the resulting modal coupling, and applying it on a simple practical configuration: a spherical resonator filled with argon, acoustic transducers being flush-mounted on the wall.

5:20

5pPAc5. Towards a theory for arbitrarily shaped sound field reproduction systems. Sascha Spors (Deutsche Telekom Laboratories, Ernst-Reuter-Platz 7, 10587 Berlin, Germany, sascha.spors@telekom.de), Jens Ahrens (Deutsche Telekom Laboratories, Ernst-Reuter-Platz 7, 10587 Berlin, Germany, jens.ahrens@telekom.de)

The simple source approach predicts that a distribution of appropriately driven loudspeakers (secondary sources) enclosing a given listening area is suitable for the physical recreation of any desired exterior virtual sound field within that listening area. A specific class of sound reproduction approaches (e.g., higher-order Ambisonics) is based upon the explicit solution of the simple source formulation with respect to the secondary source driving function. To the knowledge of the authors, such an explicit solution is currently only available for specific geometries of the sound reproduction system. This contribution presents a theoretical framework for the derivation of the explicit solution for the driving function of arbitrarily shaped secondary source arrangements. It is based upon the expansion of the reproduced wave field into a series of orthogonal basis functions. These basis functions emerge from the respective underlying geometry. It is shown that most of the sound reproduction systems that are based upon the simple source formulation can be seen as specialization of the presented approach to a particular geometry and basis function.

5:40

5pPAc6. Numerical computation of reflected and transmitted waves at a fluid/solid interface. Laure Bossy (AREVA - CEZUS Research Center, Avenue Paul Girod, 73403 Ugine Cedex, France, laure.bossy@espci.fr), Marie-Françoise Cugnet (AREVA - CEZUS Research Center, Avenue Paul Girod, 73403 Ugine Cedex, France, mariefrancoise.cugnet@areva.com), Emmanuel Bossy (Laboratoire Photons et Matière, ESPCI/CNRS, 10 rue Vauquelin, 75231 Paris Cedex 05, France, emmanuel.bossy@espci.fr), Didier Cassereau (Laboratoire Ondes et Acoustique, 10, rue Vauquelin, 75231 Paris, France, didier.cassereau @espci.fr)

In this paper, we propose a numerical computation of the different waves generated when a spherical incident pulse is reflected and transmitted by a fluid/solid interface. In addition to the standard reflected and transmitted waves that propagate inside the volume, various surface waves can also be found in both propagation media. In the fluid, we can observe the longitudinal and transverse head waves, and the so-called leaky Rayleigh wave that generalizes the Rayleigh wave in a semi-infinite free solid medium to the case of an immersed interface. Similar effects can also be observed on the transmitted displacement field inside the solid material. We compare different numerical approaches, including semianalytic methods (high-frequency approximation coupled to ray modeling approach) and implicit methods (finite elements and/or finite differences scheme), each method having its own advantages and inconvenients, and domains of validity. These different methods are used to evaluate the field reflected by the interface ; the transmitted displacement field is also analyzed from the same point of view. The plane and curved geometries will be analyzed and the influence of the curvature of the interface will be pointed out.

6:00

5pPAc7. V(z) oscillations in acoustic microscope at upward defocusing. Anton V. Kozlov (MSU, Leninskie Gory, Bld. 1-2, 119991 GSP-1 Moscow, Russian Federation, av_kozlov@inbox.ru), Vladimir G. Mozhaev (MSU, Leninskie Gory, Bld. 1-2, 119991 GSP-1 Moscow, Russian Federation, vgmozhaev@mail.ru)

Acoustic microscopy is widely used for imaging and study of elastic properties of transparent and opaque materials. As a rule, multiple periodic oscillations in the output signal *V* of a reflection acoustic microscope can be observed in the case of negative defocusing of the wide-angle acoustic lens, i.e., when its focus is a distance *z* below the solid sample surface. The main well-known mechanism for *V(z)* oscillations is the interference of radiation reflected perpendicularly from the sample surface and re-radiation of leaky Rayleigh waves generated on the sample by the lens. This effect explains high contrast imaging in reflection scanning acoustic microscopy, and it is a popular method to study properties of solids by measuring Rayleigh wave speeds. As it is shown in this work, bulk acoustic waves in the sample can also give rise to *V(z)* oscillations. A new mechanism of such oscillations is predicted in the case of positive defocusing (focusing above the sample surface) in acoustic microscopy of anisotropic plates exhibiting negative acoustic refraction. The ray model of this effect shows a possibility to find a relationship between extrema of *V(z)* curve and separate points on the acoustic slowness surface of the sample.

FRIDAY AFTERNOON, 4 JULY 2008 — ROOM 351, 5:00 TO 6:20 P.M.

Session 5pPAd

Physical Acoustics: Scattering and Diffraction II

Michael L. Oelze, Cochair
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Contributed Papers

5:00

5pPAd1. The physics of wedge diffraction: A model in terms of elementary diffracted waves. Mitsuhiro Ueda (Predio Meguro Science Laboratory, 4-20-13 Meguro, Meguro-ku, 153-0063 Tokyo, Japan, ueda-mt @nifty.com)

A model for wedge diffraction is constructed using the virtual discontinuity principle of diffraction [1]. In the model diffracted waves are described by the sum of two elementary diffracted waves that are calculated by integrating the potential along the half line issued from the vertex of wedge. The wedge of aperture angle π/n (n=1,2,3,...) is nondiffractive since its potential can be expressed by the sum of direct waves from the point source and its mirror images. The nondiffractive wedges are useless in the conventional analysis of diffraction since there are no diffracted waves in the potential. But in this model diffracted waves of these wedges are cancelled out in the summing process. Thus the elementary diffracted waves are existed even in these wedges and far field solution for them is obtained for the nondiffractive wedge. Due to its simple structure it can be extended to the arbitrary wedge without any modification and the far field solution of diffracted waves is derived by summing the extended elementary diffracted waves in the model and it coincides with the rigorous one literally. Thus the model is verified firmly by this simple calculation. 1. M. Ueda, JASA, 95, p.2354 (1994).

5:20

5pPAd2. A global search tool for the equivalent source method and its applications to the scattering problem. Yves J.r. Gounot (UFRJ/COPPE, Universidade Federal do Rio de Janeiro, 21941-972 Rio de Janeiro, Brazil, ygounot@mecanica.ufrj.br), Ricardo E. Musafir (UFRJ/COPPE, Universidade Federal do Rio de Janeiro, 21941-972 Rio de Janeiro, Brazil, rem @serv.com.ufrj.br)

Low computational-cost solutions to the acoustic scattering problem can be obtained with the equivalent source method (ESM), provided the sources are adequately positioned. Because this last point represents often a complicated task - mainly responsible for the not much widespread use of the method - a technique that hurdles this difficulty, called ESGA, has been previously proposed (Gounot and Musafir, Internoise 2004). Based on a combination of genetic algorithm with ESM, the ESGA is a global search tool that provides, given a set of monopoles, their 'optimal' positioning and complex amplitudes. The technique efficiency is here shown through a number of three-dimensional scattering problems. The algorithm is also used in order to identify, for each of the different cases considered, typical geometrical arrangements of monopoles which provide good solutions.

5:40

5pPAd3. Convergence of correlations in multiply scattering media. Eric Larose (LGIT - CNRS - Université Joseph Fourier, Maison des Géosciences, 1381 rue de la Piscine, BP 53, 38041 Grenoble, France, Eric.Larose@obs.ujf-grenoble.fr), Arnaud Derode (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, arnaud.derode@ujf-grenoble.fr), Philippe Roux (LGIT - CNRS - Université Joseph Fourier, Maison des Géosciences, 1381 rue de la Piscine, BP 53, 38041 Grenoble, France, philippe.roux@obs.ujf-grenoble.fr), Michel Campillo (LGIT - CNRS - Université Joseph Fourier, Maison des Géosciences, 1381 rue de la Piscine, BP 53, 38041 Grenoble, France, Michel.Campillo@obs.ujf-grenoble.fr)

Correlations of ambient seismic or acoustic noise are now widely used to reconstruct the impulse response between two passive receivers as if a source was placed at on of them. Applications include terrestrial and solar seismology, underwater acoustics and structural health monitoring. Nevertheless, for a given set of data, correlations do not only yield automatically the Green function between the sensors, but also contains residual fluctuations that might ″blur″ the images. [Gizon et al, Astrophys. J. 614 (2004); Weaver and Lobkis, J. Acoust. Soc. Am. **117** (2005); Sabra et al., J. Acoust. Soc. Am. 118 (2005)]. We propose a model to describe the "signal-to-fluctuations" ratio in the correlations in the case of nonstationary wavefields, and more particularly in the case of scattering media. The work includes theoretical derivations and numerical simulations. The role of multiple scattering in the rate of convergence of the correlations toward the Green function is quantitatively evaluated [Larose et al, (submitted 2008)].

6:00

5pPAd4. Reflection and transmission coefficients of a fluid slablike region containing a depth-varying random distribution of cylinders. Jean-Marc Conoir (Institut Jean Le Rond d'Alembert-UMR CNRS 7190, Université Paris 6, tour 55-65, 4 place Jussieu, 75005 Paris, France, conoir@lmm.jussieu.fr), Sébastien Robert (LOA, UMR CNRS 7587, ESPCI, 10 rue Vauquelin, 75231 Paris, France, Sebastien.Robert@espci.fr), Abdelhak El Mouhtadi (LOMC- FRE CNRS 3102- Groupe Ondes Acoustiques, Université du Havre, place R. Schuman, 76610 Le Havre, France, abdelhak.ilel@gmail.com), Francine Luppé (LOMC- FRE CNRS 3102- Groupe Ondes Acoustiques, Université du Havre, place R. Schuman, 76610 Le Havre, France, francine.luppe @univ-lehavre.fr)

This work deals with multiple scattering by a random distribution of parallel elastic cylinders immersed in a fluid slablike region. The concentration of scatterers inside the slab is supposed to vary slowly with depth, and the WKB method is used to calculate the reflection and transmission coefficients of the slab. In order to do so, the continuity conditions on the boundaries between the slab and the surrounding fluid are needed. They follow from the application of Twersky's theory to the case of a slab with a given constant concentration of scatterers, which shows that both pressure and normal displacement are continuous, provided an effective mass density of the slab is correctly defined. The results of the WKB are successfully compared to those obtained from the discretization of the slab into layers of constant concentrations of cylinders and the use of Twersky's theory.

FRIDAY AFTERNOON, 4 JULY 2008 ROOM 241, 2:00 TO 6:20 P.M.

Session 5pPPa

Psychological and Physiological Acoustics and Speech Communication: Acoustic Features and Speech Perception II

Jont B. Allen, Cochair
University of IL, 405 N. Mathews, Room 2061 Beckman Inst. (MC 251), Urbana, IL 61801, USA

Sarah Hawkins, Cochair
University of Cambridge, Department of Linguistics, Sidgwick Avenue, Cambridge, CB3 9DA, UK

Invited Papers

2:00

5pPPa1. Reception of phonetic features in fluctuating background noise maskers. Christian Lorenzi (Univ Paris Descartes, CNRS, Ecole Normale Superieure, DEC, 29 rue d'Ulm, 75005 Paris, France, lorenzi@ens.fr)

We will review recent studies investigating the reception of phonetic features (voicing, manner and place of articulation) in nonstationary background maskers. In each study, consonant identification was assessed in steady and amplitude-modulated speech-shaped noise at signal-to-noise ratios yielding about 50% correct in steady noise. The rate and depth of amplitude modulation applied to the noise masker were either fixed or systematically varied. Confusion matrices were compiled across listeners and the amount of release from masking (percent information received in nonsteady minus steady noise) was calculated for each phonetic feature and experimental condition. Speech and noise mixtures were (i) left intact (unfiltered), (ii) lowpass filtered (<1.5 kHz), or iii) processed in order to degrade spectral (place of excitation in the cochlea) and/or temporal fine structure cues. Overall, the results indicate that release from masking typically reported in normal-hearing listeners (i.e., the substantial improvement in speech reception in fluctuating noise compared to steady noise) is not identical across phonetic features. This suggests that the ability to "glimpse" into background noise valleys involves multiple auditory processes constrained by both peripheral and central factors.

2:20

5pPPa2. Identification of intervocalic consonants in stationary and nonstationary noise. Martin Cooke (Sheffield University, Computer Science Department, Regent Court, 211 Portobello St., S1 4DP Sheffield, UK, m.cooke@dcs.shef.ac.uk), Odette Scharenborg (Centre for Language and Speech Technology, Radboud University Nijmegen, Erasmusplein 1, 6525 HT Nijmegen, Netherlands, O.Scharenborg@let.ru.nl)

The factors which underlie the perception of consonants in noise remain poorly understood. In this study, native listeners identified 24 English consonants spoken by eight talkers presented in nine intervocalic contexts with varying stress position. Listeners were tested in 5 noise conditions: tokens were masked by stationary speech-shaped noise, a competing talker, three and eight speaker babble and speech-modulated noise, all of which have the long-term spectrum of speech. The rank ordering of consonant identification scores in stationary noise was highly-correlated ($r=0.9$, $p<0.0001$) with a similar condition reported by Phatak and Allen [JASA **121**: 2312-2326, 2007], but less so in the four nonstationary noise backgrounds ($r=0.74$, $p<0.001$). In particular, /y/, /r/, /l/, /f/, /ch/, /sh/, /m/ and most of the plosives showed a wide variation in ranking. These findings suggest that, in addition to the long-term spectrum of the masker, consonant identification is noise is affected by other factors such as temporal fluctuations in the masker, misallocation of foreground-/background components and attention.

2:40

5pPPa3. The interaction of glimpsing, pitch and vocal tract length in the recognition of concurrent syllables. Martin D. Vestergaard (Centre for the Neural Basis of Hearing, Department of Physiology, Development and Neuroscience, University of Cambridge, Downing Site, CB23EG Cambridge, UK, mdv23@cam.ac.uk), Nicolas R. Fyson (Centre for the Neural Basis of Hearing, Department of Physiology, Development and Neuroscience, University of Cambridge, Downing Site, CB23EG Cambridge, UK, nickfyson@gmail.com), Roy D. Patterson (Centre for the Neural Basis of Hearing, Department of Physiology, Development and Neuroscience, University of Cambridge, Downing Site, CB23EG Cambridge, UK, rdp1@cam.ac.uk)

In multispeaker environments, human listeners use the temporal misalignment of competing speech signals to improve recognition-an effect referred to as "glimpsing." When the temporal envelopes of concurrent syllables pairs are carefully matched to preclude glimpsing, listeners were observed to use vocal tract length (VTL) and glottal pulse rate (GPR) cues to improve recognition. This paper reports an investigation of the interaction between glimpsing and these vocal cues. Syllables were synthesized with a vocoder to simulate speakers with widely different combinations of GPR and VTL. Recognition of one syllable in the presence of a concurrent syllable was measured as the vocal cues and the temporal alignment of the syllables were varied. The effect of glimpsing was most pronounced when the vocal cue differences between the target and distracter syllables were small. Furthermore, there was a strong effect of consonant type (stops, fricatives, or sonorants) and an asymmetry between consonant-vowel (CV) and vowel-consonant (VC) syllables. The lowest recognition rate was observed, not at perfect temporal alignment but rather at a distracter lag of -50 ms for CVs and 100 ms for VCs. The results are analyzed with confusion matrices. Research supported by the UK MRC [G0500221, G9900369].

3:00

5pPPa4. Phoneme confusions as a function of noise, spectral resolution and L2 experience. Robert Shannon (House Ear Institute, 2100 W. Third St., Los Angeles, CA 90057, USA, shannon@hei.org), Monica Padilla (House Ear Institute, 2100 W. Third St., Los Angeles, CA 90057, USA, mpadilla@hei.org)

Consonant and vowel confusion matrices were measured from normal hearing listeners with varying degrees of experience in English. There were five listeners each in the following categories: native English speakers, and Spanish speakers who were immersed in English at the ages of 0-5 years, 5-10 years, 10-18 years, and over 18 years. 12 vowels (hVd) or 18 consonants (vCv) were presented and responses were collected in a confusion matrix. Spectral resolution was varied by using a noise-band vocoder with 2, 4, 6, 8, and 16 channels, as well as unprocessed speech. All stimuli were presented in speech shaped noise at SNR levels of -5 dB to +15 dB in 5 dB steps as well as in quiet. Noise and spectral resolution had a similar effect on voicing, manner and place of articulation, and also had similar effects as a function of L2 experience. For native speakers of Spanish the duration of L2 experience had the largest effect on voicing cues, less effect on manner cues, and almost no effect on place cues.

3:20

5pPPa5. The perceptual flow of phonetic feature processing. Steven Greenberg (Technical University of Denmark, Center for Applied Hearing Research, Ørsteds Plads, Building 352, 2800 Lyngby, Denmark, steveng@silicon-speech.com), Thomas Ulrich Christiansen (Technical University of Denmark, Center for Applied Hearing Research, Ørsteds Plads, Building 352, 2800 Lyngby, Denmark, tuc@oersted.dtu.dk)

How does the brain process spoken language? It is our thesis that word intelligibility and consonant identification are insufficient by themselves to model how the speech signal is decoded - a finer-grained approach is required. In this study, listeners identified 11 different Danish consonants spoken in a Consonant + Vowel + [l] environment. Each syllable was processed so that only a portion of the original audio spectrum was present. Three-quarter-octave bands of speech, centered at 750, 1500, and 3000 Hz, were presented individually and in combination with each other. The conditional, posterior probabilities associated with phonetic-feature decoding were computed from confusion matrices in order to deduce the temporal flow of phonetic processing. Decoding the feature, Manner-of-Articulation, depends on accurate decoding of the feature Voicing (but not vice-versa), and decoding Place-of-Articulation requires

precise decoding of Manner (but not the converse). From these data, we conclude that Voicing is processed prior to Manner-of-Articulation, and that Manner is decoded prior to Place-of-Articulation. Voicing and Manner cues are often correctly decoded in conditions where Place is not. This asymmetric pattern of feature decoding may provide extra-segmental information of utility for speech processing, particularly in adverse listening conditions.

3:40

5pPPa6. Understanding the complex modulation spectrum for consonants and consonant features. Kenneth W. Grant (Walter Reed Army Medical Center, Army Audiology and Speech Center, 6900 Georgia Ave. NW, Washington, DC 20307-5001, USA, grant@tidalwave.net), Sandeep A. Phatak (Walter Reed Army Medical Center, Army Audiology and Speech Center, 6900 Georgia Ave. NW, Washington, DC 20307-5001, USA, s.a.phatak@gmail.com), Elena Grassi (Walter Reed Army Medical Center, Army Audiology and Speech Center, 6900 Georgia Ave. NW, Washington, DC 20307-5001, USA, elena.grassi@gmail.com)

Speech intelligibility is highly dependent on the magnitude and phase characteristics of the low-frequency modulation spectrum. However, unlike more traditional representations of speech, such as the spectrogram, associating details of the modulation spectrum to specific phonemes and subphonemic units of speech has not been readily forthcoming. In the present study we used local time reversals of the speech waveform between 20-160 ms to selectively distort portions of the complex modulation spectrum. Normal-hearing subjects were tested on a consonant recognition task and a detailed analysis of the perceptual confusions was performed. Consistent with earlier results using sentence-length materials, average consonant intelligibility declined as the length of the time reversal segment increased. Further analyses were conducted to determine the effect of time-reversal segment duration on the amount of information transmitted for individual consonants (including specific consonant productions) and acoustic features for voicing, manner of articulation, and place of articulation. An acoustic analysis using a biologically motivated auditory processing model was also performed to determine the effect of time reversals on cochlear and cortical representations of speech. The relations between changes to the complex modulation spectrum and the percent information transmission of selected speech segments and features are discussed.

4:00-4:20 Break

4:20

5pPPa7. Can CV intelligibility predict speech intelligibility? Sarah Hawkins (University of Cambridge, Department of Linguistics, Sidgwick Avenue, CB3 9DA Cambridge, UK, sh110@cam.ac.uk)

This paper begins by reviewing the speech perception literature to predict cues that would and would not be expected to survive energetic masking of various types. The focus is especially (but not exclusively) on spectrotemporal cues to stops in the vicinity of the segment boundary in CV syllables. The second part of the paper discusses influences that can restrict the generality of research findings from isolated CV syllables. This includes ways in which CV syllables change in different phonetic contexts and styles of speech, contributions of the visual modality, and other uses of top-down information, such as phonotactic, lexical, semantic and syntactic probability. The paper concludes by asking whether-and how-simple measures such as CV intelligibility can be used to reflect intelligibility of speech in real-life communicative situations.

4:40

5pPPa8. The role of the cochlear processing in human speech recognition. Jont B. Allen (University of IL, 405 N. Mathews, Room 2061 Beckman Inst. (MC 251), Urbana, IL 61801, USA, jontallen@ieee.org), Marion Regnier (208 S. 3rd St. Apt 5A, Brooklyn, NY 11211, USA, marion.regnier@gmail.com), Sandeep Phatak (Walter Reed Hospital, Silver Springs, MD 20901, USA, s.a.phatak@gmail.com), Feipeng Li (University of IL, 405 N. Mathews, Room 2061 Beckman Inst. (MC 251), Urbana, IL 61801, USA, fli2@uiuc.edu)

Little is know about how the auditory system decodes speech. We may think of speech communication re Shannon's source-channel model, thus viewed, the most complex part of the speech communication channel is the auditory system (the receiver). In my speech-perception research, I have attempted to limit the assumptions, and have thus fallen back on Shannon's basic source-channel model. The basic tool is the confusion matrix (CM) for isolated natural consonant and vowels (CV), as a function of the speech to noise ratio (SNR), with several types of masking noise. We have used large numbers of talkers and listeners (i.e., 20). In a second experiment we selectively remove islands of speech in time-frequency, and then correlate the resulting modified speech against subject scores. Our most important conclusions are: (1) The across-frequency onset transient portion of the signal is typically the most important. (2) The spectral regions of these transient are used to code different consonants. (3) While the frequency regions for a given consonant are correlated to the following vowel, this may not be important for perception. (4) Compact spectral-temporal amplitude modulations components (e.g., a 10 Hz modulation) do not seem to play a significant role, at least above 1-2 kHz.

Contributed Papers

5:00

5pPPa9. Spectral and temporal modulations essential to spoken word, gender and timbre identification. Frédéric E. Theunissen (UC Berkeley, Dept. of Psychology, 3210 Tolmant Hall, Berkeley, CA 94720-1650, USA, theunissen@berkeley.edu), Taffeta Elliott (UC Berkeley, Dept. of Psychology, 3210 Tolmant Hall, Berkeley, CA 94720-1650, USA, taffeta@berkeley.edu)

Human speech and musical sounds contain complex spectral and temporal modulations. Speech intelligibility, perception of melody, and identification of source characteristics (e.g., speaker gender or musical timbre) depend on spectrotemporal modulations but can be surprisingly robust to drastic spectral and temporal degradations. We systematically explored which restricted spectral and temporal modulations are essential to the perception of complex sounds. Degraded sentences and musical sounds were obtained by a novel modulation filtering procedure performed on the sound spectrogram. Temporal modulation filtering smeared the amplitude envelope by removing changes above particular Hz. Spectral modulation filtering smeared the spectral energy across frequency bands by removing changes above particular cyc/kHz. We further complemented this low-pass filtering with more specific notch-filtering. Speech intelligibility, gender recognition

and musical instrument identification were assessed in psychophysical experiments. We determined that spectral modulations below ~3.75 cyc/kHz, and temporal modulations between 1 and 7 Hz are essential for speech comprehension. Gender identification however required the presence of higher spectral modulations. Similarly the timbre and pitch of instruments was affected differentially by notch filters in these two regions of the modulation spectrum. Our research could be used to guide the design of optimal signal processing in hearing aids and cochlear implants.

5:20

5pPPa10. A model with compression for estimating speech intelligibility in quiet and 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), Johannes Lyzenga (Vrije Universiteit Medical Center, Boelelaan 1117, 1081 HV Amsterdam, Netherlands, j.lyzenga@vumc.nl)

For speech reception thresholds (SRTs), measured in normally-hearing listeners using various types of stationary noise, the Speech Intelligibility Index (SII, ANSI S3.5-1997) model predicts a fairly constant speech proportion (of about 0.3) necessary for sentence intelligibility. For SRTs in quiet, the estimated speech proportions are often lower, and show a larger inter-subject variability, than found for speech in noise near normal speech levels. This might be related to the fact that cochlear compression is larger at normal speech levels than near the threshold for speech in quiet. The SII model does not take this into account. The present model attempts to alleviate this problem by including cochlear compression. It is based on a loudness model for normally-hearing and hearing-impaired listeners [ANSI S3 .4-2007]. It estimates internal excitation levels of the speech, accounts for the compressed effective dynamic range of the internal speech signal, and calculates the proportion of speech above threshold using similar spectral weighting as used in the standard SII. The present model and the standard SII were used to predict SRTs in quiet and noise for both normally-hearing and hearing-impaired listeners. The present model predicted speech intelligibility with less variability than the standard SII.

5:40

5pPPa11. Listeners' sensitivity to talker differences in voice-onset-time: Segments versus features. Rachel M. Theodore (Northeastern University, Dept. of Psych. - 125 NI, 360 Huntington Ave., Boston, MA 02115-5000, USA, r.theodore@neu.edu), Joanne L. Miller(Northeastern University, Dept. of Psych. - 125 NI, 360 Huntington Ave., Boston, MA 02115-5000, USA, j.miller@neu.edu)

Recent findings indicate that listeners are sensitive to talker differences in phonetic properties of speech, including voice-onset-time (VOT) in word-initial voiceless stop consonants. Here we extend earlier findings from our laboratory [J. S. Allen and J. L. Miller, J. Acoust. Soc. Am. **115**, 3171-3813 (2004)] by examining the level of representation underlying this sensitivity. In familiarization phases, listeners heard two talkers produce pain. Critically, word-initial VOTs were manipulated such that one talker produced short VOTs and the other talker produced long VOTs. In test phases, listeners were presented with a short-VOT and long-VOT variant of either pain or cane; in both cases, listeners were asked to select which of the two VOT variants was most representative of a given talker. Results to date indicate that which variant of pain is selected at test is in line with listeners' exposure during training (replicating earlier findings), and that this effect holds even when listeners are tested on cane, which begins with a different voiceless stop than heard during training. These results suggest that listeners are sensitive to talker differences in VOT at the level of a phonetic feature, rather than at the level of a particular phonetic segment.

6:00

5pPPa12. Amplitude modulation of noise cues voicing distinction in fricatives. Jonathan Pincas (University of Surrey, GU2 7XH Guildford, UK, jon@pincas.co.uk), Philip J. Jackson (University of Surrey, Centre for Vision, Speech and Signal Processing, GU2 7XH Guildford, UK, p.jackson @surrey.ac.uk)

The aperiodic noise source in fricatives is characteristically amplitude modulated by voicing. Previous psychoacoustic studies have established that observed levels of AM in voiced fricatives are detectable, and its inclusion in synthesis has improved speech quality. Phonological voicing in fricatives can be cued by a number of factors: the voicing fundamental, duration of any devoicing, duration of frication, and formant transitions. However, the possible contribution of AM has not been investigated. In a cue trading experiment, subjects distinguished between the nonsense words "ahser" and "ahzer." The voicing boundary was measured along a formant-transition duration continuum, as a function of AM depth, voicing amplitude and masking of the voicing component by low-frequency noise. The presence of AM increased voiced responses by approximately 30%. The ability of AM to cue voicing was strongest at greater modulation depths and when voicing was unavailable as a cue, as might occur in telecommunication systems or noisy environments. Further work would examine other fricatives and phonetic contexts, as well as interaction with other cues.

Session 5pPPb

Psychological and Physiological Acoustics and Computational Acoustics: Computational Auralization II

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Contributed Papers

2:00

5pPPb1. Externalization in binaural synthesis: effects of recording environment and measurement procedure. Florian Völk (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, florian.voelk@mytum.de), Fabian Heinemann (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, hef@mmk.ei.tum.de), Hugo Fastl (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, fastl@mmk.ei.tum.de)

Databases of head related impulse responses (HRIRs) for binaural synthesis can be measured either in anechoic or reflective environments. If high synthesis quality is needed, miniature microphone measurements are performed in the ear canals of each individual user (individual measurement). Sometimes impulse responses measured in the ear canals of one individual are used for synthesis for other persons (nonindividual measurement). In most other cases artificial head measurements are used. This paper considers the dependence of the perceived distance of auditory images (externalization) on the measurement procedure (individual, nonindividual, artificial head) and on the recording environment (anechoic, reflective). For each measurement the same system and the same setup, especially the same geometric parameters, are used. Differences in the corresponding impulse response databases are determined and related to the subjective relative externalization differences in the front, in the back, and to both sides. For each direction a seven point rating scale was used. Statistical analysis suggests that the measurement parameters applied influence the externalization of auditory images.

2:20

5pPPb2. Smart sound environments: merging intentional soundscapes, nonspeech audio cues and ambient intelligence. Ralf Jung (Universität des Saarlandes, LS Wahlster, FB Informatik, Bldg. E 1 1, Room 1.18, 66123 Saarbrücken, Germany, rjung@cs.uni-sb.de)

We introduce an intelligent audio notification system for multiuser environments that provides users with information about events (e.g., important emails) in a more discreet and non-distracting way. The peripheral awareness of individual-related events is done by using nonspeech audio cues which can be seamlessly integrated into artificial background soundscapes. These ambient soundscapes are self-composed with respect to well-known perceptual constraints such as auditive Gestalt laws as well as music psychological findings. To follow a hierarchical approach for the notification sounds we use notification instruments, ambient noises and traditional alert signals that are grouped by their level of intrusiveness. Since the notification system also follows a human-centered approach it takes parameters like user preferences, his/her current position in the environment and the type of event into consideration to decide which notification is the appropriate at this time. In the paper, we will describe the architecture of the personalized ambient audio notification service, compositional constraints as well as some findings of a user study in which we tested successfully the efficiency of our system with 25 subjects.

2:40

5pPPb3. A virtual auditory environment for investigating the auditory signal processing of realistic sounds. Sylvain Favrot (CAHR, Department of Electrical Engineering, DTU, Ørsteds Plads, Bygning 352, 2800 Kgs. Lyngby, Denmark, sf@oersted.dtu.dk), Jörg M. Buchholz (CAHR, Department of Electrical Engineering, DTU, Ørsteds Plads, Bygning 352, 2800 Kgs. Lyngby, Denmark, jb@oersted.dtu.dk)

In the present study, a novel multichannel loudspeaker-based virtual auditory environment (VAE) is introduced. The VAE aims at providing a versatile research environment for investigating the auditory signal processing in real environments, i.e., considering multiple sound sources and room reverberation. The environment is based on the ODEON room acoustic simulation software to render the acoustical scene. ODEON outputs are processed using a combination of different order Ambisonic techniques to calculate multichannel room impulse responses (mRIR). Auralization is then obtained by the convolution of the mRIR with an acoustic signal. The derivation of the mRIRs takes into account that (i) auditory localization is most sensitive to the location of the direct sound and (ii) that auditory localization performance is rather poor for early reflections and even worse for late reverberation. Throughout the VAE development, special care was taken in order to achieve a realistic auditory percept and to avoid “artifacts” such as unnatural coloration. The performance of the VAE has been evaluated and optimized on a 29 loudspeaker setup using both objective and subjective measurement techniques.

3:00

5pPPb4. Real-time auralization system based on beam-tracing and mixed-order Ambisonics. Markus Noisternig (LIMSI-CNRS, B.P. 133, 91403 Orsay, France, markus.noisternig@limsi.fr), Lauri Savioja (Helsinki University of Technology, Department of Media Technology, PO Box 5400, 02015 TKK, Finland, Lauri.Savioja@tkk.fi), Brian F. Katz (LIMSI-CNRS, B.P. 133, 91403 Orsay, France, brian.katz@limsi.fr)

Auralization, the final step in computational room acoustic simulations, aims to make audible the acoustics of complex virtual architectural spaces in a realistic and accurate manner. This paper presents a novel real-time auralization system comprising a geometry engine, a beam-tracer, and an audio renderer. The computation of early reflection paths is based on an efficient beam-tracing algorithm capable of real-time detection of specular reflection paths in a static geometry with one or several moving listener(s). For simpler rooms, the real-time performance is maintained even with dynamic geometries and sources. Results of the beam-tracer, sent to the audio renderer, consist of visible reflection paths and their accumulated material attenuation. From this geometrical and acoustical data, listener position-related 3D room impulse responses are generated applying a higher-order virtual Ambisonics approach. Final rendering of the binaural room impulse response (BRIR) is made taking into account the listener’s head-orientation. As higher order reflections are more diffuse in nature, they may be encoded using lower Ambisonic orders, thereby reducing computational load. The environment combines high quality audio with visual rendering realized using the open source platforms Pure Data and VirChor respectively. This auralization framework provides direct audio-visual feedback in real-time for VR environments.

Invited Papers

3:20

5pPPb5. Real-time auralization of modifiable rooms. Dirk Schröder (Institute of Technical Acoustics, RWTH Aachen University, Neustr. 50, 52066 Aachen, Germany, dsc@akustik.rwth-aachen.de), Ingo Assenmacher (Virtual Reality Group, RWTH Aachen University, Seffenter Weg 23, 52074 Aachen, Germany, assenmacher@rz.rwth-aachen.de)

Immersive virtual environments are a powerful tool for acousticians and architects to design buildings if the virtual reality system provides an interactive imaging of virtual sound sources with respect to the rooms' physical aspects. Current implementations using hybrid room acoustic simulation methods (e.g., combining image sources and ray tracing) enable the user to walk freely in such virtual architectural spaces, whereby the position/orientation of sound sources are interactively manipulable to detect possible acoustic defects, e.g., flutter echoes. In the case of coupled rooms, sound transmission effects must be included into the real-time simulation in order to identify deficient airborne sound insulation, whereby current implementations only support a change of state (open/closed) of fixed room-connecting elements, e.g., doors and windows. However, in scenarios like an architectural planning stage, it is convenient to manipulate the room geometry more freely, e.g., via the interactive positioning of stage reflector panels, but common spatial data structures, e.g., BSP- or Octrees do not efficiently support these operations. For this purpose, the concept of Spatial Hashing, which originates from computer graphics for collision detection of deformable objects, is applied to the simulation process. This adaptation also features an efficient identification and update process of image sources.

3:40

5pPPb6. Comparison of auralisation results betwee measurements and simulations of line arrays with high resolution modeling data. 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), Ralph Bauer-Diefenbach (Ahnert Feistel Media Group, Arkonastr. 45-49, 13189 Berlin, Germany, rbauer@ada-acousticdesign.de)

In a concert hall a direct comparison of several line arrays for a new sound system has been made. At different locations the binaural impulse response has been measured and used for auralisation. In a computer model of the hall the measured line array was implemented. The single array sources have been modeled in different modes like far-field cluster, simple module array or as a high-resolution loudspeaker array. By means of the new developed SpeakerLab Module these source simulators have been created and afterwards used to calculate binaural impulse responses at the corresponding seats equal to the measurements. After that an auralisation routine has been used. The results are compared for the different degrees of source resolution and with the measurements. Beside acoustic measures the subjective sound quality of the different auralisation results is reported.

4:00-4:20 Break

Contributed Paper

4:20

5pPPb7. Investigation on the restitution system influence over perceived Higher Order Ambisonics sound field: a subjective evaluation involving from first to fourth order systems. Stephanie Bertet (IRCAM, 1 Place Igor Stravinsky, 75004 Paris, France, Stephanie.Bertet@ircam.fr), Jérôme Daniel (France Telecom R&D, 2 avenue Pierre Marzin, 22300 Lannion, France, jerome.daniel@orange-ftgroup.com), Etienne Parizet (Laboratoire Vibrations Acoustique, Insa Lyon, 25 bis, av. J. Capelle, 69621 Villeurbanne Cedex, France, etienne.parizet@insa-lyon.fr), Olivier Warusfel (IRCAM, 1 Place Igor Stravinsky, 75004 Paris, France, Olivier.Warusfel@ircam.fr)

Among the spatial audio reproduction techniques over loudspeakers, the Higher Order Ambisonics (HOA) approach is based on a sound field spherical harmonics decomposition. By truncating the decomposition to the Mth order, it remains a finite number of components that form the spatial HOA format. The more components are used to encode the sound field, the finer the spatial resolution is. Similarly, the size of the area where the sound field is accurately recreated is proportional to the order. For an Mth encoding order, $N=2M+2$ equally distributed loudspeakers are recommended for a homogeneous reproduction in the horizontal plane. Adding loudspeakers does not change the spatial resolution. However, what is the influence of the restitution system on the perceived sound field? An experiment was designed in order to compare four systems (from first to fourth order) and a reference one, using similarity ratings obtained from pairwise comparisons. Two sound scenes were used, simulating an audio conference and a scene in a kitchen at home. 25 listeners participated to the experiment. The results were analysed using the Indscal method. The perceptual space appeared to be a two dimensional one, highlighting the influence of the order and the number of loudspeakers on the reproduced scenes.

Invited Papers

4:40

5pPPb8. Recording of anechoic symphony music. Tapio Lokki (Helsinki University of Technology, P.O. Box 5400, 02015 TKK, Finland, Tapio.Lokki@tkk.fi), Jukka Pätynen (Helsinki University of Technology, P.O. Box 5400, 02015 TKK, Finland, jpatynen@tml.hut.fi), Ville Pulkki (Helsinki University of Technology, P.O. Box 5400, 02015 TKK, Finland, Ville.Pulkki@tkk.fi)

When designing the acoustics of a concert hall, it would be beneficial to be able to use real recording of a symphony orchestra in auralization. The technical constraints for such recordings are high. First, the instruments have to be recorded separately, as in simultaneous recording the cross talk between microphones could not be avoided. Second, the recording room should be anechoic. Third, the instruments have different sound radiation patterns, thus they should be recorded with multiple microphones around them. Therefore, we end up recording each instrument individually in an anechoic chamber with multiple microphones. The remaining problem is to achieve a common timing as an ensemble between the individually recorded instruments. This was solved by first recording a video of a conductor conducting a pianist playing the whole score. The players in an anechoic chamber then followed the conductor in a monitor while listening the pianist on headphones. Four short passages, from two to four minutes, from different music styles were recorded. The recordings were made with 20 low-self-noise microphones, mounted on the shape of a dodecahedron. Finally, we discuss the musical and technical quality of recorded sound, and the response by the musicians, who were professional orchestra players.

5:00

5pPPb9. Uni-Verse Acoustic Simulation System: interactive real-time room acoustic simulation is dynamic 3D environments. Peter Lundén (Interactive Institute, Box 1197, SE-164 26 Kista, Sweden, plu@tii.se)

Uni-Verse Acoustic Simulation System (UVAS) is a newly developed interactive room acoustic simulation system that can handle dynamically changing 3D geometric models in real-time. The system can share such models with other application, such as visual renderers or 3D modeling tools, over a network using the Verse protocol. UVAS is implemented using the beam-tracing method. It is build as two separate but highly integrated parts. The first part is handling the geometry, it's responsibility is to find audible sound sources and relevant reflection paths in the simulated environment. The second part is handling the audio rendering, producing the audible result of the simulation based on information given by the first part. This paper will focusing on the first part

5:20

5pPPb10. Case study of measurements and computer modeling auralization results for medium-sized multipurpose halls. Hari V. Savitala (Charles M Salter Associates, Inc., 130 Sutter St, Suite 500, San Francisco, CA 94104, USA, hari.savitala@cmsalter.com), Jason Duty (Charles M Salter Associates, Inc., 130 Sutter St, Suite 500, San Francisco, CA 94104, USA, jason.duty@cmsalter.com), Christopher Peltier (Charles M Salter Associates, Inc., 130 Sutter St, Suite 500, San Francisco, CA 94104, USA, christopher.peltier@cmsalter.com)

This case study focuses on auralizations and their accuracy in modeling medium-sized multipurpose halls (400-600 seats). Within each hall, impulse response measurements were taken with drapes deployed and retracted. The same configurations were modeled in the room acoustics program ODEON. Acoustical parameters, such as T20, T30, C50, C80, were used to check the agreement of the model to the measurements. A systematic approach was then used to adjust model parameters to match the real-world measurements. The modified and unmodified auralizations were then used to determine if any differences could be perceived in an informal listening evaluation. The auralizations and listening evaluation results are presented to better understand how to accurately auralize medium-sized multipurpose halls.

Contributed Papers

5:40

5pPPb11. Real-time 3D audio for digital cinema. Pau Arumi (Universitat Pompeu Fabra - Fundació Barcelona Media, Ocata, 1, 08003 Barcelona, Spain, parumi@iua.upf.edu), David Garcia (Fundacio Barcelona Media, Carrer Ocata n° 1, 08003 Barcelona, Spain, dgarcia@iua.upf.edu), Toni Mateos (Universitat Pompeu Fabra - Fundació Barcelona Media, Ocata, 1, 08003 Barcelona, Spain, toni.mateos@barcelonamedia.org), Adan Garriga (Universitat Pompeu Fabra - Fundació Barcelona Media, Ocata, 1, 08003 Barcelona, Spain, adan.garriga@barcelonamedia.org), Jaume Durany (Universitat Pompeu Fabra - Fundació Barcelona Media, Ocata, 1, 08003 Barcelona, Spain, jaume.durany@upf.edu)

We present a real-time 3D audio system with a number of nice features: it is suited for plausible reference with the visual environment, it is real-time capable, it can process multiple moving sound sources and listeners in a normal CPU. In our approach, a database of pressure and velocities impulse-responses (IRs) is computed offline for each (architectural) environment using physically based ray-tracing techniques. During playback, the real-time system retrieves IRs corresponding to the sources and target positions, performs a low-latency partitioned convolution and smoothes IR transitions with cross-fades. Finally, the system is flexible enough to decode to any surround exhibition setup. The software has been developed within the CLAM open-source audio framework. We present a real scenario where these techniques were successfully applied: an augmented-reality film with 3D audio within the context of the IP-RACINE project for digital cinema. The shooting was done with a high-end prototype camera with zoom and position tracking which enabled the real-time motion of a subjective listener within the scene. Our technology enabled the film director to both pre-hear surround audio of an augmented-reality scene shooting and fine-tune audio rendering in post-production.

6:00

5pPPb12. Near-field binaural synthesis, experimental progress report. Dylan Menzies-Gow (De Montfort University, Queens Building, LE1 9BH Leicester, UK, dylan@dmu.ac.uk)

A methodology was previously presented for displaying high quality binaural images of near-field complex sources, using wave reconstruction. Multipole representations of objects are transformed to Fourier-Bessel and plane wave expansions at the listener, before conversion to binaural signals. One advantage of this approach is that does not require special HRTF information other than the planewave HRTFs, and can fully render the complex field of a near object. As a first step towards a full working system, a real-time implementation is described here for displaying a monopole source using a six degrees-of-freedom infrared head tracking device.

Session 5pSC

Speech Communication: Multimodal Speech Technology

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Contributed Paper

2:00

5pSC1. An ultrasound-based silent speech interface. Thomas Hueber (ESPCI - Telecom Paris, 10 rue Vauquelin, 75005 Paris, France, hueber@ieee.org), Gerard Chollet (Telecom Paris Tech, 46 rue Barrault, 75013 Paris, France, chollet@enst.fr), Bruce Denby (Université Paris VI, ESPCI - Laboratoire d'Electronique, 10 rue Vauquelin, 75005 Paris, France, denby@ieee.org), Gerard Dreyfus (Université Paris VI, ESPCI - Laboratoire d'Electronique, 10 rue Vauquelin, 75005 Paris, France, gerard.dreyfus@espci.fr), Maureen Stone (Vocal Tract Visualization Lab, Depts of Biomedical Sciences and Orthodontics, University of Maryland Dental School, 650 W. Baltimore St., Baltimore, MD 21201, USA, mstone @umaryland.edu)

The paper proposes the use of ultrasound scans of tongue movement and video sequences of the lips to synthesize speech. A speech synthesizer driven only by video acquisitions may be qualified as a "silent speech interface," which could be used by laryngectomyzed patient as an alternative to tracheo-esophageal speech, for voice communication where silence must be maintained, or in very noisy environments. Our system is based on the building of a one-hour audiovisual corpus of phonetic units, which associates visual features extracted from video with acoustic observations. The ultrasound and optical images are interpreted as a linear combination of standard configurations obtained by principal components analysis (PCA) from a phonetically balanced subset of typical frames. HMM-based stochastic models trained on these visual features sequences are subsequently used to predict phonetic targets from video-only data. Finally, a Viterbi unit selection algorithm is used to find the optimal sequence of acoustic units given both this phonetic prediction and the sequence of visual features. The system is able to perform phonetic transcription from video-only speech data with over 55% correct recognition, on continuous speech, using neither phonotactic nor linguistic constraints.

Invited Papers

2:20

5pSC2. Multimodal control of talking heads. Gerard Bailly (GIPSA-lab. Dept Speech & Cognition, INPG, 46, av. Félix Viallet, 38031 Grenoble, France, gerard.bailly@gipsa-lab.inpg.fr), Oxana Govokhina (GIPSA-lab. Dept Speech & Cognition, INPG, 46, av. Félix Viallet, 38031 Grenoble, France, oxana.govokhina@gipsa-lab.inpg.fr), Gaspard Breton (Orange R&D, 4 rue du Clos Courtel, 35512 Cesson-Sévigné, France, gaspard.breton@orange-ftgroup.com)

Multimodal speech synthesis has been devoted for years to the rendering of linguistic or paralinguistic content - i.e., parametrized but discrete information - by continuous audible and visible consequences of speech articulation, eventually complemented by facial expressions, gaze and other body gestures including head, hand, and arm movements. Articulatory synthesizers (producing sounds from gestures) intrinsically compute coherent audiovisual signals but do not presently compete with data-driven techniques: most talking heads are nowadays controlled by models built using human audiovisual data. These control models should replicate the laws governing the coherence of observed multimodal signals and the correct phasing relations between salient events of the multimodal stream. We will report on two comparative evaluations of various lip-sync models (dealing with post-synchronization between speech sounds and articulatory movements) and present a trainable control model that learns automatically phasing relations between acoustic and gestural events. This model can be further extended to capture the fine temporal structure of multimodal scores and a first application to the synchronization between speech and head, face and hand movements during cued speech production will be presented.

2:40

5pSC3. Statistical conversion of speech parameter trajectory for mapping between features of different modalities. Tomoki Toda (Nara Institute of Science and Technology, 8916-5 Takayama-cho, Ikoma, 630-0192 Nara, Japan, tomoki@is.naist.jp)

A state-of-the-art speech parameter conversion technique and its application to a mapping between features of different modalities are reviewed. Many statistical approaches to the parameter conversion have been studied particularly for voice conversion in speech synthesis research. A typical method conducts the parameter conversion frame by frame based on the minimum mean square error using a Gaussian mixture model of the joint probability density of input and output parameters [Y. Stylianou et al., IEEE Trans. SAP, **6**(2), 131-142 (1998)]. Although this method is reasonably effective, the deterioration of the conversion accuracy is caused by essential problems of the frame-based conversion process. Recently a conversion method based on the maximum likelihood estimation of a parameter trajectory has been proposed [T. Toda et al., IEEE Trans. ASLP, **15**(8), 2222-2235 (2007)]. This method realizes the appropriate converted parameter sequence by (1) using not only static but also dynamic feature statistics and (2) considering a global variance feature of the converted parameters. It has been reported that this method is effective in several applications such as a spectral determination from articulatory movements, an acoustic-to-articulatory inversion mapping, and a conversion of body-transmitted speech into air-transmitted speech.

Contributed Paper

3:00

5pSC4. A comparison of visual features for audiovisual automatic speech recognition. Nasir Ahmad (Loughborough Univ, LE11 3TU Leicestershire, UK, n.ahmad@lboro.ac.uk), Sekharjit Datta (Loughborough Univ, LE11 3TU Leicestershire, UK, s.datta@lboro.ac.uk), David Mulvaney (Loughborough Univ, LE11 3TU Leicestershire, UK, d.j.mulvaney@lboro.ac.uk), Omar Farooq (Loughborough Univ, LE11 3TU Leicestershire, UK, o.farooq@lboro.ac.uk)

The use of visual information from speaker's mouth region have shown to improve the performance of automatic speech recognition (ASR) systems. This is particularly important in presence of noise which even in moderate form severely degrades the speech recognition performance of systems using only audio information. Various sets of features extracted from speaker's mouth region have been used to improve upon the performance of an ASR system in such challenging conditions and have met many successes. To the best of authors knowledge, the effect of using these techniques on recognition performance on the basis of phonemes have not been investigated yet. This paper presents a comparison of phoneme recognition performance using visual features extracted from mouth region-of-interest using discrete cosine transform (DCT) and discrete wavelet transform (DWT). New DCT and DWT features have also been extracted and compared with the previously used one. These features were used along with audio features based on Mel frequency cepstral coefficients (MFCC). This work will help in selecting suitable features for different application and identify the limitations of these methods in recognition of individual phonemes.

3:20-3:40 Break

Invited Paper

3:40

5pSC5. Spatial rendering of audiovisual synthetic speech use for immersive environments. Markus Noisternig (LIMSI-CNRS, B.P. 133, 91403 Orsay, France, markus.noisternig@limsi.fr), Brian F. Katz (LIMSI-CNRS, B.P. 133, 91403 Orsay, France, brian.katz@limsi.fr), Christophe D'Alessandro (LIMSI-CNRS, B.P. 133, 91403 Orsay, France, cda@limsi.fr)

Synthetic speech is usually delivered as a mono audio signal. In this project, audiovisual speech synthesis is attributed to a virtual agent moving in a virtual three-dimensional scene. More realistic acoustic rendering is achieved by taking into account the position of the agent in the scene, the acoustics of the room depicted in the scene, and the orientation of the virtual character's head relative. 3D phoneme dependant radiation patterns have been measured for two speakers and a singer. These data are integrated into a Text-To-Speech system using a phoneme to directivity pattern transcription module which also includes a phoneme to viseme model for the agent. In addition to the effects related to agent's head orientation for the direct sound, a room acoustics model allows for realistic rendering of the room effect as well as the apparent distance as depicted in the virtual scene. Real-time synthesis is implemented in a 3D audio rendering system.

4:00

5pSC6. Audiovisual automatic speech recognition: Progress and challenges. Gerasimos Potamianos (IBM T. J. Watson Research Center, RTE 134, Yorktown Heights, NY 10598, USA, gpotam@us.ibm.com)

The paper overviews recent progress and challenges in a number of audiovisual speech processing technologies with main emphasis on the problem of automatic speech recognition. It is well known that visual channel information can improve automatic speech processing for human-computer interaction. To automatically process and incorporate such information into automatic systems, a number of steps are required that are surprisingly similar accross speech technologies. Crucial above all is the issue of feature representation of visual speech and its robust extraction. In addition, appropriate integration of the audio and visual representations is required, in order to ensure improved performance of the bimodal systems over audio-only baselines. These topics are discussed in detail in the talk, with main emphasis on their application to the speech recognition problem in the challenging environments of automobiles and smart rooms.

Contributed Paper

4:20

5pSC7. Analysis and synthesis of nonverbal facial motion. Jonas Beskow (KTH Speech, Music and Hearing, Lindstedtsvägen 24, 10044 Stockholm, Sweden, beskow@kth.se), Björn Granström (KTH Speech, Music and Hearing, Lindstedtsvägen 24, 10044 Stockholm, Sweden, bjorn @speech.kth.se)

Until recently, most efforts in audio-visual speech synthesis have been concerned with verbal content. However, in human-human communication it is obvious that nonverbal signals plays an important role, such as when expressing emotions and attitudes. Interaction is also often regulated using facial cues, for example gaze, head and eyebrow movements. Some of these cues have a direct coupling to the speech signal, while other occur during both while speaking and listening. When applying interactive talking agents in man-machine systems, nonverbal signals may be very important in easing the flow of communication. In a series of experiments we have been exploring the function of nonverbal facial motion. These studies include an experiment on the interaction between expressive speech and prominence, as well as an attempt to synthesize emotions and attitudes in a talking head, using 3D motion capture data. Further we will report on a real-time experiment with human-human avatar-mediated conversation, where the subjects' turn-taking behavior is affected by facial motion in the avatars.

Session 5pUWa

Underwater Acoustics and ECUA: Sound Propagation in 3D Environments II

David C. Calvo, Cochair
U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA

Michael Taroudakis, Cochair
University of Crete & FORTH/IACM, Vassilika Vouton, P.O. Box 1385, Heraklion, 711 10, Greece

Invited Paper

1:40

5pUWa1. Stability of wavefronts at sound propagation in highly structured three-dimensional environments. Oleg A. Godin (NOAA/ESRL, 325 Broadway, Mail Code R/PSD99, Boulder, CO 80305-3328, USA, Oleg.Godin@noaa.gov)

Extensive numerical modeling of long-range propagation of sound and seismic waves as well as observations of underwater acoustic fields with line arrays reveal that wavefronts are often much more stable and predictable than the rays comprising these wavefronts. This paper considers multiple scattering of sound by environmental inhomogeneities with spatial scales small compared to the propagation range but large compared to the wavelength. These inhomogeneities include 3D variations in sound speed and current velocity that are small compared to the average sound speed, can be either random or deterministic, and are superimposed on an arbitrary slowly-varying background. A theoretical explanation of wavefront stability in highly-structured environments is achieved by demonstrating that end points of rays launched from a point source and having a given eikonal (phase) are scattered primarily along the wavefront corresponding to the same eikonal in the unperturbed environment. The ratio of displacements of the ray end points along and across the unperturbed wavefront is proportional to the number of uncorrelated scattering events. The results apply to conventional rays and to horizontal rays describing propagation of adiabatic normal modes in almost-layered media. The origin of relative stability of wavefronts compared to rays is traced back to Fermat's principle.

Contributed Paper

2:00

5pUWa2. Range and cross-range propagation effects in a liquid wedge overlaying an elastic bottom. Piotr Borejko (Vienna University of Technology, Karlsplatz 13/E206/3, A-1040 Vienna, Austria, pb@allmech.tuwien.ac.at)

The penetrable-wedge model, a constant density isospeed layer of fluid with a pressure-release horizontal surface and a sloping elastic bottom, provides an extensive insight into the role of the ocean bottom in acoustic propagation from an underwater source. In particular, it is a realistic model of a rock-bottom ocean near a shoreline that accounts for horizontal refraction and allows for a ground wave. This paper discusses some new results for small and large range propagation for two penetrable-wedge models: one where the shear wave speed in the bottom is lower than the sound speed in the fluid and the other where the shear wave speed is higher. The operational representation of the ray-integral solution for the acoustic field from a point source in a penetrable wedge of fluid is further developed to the stage at which one can compute the exact, other than the omission of diffraction at the wedge apex, pressure response curve, as recorded at a receiver, due to an arbitrary time variation of the pressure at the source in a 3° wedge. The three-dimensional propagation effects are examined for range transmission when the receivers are located up-slope and down-slope of the source, and for cross-range transmission when the receivers are located cross-slope of the source.

Invited Papers

2:20

5pUWa3. Depth-dependent resonant target strength analysis of a dense Atlantic Herring school from wide-area OAWRS and localized 3D morphology sensing. Daniel Cocuzzo (Northeastern University, 302 Stearns Center, Rm 311, 360 Huntington Ave, Boston, MA 02115, USA, dcocuzzo@ece.neu.edu), Zheng Gong (Northeastern University, 302 Stearns Center, Rm 311, 360 Huntington Ave, Boston, MA 02115, USA, zgong@ece.neu.edu), Mark Andrews (Northeastern University, 302 Stearns Center, Rm 311, 360 Huntington Ave, Boston, MA 02115, USA, Andrews.mar@neu.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), 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), Purnima Ratilal (Northeastern University, 302 Stearns Center, Rm 311, 360 Huntington Ave, Boston, MA 02115, USA, purnima@ece.neu.edu)

The depth-dependent target strength of Atlantic Herring is estimated at several distinct bandwidths close to their resonance frequency for a localized, highly dense school observed during the NOPP-sponsored Gulf of Maine Experiment on September 22, 2006. An ocean acoustics waveguide remote sensing (OAWRS) system was deployed near George's Bank to investigate the migration and spawning behavior of fish over wide areas. In conjunction with OAWRS, a Simrad EK60 conventional fish-finding echosounder (CFFS) and a Reson Seabat 7125 multibeam sonar system were deployed to provide local depth extent and 3D volume morphology of the dense herring school. The calibration of low-frequency target strength derived from OAWRS data using localized CFFS density and multibeam 3D volume estimates as inputs is discussed. The correlation between the mean depth of the vertically migrating herring school and its resonance frequency is investigated. The results are compared with a theoretical model for 3D resonance scattering from fish swimbladder modeled as a spheroidal bubble. This analysis may allow inference of fish depth and species classification based on the scattered frequency response of targets imaged by OAWRS. Implications for classifying general localized targets, biological or man-made, are discussed further.

2:40

5pUWa4. Multiple forward scattering through an ocean waveguide with 3D random inhomogeneities. Nicholas Makris (Massachusetts Institute of Technology, Room 5-212, 77 Massachusetts Avenue, Cambridge, MA 02139, USA, makris@mit.edu), Purnima Ratilal (Northeastern University, 302 Stearns Center, Rm 311, 360 Huntington Ave, Boston, MA 02115, USA, purnima@ece.neu.edu), Tianrun Chen (Massachusetts Institute of Technology, Room 5-212, 77 Massachusetts Avenue, Cambridge, MA 02139, USA, trchen@mit.edu)

Analytic expressions have been derived for the mean and spatial covariance of the acoustic field multiply forward scattered though a stratified ocean waveguide containing 3D random surface or volume inhomogeneities [Ratilal and Makris, J. Acoust. Soc. Am. **118**, 3532-3559 (2005)]. These expressions are further used to derive the temporal coherence of an acoustic signal propagated through 3D random inhomogeneities. Field moments are given in terms of moments of the scatter function density of the 3D random inhomogeneities, which enables straightforward application to a broad range of 3D scatterers. Here we give examples of the attenuation, dispersion and loss of temporal coherence expected after multiple forward scattering through (1) random internal waves in both continental shelf and deep ocean environments, (2) fish schools, and (3) random wind-generated bubbles in continental shelf and surf-zone area. We show that 3D scattering effects become important when the Fresnel width exceeds the cross-range coherence scale of the inhomogeneities, and can lead to substantial power loss.

Contributed Papers

3:00

5pUWa5. Application of the matrix Rytov method to the calculation of the coherence function of a sound field in an oceanic waveguide. Alex G. Voronovich (NOAA/Earth System Research Laboratory, 325 Broadway, Boulder, CO 80305, USA, alexander.voronovich@noaa.gov), Vladimir E. Ostashev (NOAA/Earth System Research Laboratory, 325 Broadway, Boulder, CO 80305, USA, vladimir.ostashev@noaa.gov)

Closed equations for the coherence function of a monochromatic sound field propagating in a statistically inhomogeneous 3D oceanic waveguide have high dimensions and are difficult to solve even with the use of modern computers. Significant reduction of the dimension of the problem was achieved by assuming that sound speed fluctuations are statistically isotropic in a horizontal plane. However, even in this case calculation of the coherence function for a megameter range takes about a day. In this paper, we develop an approximate solution of the closed equations for the coherence function which is similar to a matrix version of the Rytov method. An explicit expression for the coherence function is obtained which contains exponent of an ″interaction″ matrix. This matrix is determined in terms of the acoustic and internal wave modes and spatial spectrum of the sound speed fluctuations. It is shown that the matrix Rytov method provides an accurate solution for the coherence function which coincides with the solution of the closed equations within a few percent. Calculation of the coherence function now takes only about an hour. This allows us to study in detail the dependence of the coherence function on parameters of the problem.

3:20

5pUWa6. Measurements of 3D propagation in the shelf environment. Kevin D. Heaney (Oasis Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, USA, oceansound04@yahoo.com)

In the Fall of 2007 measurements were made to calibrate the acoustic environment on the shallow water shelf off the coast of Florida. The continental shelf is quite flat (less than 1 degree slope) for a region approximately 10 km wide. Transmissions to a cross-shelf 900 m horizontal line array were made from a source transiting along the shelf. The signal transmissions include broadband LFM (20-420 Hz) and a comb of narrow band frequencies spanning the same range. Narrowband beamforming results show the clearly identified source. The arrival angle for the source is as expected until a distance of approximately 30-40 km when there is an apparent bearing shift in-shore of up to 25 degrees. This behavior is expected in propagation on a shelf (of greater slope) but its behavior is surprising. The phenomenon was observed for several runs at various source and water depths. In order to explain the phenomenon, a hybrid adiabatic normal mode-Parabolic Equation method will be applied to the environment. This model computes the vertical modes and phase speeds at each location and then uses the PE to propagate each mode individually using it's spatially varying phase speed and attenuation. Comparisons of theory and data will be made.

3:40

5pUWa7. Observations of out of plane arrivals for long range low frequency transmission in shallow water. Harry Deferrari (Univ. of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149, USA, hdeferrari@rsmas.miami.edu)

Two recent experiments have used long horizontal arrays to receive broadband low frequency signals propagated over long ranges in shallow water. Both used m-sequence signals that resolve pulse arrivals in time with each arrival associated with a single acoustic mode of propagation. At moderate propagation ranges, out to 20 km, wave fronts for all modes are observed to be parallel, implying an orderly two-dimensional propagation. At a much longer range, 80 km, a number of separable arrivals are observed but not necessarily with a one-to-one correspondence with modes. The paths appear to be stable and coherent in time implying that they are true Fermat paths, but their wavefront arrival angles differ suggesting the same mode is arriving from several directions, that is, by curved (out of plane paths). The paths could result from wedge effects from gentle slopes perpendicular to the propagation path or possibly from chaotic interaction with random facets of the bottom. In any case, the ultimate limitation for horizontal spatial coherence and array resolution may be the multipath interference of bundles of out of plane arrivals.

4:00-4:20 Break

Invited Papers

4:20

5pUWa8. Hydroacoustic blockage prediction and measurement at Diego Garcia using the Adiabatic Mode Parabolic Equation Model. Zachary Upton (BBN Technologies, 1300 N. 17th Street, Suite 400, Arlington, VA 22209, USA, zupton@bbn.com), Michael D. Collins (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, michael.collins@nrl.navy.mil), Jay Pulli (BBN Technologies, 1300 N. 17th Street, Suite 400, Arlington, VA 22209, USA, jpulli@bbn.com)

Underwater explosion monitoring with sparse sensors at long ranges relies on the efficient propagation of acoustic energy in the sound fixing and ranging (SOFAR) channel. When sound traveling in this channel encounters an island or seamount, it will either diffract, scatter, or be converted into seismic energy. Signals observed on the opposite side of these obstructions have been affected by some combination of these processes, and models of global detection and localization depend on knowing these effects. We present a study using the Adiabatic Mode Parabolic Equation (AMPE) model to predict these processes in three dimensions at the Chagos Archipelago. Predictions at 5, 10, and 20 Hz are compared with measurements of approximately 300 T-wave signals from six years of earthquakes on either side of the Chagos Archipelago. These have been recorded at the hydrophone arrays around Diego Garcia. The result of this 360-degree analysis, and the agreement with observed data, demonstrate the utility of the model in understanding the physical effects of these obstructions.

4:40

5pUWa9. Investigation of 3D benchmark problems in underwater acoustics: a uniform approach. Frederic Sturm (Laboratoire de Mécanique des Fluides et d'Acoustique (UMR CNRS 5509), Ecole Centrale de Lyon, Centre acoustique, 36, avenue Guy de Collongue, 69134 Ecully Cedex, France, frederic.sturm@ec-lyon.fr)

In underwater acoustics, most of the three-dimensional effects on sound wave propagation are usually described by modelers considering one of the following shallow-water benchmark problems: a wedge-shaped waveguide, a canyon, a seamount, and a sinusoidal (corrugated) bottom. These test cases have been thoroughly analyzed individually considering both harmonic point sources (emitting at very low frequencies, for some obvious problems of CPU time and memory limitation) and broadband source pulses (with also very-low central frequencies). In the present work, we report numerical results corresponding to the propagation of broadband pulses in the four above-mentioned test cases. The numerical simulations are performed using a fully 3D parabolic equation based model coupled with a Fourier synthesis technique to handle the time dependence of the source signal. The objective is to propose a uniform representation of the numerical results so as to facilitate the comparison of the 3D effects present in each of the four benchmarks. Snapshots of the propagating pulses at very close successive times are compared with 2D results. In addition, movies of the propagating pulses are shown for each test case and compared to each other. Movies strongly facilitate the observation and thus the understanding of the 3D effects experienced by all the propagating waves.

Contributed Paper

5:00

5pUWa10. Using parallel programming and a three-dimensional visualization cave to map the acoustic energy distribution from a seismic array in the ocean. Natalia Sidorovskaia (Department of Physics, University of Louisiana, UL BOX 44210, Lafayette, LA 70504-4210, USA, nas@louisiana.edu), Arslan Tashmukhambetov (Department of Physics, University of New Orleans, New Orleans, LA 70148, USA, atashmuk@uno.edu), George E. Ioup (Department of Physics, University of New Orleans, New Orleans, LA 70148, USA, geioup@uno.edu), Juliette W. Ioup (Department of Physics, University of New Orleans, New Orleans, LA 70148, USA, jioup@uno.edu)

Modeling and visualization of the dynamic acoustic field during a seismic exploration survey represent a computational challenge due to broadband, directional nature of the acoustic signal radiated by a seismic array. Standard acoustic propagation models (RAM and SWAMP) are upgraded for parallel processing and tested in the LONI (the Louisiana Optical Network Initiative) environment, using the Louisiana fiber optics grid computing network to model the three-dimensional time-varying acoustic field in the ocean during a seismic exploration survey. The generated volume of data is transferred and visualized in the advanced immersive visualization environment, supported by Louisiana Immersive Technology Enterprise (LITE) facilities. The proposed technology is one of the first steps in developing real-time monitoring of the acoustic energy distribution in a large oceanic volume. This can be beneficial for environmental impact assessment and regulation and for seismic survey design. [Research supported in part by the Joint Industry Project through OGP and by ITI of University of Louisiana at Lafayette.]

Invited Paper

5:20

5pUWa11. Effects of solitons on acoustic energy flow in three dimensions. Kevin B. Smith (Naval Postgraduate School/Naval Undersea Warfare Center, Code PH/Sk, Department of Physics, Monterey, CA 93943, USA, kbsmith@nps.edu), John A. Colosi (Naval Postgraduate School, Code OC/Cj, Department of Oceanography, Monterey, CA 93943, USA, jacolosi@nps.edu)

The impact of a train of nonlinear solitons on the propagation of acoustic energy in shallow water is examined. The soliton perturbations are based on an analytic formulation that produces a train of five soliton waves. Each wave front is parallel and has infinite extent in the horizontal direction. The acoustic field is modeled using a three-dimensional (3D) split-step Fourier parabolic equation (SSF/PE) approach defined in Cartesian coordinates. The standard PE approximation is employed in both depth and cross-range directions. Both pressure and particle velocity fields are computed in a self-consistent manner, allowing a full description of the 3D acoustic intensity field which describes the flow of energy in the presence of the solitons. Individual, low-order modes are extracted from the propagating field so that the impact on specific modes may be examined. The analysis is performed at various frequencies and

for various source-receiver geometries relative to the soliton train. Emphasis is placed on the focusing and defocusing of acoustic energy between the various soliton waves. The impact of such soliton perturbations on signal variability and bearing resolution at the receiver will be quantified. [Work supported by ONR 321OA.]

Contributed Papers

5:40

5pUWa12. High-frequency underwater acoustic propagation in a port using the three-dimensional method of images. Pierre-Philippe J. Beaujean (Florida Atlantic University, SeaTech Campus, 101 North Beach Road, Dania Beach, FL 33004, USA, pbeaujea@seatech.fau.edu), Matthew D. Staska (International Transducer Corporation, 869 Ward Drive, Santa Barbara, CA 93111, USA, MStaska@channeltech.com)

A computer-efficient model for underwater acoustic propagation in a shallow, three-dimensional rectangular duct closed at one end has been developed using the method of images. The duct simulates a turning basin located in a port, surrounded with concrete walls and filled with sea water. The channel bottom is composed of silt. The modeled impulse response is compared with the impulse response measured between 15 kHz and 33 kHz. Despite small sensor-position inaccuracies and an approximated duct geometry, the impulse response can be modeled with a relative echo magnitude error of 1.62 dB at worst, and a relative echo location error varying between 0% and 4% when averaged across multiple measurements and sensor locations. This is a sufficient level of accuracy for the simulation of an acoustic communication system operating in the same frequency band and in shallow waters, as time fluctuations in echo magnitude commonly reach 10 dB in this type of environment.

6:00

5pUWa13. Acoustic mode beam effects of nonlinear internal gravity waves in shallow water. Timothy Duda (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 107, MS-12, Woods Hole, MA 02543, USA, tduda@whoi.edu), Ying-Tsong Lin (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 107, MS-12, Woods Hole, MA 02543, USA, ytlin@whoi.edu), James F. Lynch (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, jlynch@whoi.edu)

Ducting of sound between short-wavelength nonlinear internal gravity waves in coastal environments has been demonstrated by substantial evidence. The ducting takes a unique form for each of the acoustic normal modes. Some consequences of this are examined here using three-dimensional parabolic equation modeling and theory. For a pair of waves having a broadband 200-Hz source placed between (i.e., in the duct), strong interference patterns within the duct are developed for each mode after a few kilometers. Some of the energy escapes at high angle with respect to the duct direction. Termination of the internal wave duct, an observed feature, results in beams of energy unique to each mode to radiate outward. Specific cases having water depths of order 80 m and propagation distances of 20 to 30 km are examined. Situations where one or more modes are completely absent at selected positions are compared with similar events observed in the field.

FRIDAY AFTERNOON, 4 JULY 2008 — ROOM 342A, 1:40 TO 6:20 P.M.

Session 5pUWb

Underwater Acoustics and ECUA: Scattering From Objects Near Boundaries

Eric Thorsos, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA

Mario Zampolli, Cochair
NATO Undersea Research Centre, Viale San Bartolomeo 400, La Spezia, 19126, Italy

Invited Papers

1:40

5pUWb1. Measurement and modeling of targets deployed on and within sand sediments. Kevin L. Williams (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, williams@apl.washington.edu), Eric Thorsos (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, eit@apl.washington.edu), Steven Kargl (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, USA, kargl@troutmask.apl.washington.edu), Joseph Lopes (Naval Surface Warfare Center - Panama City Division, 110 Vernon Ave, Panama City, FL 32407, USA, joseph.l.lopes@navy.mil), Raymond Lim (Naval Surface Warfare Center - Panama City Division, 110 Vernon Ave, Panama City, FL 32407, USA, raymond.lim@navy.mil), Carrie Dowdy (Naval Surface Warfare Center - Panama City Division, 110 Vernon Ave, Panama City, FL 32407, USA, carrie.dowdy@navy.mil)

Acoustic signatures of elastic targets located near sediment interfaces include effects due to energy interacting with the sediment. Therefore, modeling target response also requires models of scattering from, penetration into and propagation within ocean sediments. We first describe at-sea and test pond measurements carried out on “proud” (target resting on the sediment) and buried targets at frequencies in the range of 2 to 50 kHz. The results from some of these measurements are then compared to models incorporating various levels of sophistication relative to both the target and the sediment physics. The modeling hierarchy includes the following: (1) simple sonar equation estimates that treat the target physics via a frequency dependent target strength and use formally averaged results for sediment scattering, (2) realization level modeling that allows calculation of sediment and target scattering for individual pings with

sufficient fidelity to carry out synthetic aperture processing (for a proud target only its geometrical scattering is considered while the elastic response can be included for a buried target), (3) T-matrix and finite element modeling in which the target elastic response is included but sediment scattering is treated using formal averages and/or flat surface approximations. [Work supported by the Office of Naval Research and the Strategic Environmental Research and Development Program, USA.]

2:00

5pUWb2. Synthetic aperture sonar imaging of simple finite targets near a sediment-water interface. Steven Kargl (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, USA, kargl@troutmask.apl.washington.edu) , Kevin L. Williams (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, williams@apl.washington.edu), Eric Thorsos (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, eit@apl.washington.edu), Darrell R. Jackson (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, USA, drj@apl.washington.edu), Dajun Tang (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, djtang@apl.washington.edu)

Synthetic aperture sonar (SAS) is used often to detect targets that are either proud or buried below a sandy sediment interface where the nominal grazing angle of incidence from the SAS to the point above a buried target is below the critical grazing angle. A numerical model for scattering from simple targets in a shallow water environment will be described, and can be used to generate pings suitable for SAS processing. For buried targets, the model includes reverberation from the rough seafloor, penetration through the interface, target scattering, and propagation back to the SAS. The reverberation and penetration components are derived from first order perturbation theory where small-scale roughness and superimposed ripple can be accommodated. For proud targets, the simulations include the scattering from the target where interaction with the seafloor is included through simple acoustic ray models. The interaction of the target with an incident field is based on a free field scattering model. Simulations will be compared to both benchmark problems and measurements over a frequency range of 10-30 kHz. These comparisons further support sediment ripple structure as the dominant mechanism for subcritical penetration in this frequency range. [Work supported by the US Office of Naval Research.]

2:20

5pUWb3. Experiments and numerical modeling of low to midfrequency scattering from elastic objects near the sea floor. Mario Zampolli (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, zampolli@nurc.nato.int), Alessandra Tesei (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, tesei@nurc.nato.int), Finn B. Jensen (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, jensen@nurc.nato.int), Gaetano Canepa (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, canepa@nurc.nato.int)

The scattering of low to mid-frequency sound (1-10's of kHz) from submerged elastic structures of size O(1m) is a topic of interest to the underwater acoustics community. In the first part of the presentation, a brief description of the relevant components of the EVA experiment is given. The purpose of the sea trial was the acquisition of high-fidelity echoes from submerged spherical and cylindrical targets, made of composite materials with internal layered structure. The second part of the presentation is focused on the finite-element modeling technique developed at NURC for investigating the scattering from axially symmetric submerged elastic objects. Particular attention is dedicated to the computation of the far field at a distance from the target via the Helmholtz-Kirchhoff integral, using the near field sampled on the target surface, together with Green's functions capable of describing a two-layered water-sediment fluid medium. Those geometries, for which the overall axial symmetry is broken by the presence of the water-sediment boundary, can be treated approximately by taking into account the boundary-reflected incident field, as well as the first order interaction between the target-scattered echo and the sea floor. The numerical technique is validated by comparison with data collected during the EVA trial.

2:40

5pUWb4. Modeling bottom penetration for buried target detection. Raymond Lim (Naval Surface Warfare Center - Panama City Division, 110 Vernon Ave, Panama City, FL 32407, USA, raymond.lim@navy.mil), Gary S. Sammelmann (Naval Surface Warfare Center - Panama City Division, 110 Vernon Ave, Panama City, FL 32407, USA, gary.sammelmann@navy.mil)

Sonar detection of targets buried in underwater sediments has been found to be complicated by surface roughness. In particular, current-induced ripples can diffract energy down into sandy sediments to enhance buried target detection at shallow sonar grazing angles. To validate these effects, models encompassing the dominant propagation mechanisms as well as faithfully representing the target in the environment have been used. This paper describes our efforts to adapt transition matrix and perturbation theory models to provide realistic predictions of buried target response for spherical and cylindrical shapes. Combining these models of scattering and penetration required adopting some approximations to reduce computation time while retaining accuracy. Steps taken to verify and exercise the resulting models reveal some sensitivities that accentuate the need for accurate environmental and setup ground truth for validation of detection mechanisms. [Work supported by the Office of Naval Research and the Strategic Environmental Research and Development Program, USA.]

3:00

5pUWb5. Scattering by a partially exposed nearly rigid cylinder: Experiments and analysis. Kyungmin Baik (Washington State University, Physics and Astronomy Department, Pullman, WA 99164-2814, USA, nupho27@dreamwiz.com), Philip L. Marston (Washington State University, Physics and Astronomy Department, Pullman, WA 99164-2814, USA, marston@wsu.edu)

The backscattering from a partially exposed circular cylinder was measured for broad side illumination under conditions where the contributions associated with the elastic response of the cylinder were expected to be weak. Grazing illumination was used. Since the objective was to investigate the transition in the number of reflected rays with increasing exposure, it was convenient to partially submerge the cylinder through the free surface of a tank of water. The magnitude of the scattering was measured for ka between 9.6 and 16 where k is the acoustic wave number and a is the radius of the cylinder. The scattering varied smoothly as a function of the cylinder's

exposure in agreement with analytical results based on a Kirchhoff approximation [K. Baik and P. L. Marston, IEEE J. Oceanic Eng. (accepted)]. The analysis is easily modified for the case a cylinder breaking through a flat hard interface. For the different types of rays, path length calculations (as a function of the exposure) that are part of that analysis are also relevant to identifying echoes in SAS images of partially exposed cylinders and spheres. [Research supported by ONR.]

3:20

5pUWb6. Elastic and interfacial contributions to SAS images of tilted metal cylinders: Laboratory experiments. Jon La Follett (Washington State University, Physics and Astronomy Department, Pullman, WA 99164-2814, USA, LAFOLLEJ@mail.wsu.edu), Kyungmin Baik (Washington State University, Physics and Astronomy Department, Pullman, WA 99164-2814, USA, nupho27@dreamwiz.com), Philip L. Marston (Washington State University, Physics and Astronomy Department, Pullman, WA 99164-2814, USA, marston@wsu.edu)

Laboratory experiments were carried out to explore interfacial and elastic contributions to synthetic aperture sonar (SAS) images of a solid aluminum circular cylinder having flat ends. Some of the elastic responses for free field backscattering as a function of tilt angle could be interpreted using prior ray-based theory of generalized Rayleigh wave contributions [K. Gipson and P.L. Marston, J. Acoust. Soc. Am. 106, 1673-1680 (1999); 107, 112-117 (2000)]. Simplified acoustic holography was also used to interpret aspects of the free field bistatic response. To study the effects of proximity to a flat reflecting surface, the cylinder was hung through the free surface of a water tank and monostatic SAS images were acquired by scanning the transducer location along a horizontal line. This arrangement partially simulates SAS images of cylinders on the ocean bottom at grazing incidence. There were bright contributions to the SAS images of tilted cylinders associated with direct elastic rays as well as with indirect elastic rays due to acoustic reflections from the free surface of the water tank. [Research supported by ONR.]

3:40

5pUWb7. Elastic scattering by partially-solid-filled spherical shell on the seabed: Model-data comparison and physical understanding. Alessandra Tesei (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, tesei@nurc.nato.int), Mario Zampolli (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, zampolli@nurc.nato.int), Gaetano Canepa (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, canepa@nurc.nato.int)

Low- to mid-frequency elastic scattering measurements were conducted in the range from 5 to 40 ka on a spherical composite shell deployed proud on a sandy seabed. The object consists of a thin-walled shell made of layers of a random-fiber material, and then filled partially with an isotropic solidified epoxy resin and partially with sea water. A scaled version of the object was measured in a water tank under free field conditions. The target responses obtained with and without interaction with the seafloor boundary were compared to simulations achieved by the NURC modeling tool Axiscat. The temporal echoes of the objects were analyzed in terms of elastic waves supported by the structure, on the basis of a ray model. The strongest elastic components come from the interior solid filler. The experimental data of the sphere on the seabed were acquired in October 2006 during the EVA'06 trial off the Island of Elba. The free field data were collected in the NURC water tank.

4:00-4:20 Break

4:20

5pUWb8. Robust recognition and characterization of man-made objects in shallow water using time-frequency analysis. Shaun D. Anderson (Georgia Institute of Technology, Woodruff School of Mechanical Engineering, Graduate Box 1000, Atlanta, GA 30332, USA, sanderson49@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), Manell E. Zakharia (French Naval Acadamey, BP 600, 29240 Brest-Armees, France, manell.zakharia@ecole-navale.fr), Mario Zampolli (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, zampolli@nurc.nato.int), Henrik Schmidt (MIT, 77 Mass Ave, 5-204, Cambridge, MA 02139, USA, henrik@mit.edu), William A. Kuperman (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, wkuperman@ucsd.edu)

For underwater sonar, time-frequency analysis, in particular Wigner-Ville analysis, has been shown to be a relevant tool for discriminating a man made target (shell) from a natural one of the same shape (solid) and even to estimate some target characteristics (shell thickness, shear velocity...). This processing tool takes advantage of the evolutional, time dependent aspect of the echo spectrum. The estimated time-frequency patterns can be used for detection and wideband classification of sonar echoes in order to reduce false alarms. In particular, the so-called "coincidence pattern" appearing for specific frequency range is a robust time-frequency signature of man-made shells. A time-frequency analysis will be presented to understand echo formation mechanisms using a standard spherical shell model target model. The influence of the medium parameters as well as the source-receiver configuration will be investigated in free space and then extended to the case of a shallow water waveguide. The proposed approach will be tested using target scattering data collected during Experiments for Validation of Acoustic modeling techniques (EVA) sea test on the north shore of Isola D'Elba, Italy. Application to mine-hunting sonar systems will be discussed.

4:40

5pUWb9. Bi-static scattering from buried, elastic objects in shallow water waveguides. Henrik Schmidt (MIT, 77 Mass Ave, 5-204, Cambridge, MA 02139, USA, henrik@mit.edu), Deep Ghosh (MIT, 77 Mass Ave, 5-204, Cambridge, MA 02139, USA, dghosh @mit.edu)

The emerging autonomous network technology is enabling new operational paradigms for the concurrent detection, classification and localization of seabed objects by collaborating AUVs. Thus, such networks can exploit the bi-static enhancement of targets which are stealthy to conventional mono-static sonars, and the resonance properties of manmade targets. Under the GOATS and SWAMSI programs MIT in collaboration with NURC have addressed the fundamental issues associated with the development of such a new sonar concept. Through a series of joint experiements, various aspects of the interaction of elastic targets, completely or partially buried in the seabed have been investicated, including the evanescent coupling of low-frequency sound (1-10 kHz) into the seabed, the coupling with structural waves in the targets, and the 3D scattering back into the water column. The analysis is performed using a spectral virtual source scattering model with an embedded spectral Green's function generator which incorporates all multiple scattering effects between the target and the seabed. The target response is represented uniquely by an impedance matrix which may be computed separately using analytical or numerical methods, depending on the target geometry. The scattering model has been combined with the OASES code to provide a comprehensive simulation environment including all the shallow water waveguide physics.[Work Supported by ONR].

Contributed Paper

5:00

5pUWb10. Full field modeling of multiaspect scattering from buried objects. Ilkka Karasalo (FOI, Gullfossgatan 6, SE 16490 Stockholm, Sweden, ilkka.karasalo@foi.se)

Results are presented from a computational study of imaging of objects buried in the seabed under shallow water, using a rail-mounted active sonar and synthetic aperture processing. The medium is modeled as range independent, composed of a shallow water layer above a seabed of a muddy sediment containing the objects, and deeper subbottom layers. The objects have simple shape and structure, with diameters in the range 15-195 cm. From selected positions along the rail, the sonar insonifies the objects by LFM pulses from a directive transmitter, and records the backscattered echoes with a horizontal uniform line array (ULA), or optionally a pair of vertically separated ULAs. The signals from all sonar positions are integrated coherently by synthetic aperture processing for enhanced azimuthal resolution in the images of the objects. The parameters of the models of the medium, the objects and the experimental geometry are chosen to approximate those of a sea trial conducted in the Stockholm archipelago in 2004. The model-predicted scattered field is computed using the XFEM-S code, based on a frequency-domain boundary integral equation (BIE) formulation of scattering from a smooth object in a layered fluid-solid medium. Comparisons of model predictions with experimental results are presented.

Invited Papers

5:20

5pUWb11. Broadband scattering from spherical shells in a waveguide: modeling and classification. John A. Fawcett (DRDC Atlantic, PO Box 1012, Dartmouth, NS B2Y 3Z7, Canada, john.fawcett@drdc-rddc.gc.ca)

In this presentation, the exact expression for scattering from a sphere in a Pekeris waveguide (e.g., Sammelmann and Hackman, J. Acoust. Soc. Am., 82, 1987) is discussed. The importance of the sphere/interface rescattering terms is considered. A computationally faster multipath expansion approach is derived and its accuracy compared with the exact approach. For sufficiently high frequencies and in the case where the rescattering terms can be ignored, the multipath approach yields accurate predictions. The broadband scattering from an elastic-shelled sphere in a Pekeris waveguide is considered as a function of the frequency (or time for a pulse) and the sphere's range and depth in the waveguide. The classification problem is also discussed. The echos (time series or spectra) from a large set of spheres with varying parameters are generated and grouped into six classes corresponding to the various shell thicknesses and materials. Simple classifiers based upon temporal or spectral representations are considered for the spheres in free space and in a waveguide

5:40

5pUWb12. Backscattering from an elastic target near a water-sediment interface at oblique incidence: First results of tank experiments. Jean-Pierre Sessarego (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, sessarego@lma.cnrs-mrs.fr), Anatoliy N. Ivakin (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, USA, ivakin@apl.washington.edu), Régine Guillermin (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, guillermin@lma.cnrs-mrs.fr)

Sound scattering from a target situated near a water-sediment interface was studied in laboratory conditions in order to control separately all the parameters involved in the scattering process. Targets of different sizes were ensonified with wide band transducers covering the frequency range 200 kHz to 1 MHz. First, the target scattering strength was measured in the free space conditions, and the scattering strength of the water-sediment interface was measured at oblique incidence. These characteristics were used to provide a rough estimate for the signal-to-noise ratio for the second set of experiments where the target was situated near the interface to study effects of target-boundary interactions. The intensity of the total scattered field was measured as a function of the beamwidth, transducer/object and object/interface distances, frequency, grazing angle, target size and the interface roughness parameters. The interface considered here is a flattened sand surface which was studied earlier [Ivakin and Sessarego, High frequency scattering from flattened sand sediments: effects of granular structure, J. Acoust. Soc. Am., **122**, (5) 2007]. The targets were spherical glass beads of different size. Side scan sonar images are presented and possibilities of their qualitative interpretation are discussed.

Contributed Paper

6:00

5pUWb13. Estimates of scattering strength for buried cylindrical targets ensonified by evanescent waves. David C. Calvo (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, david.calvo@nrl.navy.mil), Mario Zampolli (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, zampolli@nurc.nato.int), Alessandra Tesei (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, tesei@nurc.nato.int)

It is known that low-frequency subcritical sound waves can significantly scatter from targets buried in a seabed due to the significant penetration depth of the incident evanescent wave. Past computational work on scattering by buried spherical shells has been done, for example, using a T-matrix Method [R. Lim et al., J. Acoust. Soc. Am. **93**, 1762-1783 (1993)], a Virtual Source Method [I. Lucifredi and H. Schmidt., J. Acoust. Soc. Am. **120**, 3566-3583 (2006)], or finite-element methods [Zampolli et al., J. Acoust. Soc. Am., in press]. In addition to high-fidelity results which are expected from the preceding numerical methods, it desirable to have approximate analytical/asymptotic predictions of multistatic scattering strength for a variety of homogeneous or layered buried targets. Focusing on buried cylinders of infinite or finite-length, we first compute scattering using an approximate method that makes use of separation of variables and neglects multiple scattering between the interface and the target. Results are compared with those generated using the Axiscat/NURC/COMSOL finite-element method. Asymptotic estimates are then presented for scattering strength for objects completely buried in the seafloor for a flat interface (Work sponsored by ONR and NURC.)

FRIDAY AFTERNOON, 4 JULY 2008 ROOM 341, 2:00 TO 6:00 P.M.

Session 5pUWc

Underwater Acoustics and ECUA: Automatic Target Recognition, Sensors and Algorithms

Gerald Dobeck, Cochair
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School of Engineering and Physical Sciences, Heriot-Watt University, Edinburgh, EH14 4AS, UK

Invited Paper

2:00

5pUWc1. A track-before-detect algorithm for active sonar based on a hidden Markov model. Nigel H. Parsons (Thales Underwater Systems Ltd., Dolphin House, Ashurst Drive, Bird Hall Lane, Cheadle Heath, SK3 0XB Stockport, UK, nigel.parsons@uk.thalesgroup.com)

An active sonar track-before-detect algorithm is described. It is based on a hidden Markov model which uses a Viterbi algorithm to estimate the log-likelihood ratio of the presence or absence of a target in tracks within a state space representing a set of ranges, bearings, range rates and bearing rates, assuming a set of transition probabilities of changes in range rate and bearing rate. A detection is declared if the log-likelihood ratio exceeds a certain threshold and subsequently an HMM tracker, operating on a much smaller state space, is then employed. The performance of this algorithm on simulated data is evaluated. It is shown that, for moving and manoeuvring targets, the detection performance is significantly better than that of a conventional algorithm.

Contributed Paper

2:20

5pUWc2. Forward looking techniques for environment modeling, obstacle detection and characterization. Isabelle Quidu (ENSIETA - E3I2 Lab., 2 rue François Verny, 29806 Brest Cedex 9, France, isabelle.quidu@ensieta.fr), Yann Dupas (Groupe d'Etudes Sous-Marines de l'Atlantique (GESMA), BP 42, 29240 Brest Armées, France, yann.dupas@dga.defense.gouv.fr)

Military underwater robots are designed to perform complex underwater missions in both known and unknown environments. To achieve these tasks, an Autonomous Underwater Vehicle (AUV) must be supplied with appropriate sensors to deal with unpredictable events that can put it in danger, and with a high degree of decisional autonomy. In this paper, we have studied the architecture of forward looking sensors to allow the creation of a 3D model of the environment presenting the seabed and the obstacles that are on the path of the AUV. Our approach is based on experimental trials using different and complementary ways (sonars with several configurations) to gather an information as complete as possible. This information will be processed by the vehicle during a survey mission. Practically, we create a reference model of a static environment using a multibeam system which produces bathymetric images at different grazing angles. In the same environment we then use a Forward Looking Sonar intended for the recognition of detected echoes in comparison with the reference model. If an echo cannot be related to a known object on the reference map, it is considered as an obstacle, and the map is updated.

Invited Paper

2:40

5pUWc3. Automated change detection with area matching. John Dubberley (Naval Research Laboratory, Bldg. 1005 Rm D-23, Stennis Space Center, MS 39529, USA, john.dubberley@nrlssc.navy.mil), Marlin Gendron (Naval Research Laboratory, Bldg. 1005 Rm D-23, Stennis Space Center, MS 39529, USA, marlin.gendron@nrlssc.navy.mil), Maura Lohrenz (Naval Research Laboratory, Bldg. 1005 Rm D-23, Stennis Space Center, MS 39529, USA, maura.lohrenz@nrlssc.navy.mil)

When resurveying a geographic area of the seafloor during sidescan change detection operations, an automated method to match bottom objects imaged previously with objects imaged in the resurvey can increase efficiency and accuracy. The geographic position of a new object relative to a historical object is a good indicator of a match. However, due to position error within either survey, there may be more than one spatially-close object in the new imagery. To complicate matters further, the reflected energy from the new object may be significantly different given a different incidence angle in the resurvey or the partial burial of the object. In addition, the resurveyed object image may be below the threshold set for automatic recognition and falsely eliminated. This presentation will address these problems and suggest possible methods for matching “constellations” of bottom objects by Dijkstra’s minimum cost - maximum flow algorithm, control point matching, and the data-association procedure.

Contributed Paper

3:00

5pUWc4. An acoustic barrier based on amplitude variations of the ray paths and double beamforming. Barbara Nicolas (GIPSA-lab, dep. DIS, 961, rue de la Houille Blanche, 38402 St Martin d’Hères, France, barbara.nicolas@gipsa-lab.inpg.fr), Philippe Roux (LGIT - CNRS - Université Joseph Fourier, Maison des Géosciences, 1381 rue de la Piscine, BP 53, 38041 Grenoble, France, philippe.roux@obs.ujf-grenoble.fr), Ion Iturbe (GIPSA-lab, dep. DIS, 961, rue de la Houille Blanche, 38402 St Martin d’Hères, France, ion.iturbe@gipsa-lab.inpg.fr), Jérôme I. Mars (GIPSA-lab, dep. DIS, 961, rue de la Houille Blanche, 38402 St Martin d’Hères, France, jerome.mars@gipsa-lab.inpg.fr)

The objective of this work is to build an acoustic barrier to detect and localize a target between two vertical arrays of sensors. To perform this detection/localisation, we record the signal between each source (of the source array) and each reveiver (of the receiver array). Using these data, we extract the different ray paths between sources and receivers thanks to a new signal processing method: double beamforming. Then, we show that ray paths and their arrival times are not affected by a target in the medium but that ray amplitudes change. As a result, it is possible to use amplitude variation of the rays to find the target localisation. To validate these methods we perform ultrasonic experiments in a tank. These experiments are often used in underwater acoustics as they emulate shallow water waveguides: indeed, by multiplying the frequency by a factor x, distances are divided by the same factor. As acoustic and elastic propagation properties are not affected by this scaling down, it is possible to achieve “oceanic experiments” in a simple tank. Results of double beamforming and target detection are shown.

Invited Papers

3:20

5pUWc5. Advanced ATR techniques based on High-Resolution SAS Sensors. Enrique Coiras (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, coiras@nurc.nato.int), Johannes Groen (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, groen@nurc.nato.int), Benjamin Evans (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, evans@nurc.nato.int), Marc Pinto (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, pinto@nurc.nato.int)

Automatic Target Recognition (ATR) is a key element of expeditionary Mine Countermeasures (MCM) and port protection operations. Most existing approaches to ATR are currently based on high resolution sonar sensors, which provide enough information to obtain satisfactory detection and classification performance for the large World War 2 mine types (e.g., 2 m long cylinder). False alarm rates, however, are still unacceptably high for modern mines, which constrains the way operations are undertaken and often requires either confirmation or re-evaluation by a human operator. The introduction of new AUV-mounted Synthetic Aperture Sonars (SAS) increases the resolution, quality and range of acquired sonar images, which broadens the set of machine vision and computer image analysis techniques that can be used for underwater ATR operations. In this paper we study the impact that the increased quality and resolution have on performance gains and false alarm reduction. A number of classification algorithms are selected to represent the pool of existing approaches to target detection and classification, and their performances are estimated using both simulated and real image data in order to quantify the benefits associated to the new SAS technology. Evolution and near-future plans are discussed, introducing emerging bio-sonar sensors, anomaly detectors and autonomous AUV systems.

3:40

5pUWc6. Rapid distinction of dumpsite objects using Multiple-Aspect Scattering - Results from scaled tank experiments. Philippe Blondel (University of Bath, Department of Physics, Claverton Down, BA2 7AY Bath, UK, pyspb@bath.ac.uk)

Toxic dumpsites on the seafloor are causing increasing environmental concern, but traditional sonar imaging strains to distinguish objects in unconsolidated sediments, in particular in cluttered terrains. Scaled tank experiments were conducted with four different cylinders (fluid-filled and solid aluminium, air-filled and solid stainless steel, respectively) and two seabed types (silt and gravel), using the facilities at the University of Bath. The setup was a 10:1 scaled version of the EC-SITAR sea trials site in the Stockholm Archipelago (Sweden). The main aim of these experiments was to design efficient surveying strategies, later used at sea. Our studies showed large variations depending on the aspect of these targets and their bistatic imaging configuration. These variations can be directly related to the shapes of the targets (e.g., dimensions, presence of ribs), their content (hollow or solid) and the material of the shells (e.g., stainless steel or aluminium). They are quantified using the combined L4 norm of the time-domain signals at each aspect. Using appropriate ranges of multistatic configurations and imaging each target at three distinct aspects (45° apart), it is possible to successfully distinguish between similar targets with distinct contents and/or material, even in cluttered terrains.

4:00-4:20 Break

Contributed Papers

4:20

5pUWc7. Target detection of man made objects in side scan sonar images - segmentation based false alarm reduction -. Max Neumann (Freie Universität Berlin, Takustr. 9, 14195 Berlin, Germany, papperlapapp@gmail.com), Christian Knauer (Freie Universität Berlin, Takustr. 9, 14195 Berlin, Germany, christian.knauer@inf.fu-berlin.de), Bodo Nolte (Forschungsanstalt der Bundeswehr für Wasserschall und Geophysik, Klausdorfer Weg 2-24, 24148 Kiel, Germany, bodonolte@bwb.org), Dieter Brecht (Forschungsanstalt der Bundeswehr für Wasserschall und Geophysik, Klausdorfer Weg 2-24, 24148 Kiel, Germany, dieterbrecht@bwb.org), Wolfgang Jans (Forschungsanstalt der Bundeswehr für Wasserschall und Geophysik, Klausdorfer Weg 2-24, 24148 Kiel, Germany, wolfgangjans@bwb.org), Alfons Ebert (FGAN-FOM, Gutleuthausstraße 1, 76275 Ettlingen, Germany, a.ebert@fom.fgan.de)

This paper presents a fast and robust algorithm for significantly reducing the number of false detections caused by screening algorithms for side scan sonar (SSS) images. The presented algorithm consists of two processing steps. First, an iterative segmentation process is carried out for seperating the image into shadow and background. This segmentation is based on an energy function which combines the local neighborhood segment information and the amplitude of a pixel. By minimizing this function, a clear shadow, the most significant target characteristic, will be extracted. Second, based on the region of interest (ROI) and the shadow contour, a robust classification approach is applied, utilizing the area of the shadow, the first two statistical moments of the pixel amplitude and the existence of parallel lines (Hough transformation). This algorithm was tested using a data set with approx. 2400 ROIs containing about 200 targets and 270 targetlike stones or sandrippel. This data set was gathered during five different measurement campaigns in the Baltic Sea and the Mediterranean Sea using three different SSS systems Benthos C3D, Klein2000 and Marine Sonics). These data were collected by FWG and WTD71 as well as by Atlas Electronik with the SeaOtter MK1 AUV.

4:40

5pUWc8. Sonar target-phase measurement and effects of transducer-matching. Philip Atkins (University of Birmingham, Department of Electronic, Electrical and Computer Engineering, Edgbaston, B15 2TT Birmingham, UK, p.r.atkins@bham.ac.uk), Alan Islas (University of Birmingham, Department of Electronic, Electrical and Computer Engineering, Edgbaston, B15 2TT Birmingham, UK, AXI743@bham.ac.uk), Kenneth G. Foote (Woods Hole Oceanographic Institution, Woods Hole, MA 02543, USA, kfoote@whoi.edu)

Active sonar systems normally detect and classify a target based on the amplitude of the received echo or the induced Doppler shift. However, additional classification information may be available from the phase shift introduced by some targets as a result of the boundary conditions. For example, reverberation from the sea surface and scattering from fish swimbladders introduce an additional phase shift that may not be present in returns from an acoustically stiffer seabed or synthetic target. Algorithms based on the use of subband correlators are presented for measuring the phase shifts introduced by the boundary conditions on stationary and moving targets when insonified by broadband transmissions. These techniques are used to remove the phase shifts introduced by the unknown target. However, the unknown phase characteristics of the transducer, matching circuit, and electronic circuitry of a sonar system imply that target-phase measurements are very difficult to make in any practical system. The effects of adding a Butterworth-derived matching circuit to a Reson TC2130 transducer are presented for the case of sinusoidal frequency-modulated excitation of solid elastic and thin elastic-shelled hollow spheres. It is concluded that target-phase measurements can enhance the classification performance of a suitably calibrated sonar system.

Invited Papers

5:00

5pUWc9. Evaluation of portable high-frequency sonars for diver identification. Anna Crawford (Defence R&D Canada Atlantic, P.O. Box 1012, 9 Grove St, Dartmouth, NS B2Y 3Z7, Canada, anna.crawford@drdc-rddc.gc.ca), Vance Crowe (Defence R&D Canada Atlantic, P.O. Box 1012, 9 Grove St, Dartmouth, NS B2Y 3Z7, Canada, vance.crowe@drdc-rddc.gc.ca), Thomas Pastore (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, Pastore@nurc.nato.int), Ronald Kessel (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, Kessel@nurc.nato.int)

Obtaining a positive identification is a critical step in most tiered harbour protection strategies for countering underwater intruders. It is generally recognised that sonar is one of the best tools for underwater imaging, however operating in a harbour environment presents challenges. As part of an on-going harbour protection research project, small easily portable high-frequency sonar systems are being investigated as a means to equip small response craft with intruder identification capability. Several systems are being considered, with the most comprehensive testing by DRDC to date being done on small Canadian-made sonars. Tests were conducted in local harbour waters in Halifax, Canada, and in La Spezia, Italy, through participation in the NATO Undersea Research Centre Response Against Diver Intrusions (RADI) joint trial, conducted in November 2007. A variety of small sonars and manned and unmanned response craft were used during the RADI trial. Evaluation of the performance of these devices for the task of diver identification in realistic conditions will be discussed.

5:20

5pUWc10. Features for propagation-invariant classification of underwater targets. Patrick Loughlin (University of Pittsburgh, 348 Benedum Engineering Hall, Dept. of Electrical & Computer Engineering, Pittsburgh, PA 15261, USA, loughlin@engr.pitt.edu), Greg Okopal (University of Pittsburgh, 348 Benedum Engineering Hall, Dept. of Electrical & Computer Engineering, Pittsburgh, PA 15261, USA, gno1@pitt.edu)

As sound propagates in shallow water, it is subject to frequency-dependent spreading and attenuation (dispersion and damping). In active sonar, these propagation-induced changes can be detrimental to automatic classification because the observed backscatter depends on the propagation environment and how far the wave has traveled. One way to address this problem is to develop propagation-invariant

features of the wave that can be used in automatic classification. In this talk, we present temporal, spectral, and cepstral moment-like features of a wave that are invariant per mode to dispersion and damping. Classification results on numerical simulations of the back-scatter from different steel shells propagating in a Pekeris waveguide with damping and random variations will be presented. [Supported by ONR grants N00014-06-1-0009 and N00014-07-10355]

Contributed Paper

5:40

5pUWc11. MCM sensor requirements: performance measures. Samantha Dugelay (Dstl, Building A32, Winfrith Technology Centre, DT2 8WX Dorchester, UK, sdugelay@dstl.gov.uk)

This paper presents a comparison of two measures of performance suitable to characterise operational performance of mine hunting sonars in various environmental conditions. The first measure is a direct adaptation of information theory and bounds the capability of a sonar to distinguish between objects, i.e., a classification capability. The bounds take into account the amount of distinguishable pixels between objects and the statistical information content of each pixel. Successfully applied in Radar, this approach has also demonstrated the performance of synthetic aperture sonar at NURC using a Rayleigh distribution for statistical pixel distribution. The measure is now being further developed to include high resolution distributions such as K-law which readily appear in high resolution images. The second measure, Johnson's criteria aims to provide guidance on resolution required for operator detection, classification and identification. This measure originally derived from experiments on night vision images is now being modified to simultaneously incorporate target highlight and shadow information in varying environmental conditions. Finally, the predictions of these measures will in the future be compared to each other and to current system performances.

FRIDAY AFTERNOON, 4 JULY 2008 — ROOM 342B, 2:20 TO 3:40 P.M.

Session 5pUWd

Underwater Acoustics and ECUA: Acoustic Data Fusion

Eric Maillard, Cochair
Reson Inc, Golet, CA, USA

Benoit Zerr, Cochair
DGA/GESMA, BP42, Brest Armées, 29240, France

Contributed Papers

2:20

5pUWd1. Multisegmentation of sonar images using belief function theory. Mounir Dhibi (ENSIETA E3I2, 2 rue François Verny, 29806 Brest Cedex 9, France, mounir.dhibi@ensieta.fr), Romain Courtis (I2ETA Romain Courtis, GESMA/SDP/GDM, BP 42, 29240 Brest Armées, France, Romain.Courtis@dga.defense.gouv.fr), Arnaud Martin (ENSIETA E3I2, 2 rue François Verny, 29806 Brest Cedex 9, France, arnaud.martin@ensieta.fr), Isabelle Quidu (ENSIETA - E3I2 Lab., 2 rue François Verny, 29806 Brest Cedex 9, France, isabelle.quidu@ensieta.fr)

Today side scan sonar is one of the most efficient sensors for Rapid Environment Assessment missions. Unfortunately, features extracted from a given area are strongly dependent on the relative position of the sensor (e.g., due to the shadow or the gain variation). That could conduct to a bad segmentation of the seabed. However, due to the fact that operational systems give very often multiple views of the same area we use the redundancy. In this work, we propose to fuse multiview segmentations in order to outperform the seabed classification. First we present a way to characterize the seabed using as a start point, a texture analysis in order to extract parameters on images. Then, a classification method allows allocating a class according to the type of sediment for the different standpoints. The proposed classifier fusion is based on the belief function theory. We present results from a set of experiments conducted to evaluate the proposed approach with real sonar images and we discuss them.

2:40

5pUWd2. Elimination of corner-turning in FFT-based sonar array beamforming. Jacob Barhen (Oak Ridge National Laboratory, 1 Bethel Valley Road, Oak Ridge, TN 37831-6015, USA, barhenj@ornl.gov), Travis Humble (Oak Ridge National Laboratory, 1 Bethel Valley Road, Oak Ridge, TN 37831-6015, USA, humblets@ornl.gov), Michael Traweek (Office of Naval Research, 875 North Randolph Street, Arlington, VA 22203, USA, Mike.Traweek@navy.mil)

The expected availability, in the near future, of an ultralow power version of the revolutionary IBM CELL multicore processor opens unprecedented opportunities for implementing sophisticated signal processing algorithms faster and within a much lower energy budget. The concept of "corner turning" has been, for many decades, at the heart of array beamforming via Fourier transforms. As widely reported in the open literature (both for sonars and radars), the computational sequence involving corner turning operations, i.e., the sequence: temporal Fourier transforms --> data cube corner turning --> spatial Fourier transforms, constitutes one of the primary obstacles to achieving high-performance and lower power dissipation (by reducing the number of times memory is accessed). Even with the emergence of novel multicore processors, leading providers (e.g., Mercury Computers) still include explicit corner turning stages in their computational flow architectures for multidimensional array processing. The primary innovation reported in this paper addresses the development of a computational scheme that avoids altogether the corner turning stage. We discuss its implementation on currently available CELL technology (65 nm SOI) and demonstrate close to an order of magnitude speed-up compared to the scheme with corner turning implemented on the same processor.

3:00

5pUWd3. Multiple-sensor fusion approach to seabed classification.
Benoit Zerr (DGA/GESMA, BP42, 29240 Brest Armées, France, benoit.zerr@dga.defense.gouv.fr), David Kerneis (ENST Bretagne Dept. ITI, Technopôle Brest-Iroise - CS 83818, 29238 Brest Cedex 3, France, david.kerneis@enst-bretagne.fr), Basel Solaiman (ENST Bretagne Dept. ITI, Technopôle Brest-Iroise - CS 83818, 29238 Brest Cedex 3, France, basel.solaiman@enst-bretagne.fr)

Seabed classification is key issue for civilian and military underwater applications, from offshore exploitation to mine counter measure. Most of the existing automated classification techniques relies on the analysis of the data provided by a single sensor, supposed to unambiguously separate the different classes of seabed. In this paper we present a different approach which considers that, even if a sensor cannot tell the differences between two classes, classification will improve by considering that the seabed belongs to one of these two classes, and, further, that the analysis of the data from another sensor can resolve the ambiguity. For each sensor, the classification is achieved in a conventional way by feature extraction and supervised classification. The fusion of the results implements the theory of evidence through Dempster-Shafer method. After a description of the method, the paper discusses the experimental results from the fusion of information delivered by three sensors: an imaging sidescan sonar, a vertical echo sounder and an interferometric bathymetric sonar. The major part of the experimental data has been acquired by towed or hull mounted sensors. As these sensors are a subset of the payloads operated simultaneously by the new DGA-SHOM DAURADE AUV, preliminary seabed classification results in covert REA missions will also be presented and discussed.

3:20

5pUWd4. Acoustic data fusion devoted to underwater vegetation mapping.
Claire Noel (Semantic TS, 39 Ch Buge, 83110 Sanary, France, noel@semantic-ts.fr), Christophe Viala (Semantic TS, 39 Ch Buge, 83110 Sanary, France, viala@semantic-ts.fr), Michel Coquet (Semantic TS, 39 Ch Buge, 83110 Sanary, France, coquet@semantic-ts.fr), Benoit Zerr (DGA/GESMA, BP42, 29240 Brest Armées, France, benoit.zerr @dga.defense.gouv.fr), Thierry Perrot (CEVA, Presqu'île de Pen Lan BP3, 22710 Pleubian, France, thierry.perrot@ceva.fr)

This paper presents research tasks conducted by SEMANTIC TS, in collaboration with GESMA, aimed to develop a mapping method for underwater vegetation lying on seabed. First stage is to develop a method for detecting and characterizing vegetation on the seabed using the acoustic response from a conventional single beam echo sounder. This new method is then operated simultaneously with multibeam sonar producing micro-relief information and side scan sonar providing gray scale levels associated with bottom reflectivity. Then fusion of these three data is processed. We show efficiency of these multisensor data fusion concept to get very precise seabed vegetation mapping in a way reducing truth control (video and diving investigations). Sensors and method accuracy allow obtaining, like in biomedical field, real 3D scan pictures of seabed vegetation. This study is first applied to posidonia and cymodocea, which play a key role in Mediterranean's echosystem. Then, extension of the method is investigated to address laminaria which may significantly affect the performance of acoustic and optical sensors used for sea-mines detection and this paper presents results of data fusion mapping on an Atlantic sea area covered by luminaria, studied and well known by the CEVA.

FRIDAY AFTERNOON, 4 JULY 2008 ROOM 342B, 4:00 TO 6:20 P.M.

Session 5pUWe

Underwater Acoustics and ECUA: Noise Suppression, Robust Direction of Arrival, and Target Strength Estimation

Georges Bienvenu, Cochair
Thales Naval Division, Thales Underwater Systems, France

Ronald A. Wagstaff, Cochair
National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA

Contributed Papers

4:00

5pUWe1. "Venus's-noisetrap" high-gain adaptive processor.
Ronald A. Wagstaff (National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA, rwagstaf@olemiss.edu), Heath E. Rice (National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA, herice@olemiss.edu)

The Venus's-noisetrap is a high gain adaptive signal processor inspired by Venus's-flytrap, an insect eating plant. The Venus's-noisetrap utilizes single sensor or beamformed spectral data. It "traps" the data samples by modifying the governing equations in a manner that each spectral sample is adaptively "trapped" before averaging. The trapping process is a fluctuation-based temporal coherence determination of whether the time history in each spatial or spectral bin is signal or noise. Signals are set "free", while the noise remains trapped, blocking it from participating in the averaging process. For ocean acoustic data, the processor automatically identifies signals from submerged sources. In addition, the Venus's-noisetrap achieves large signal-to-noise ratio (SNR), high spectral and spatial resolution, and auto-identification of signals. The techniques that cause this processor to mimic the Venus's-flytrap will be discussed, and the method of continuously adapting the governing equations to the unpredictable signal and noise environment will be illustrated. Results will be presented to show the processor's large SNRs and corresponding enhancements in spectral and spatial resolution.

4:20

5pUWe2. Suppression of oceanic reverberation by subspace methods.
Xuan Li (Institute of Acoustics, Chinese Academy of Sciences, 100080 Beijing, China, happyxuanli@hotmail.com), Xiaochuan Ma (Institute of Acoustics, Chinese Academy of Sciences, 100080 Beijing, China, maxc@mail.ioa.ac.cn), Chaohuan Hou (Institute of Acoustics, Chinese Academy of Sciences, 100080 Beijing, China, hch@mail.ioa.ac.cn)

Acoustic signal processing in shallow water environments is a challenging problem because of the presence of reverberation. Based on some models of reverberation, data from sensors array is pretreated to suppress reverberation. Considering reverberation as a sum of echoes of transmitted

signal, the principal component inverse (PCI) algorithm deletes the largest singulars of data matrix, which is constructed from array data. However, estimating a threshold which is needed in PCI is difficult in practice. In this paper, two new subspace methods, Deleting Big Eigenvalues and Subspace Projection are proposed. The two novel methods, substituting automatic signal-number estimation for threshold estimation, are operated via eigendecomposition. According to a simulation which takes broadband linear modulated frequency signal as transmitted signal, these two methods show a similar performance but smaller computing quantity compared with PCI.

4:40

5pUWe3. Coherent effects of flow- and pressure-hull of a generic submarine on target scattering in an active sonar performance model. Pieter Schippers (TNO-D&V-Underwater Technology, Oude Waalsdorperweg 63, Post Box 96864, 2509 JG The Hague, Netherlands, pieter.schippers @tno.nl)

Since the late eighties the sonar performance model ALMOST for active and passive sonar is under development at TNO. For active detection performance, first a point target was used, with a single target strength value dependent on parameters like aspect angle, based on measurements or other sources. However by now there is a demand for TS of ships and wakes with realistic dimensions and characteristics. A generic sub was modeled with additional software routines, as a pixel file. A newly developed time domain model for hull reflection was implemented, also using scattering pixels, assuming multiple scattering with damping in the metal hull layer. This extension of the pixel modelling is evaluated versus literature data. Some modeling results of target strength computations are shown, for a generic submarine with pressure hull, with aspect angle, frequency and bandwidth as parameters.

5:00

5pUWe4. Robust capon beamformer for port/starboard discrimination of twin-line array. Zaixiao Gong (National Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, NO.21, Bei-Si-huan-Xi Road, 100080 Beijing, China, gzx@mail.ioa.ac.cn)

Compared with the single-line array, the twin-line array have the potential to solve the port/starboard discrimination problem. Conventional method of geometric phase shifting has been used to distinguish the port/starboard of the target. But it can only be used to solve the problem when the frequency bandwidth is limited. The method based on optimum beamforming for hydrophone triplets can hardly be applied with the twin-line arrays. Aimed at the twin-line arrays port/starboard discrimination problem, a method based on the robust capon beamformer is proposed in this paper, which has advantages of concise algorithm and robustness against the aberration of the array shape. And the method works better with wider frequency bandwidth. Simulation and on sea experiment data are analyzed to verify the method.

5:20

5pUWe5. A numerical method for array sensor noise field calculation in detection performance optimization. Chao Sun (Institute of Acoustic Engineering, Northwestern Polytechnical University, 710072 Xi'an, China, csun@nwpu.edu.cn), Bo Yang (Institute of Acoustic Engineering, Northwestern Polytechnical University, 710072 Xi'an, China, yangbo@mail.nwpu.edu.cn), Yixin Yang (Institute of Acoustic Engineering, Northwestern Polytechnical University, 710072 Xi'an, China, yxyang @nwpu.edu.cn)

Different from the generally adopted criteria of minimizing the sidelobe level and maximizing the array gain in the weighting design of a sonar array, an approach was proposed recently to optimize the shading weights with the aim of maximizing the deflection coefficient in the square-law detector, which in essence suppresses the self-noise by including the noise information at array sensors in the optimization procedure. When several noise sources are present and/or an analytical expression of the noise transfer function is not available, the sensor noise needs to be measured in forming the sensor noise response cross correlation matrix required in the optimization which is very demanding when an array with large number of sensors is considered. To avoid the tedious work of noise measurement, a numerical method is developed in this paper. In this method, the main self-noise sources are assigned with different positions and strengths and the noise field at the array sensors is calculated via either the finite element method or the boundary element method. Tank experiment validated the effectiveness of this method. By using this method, the detection performance of a sensor array can be predicted before it is physically built and tested in under practical conditions.

5:40

5pUWe6. Improving spatial resolution of interferometric bathymetry in multibeam echosounders. Gerard Llort-Pujol (ENST-Bretagne, Dpt ITI, CS 83818, 29238 Brest Cedex 03, France, gerard.llort @enst-bretagne.fr), Christophe Sintes (ENST-Bretagne, Dpt ITI, CS 83818, 29238 Brest Cedex 03, France, christophe.sintes @enst-bretagne.fr), Xavier Lurton (Institut Français de Recherche pour l'Exploitation de la Mer, NSE/AS, BP 70, 29280 Plouzané, France, lurton @ifremer.fr)

Most multibeam echosounders used in seafloor mapping use the interferometry method for bathymetry measurement, based on the zero-crossing of the phase difference between two sub-arrays. In this approach, only one sounding is computed per formed beam, and the spatial resolution is linked to the beam footprint extent. Using the whole content of the phase difference signal vs time makes it possible ideally to get a bathymetry data sampled at the very resolution of the digitised signal. A trade-off is however to be found between this resolution improvement and the increase in the measurement noise. The performances of interferometric bathymetry are presented, together with the constraints of the resolution-accuracy trade-off, and a presentation of a high-resolution algorithm for interferometric bathymetry applicable to multibeam systems. This is illustrated by results obtained on a wreck, showing the remarkable improvement achievable in resolution over a real target. The potentialities for this method are promising, since it allows to process, in the same beam, targets on the seafloor and inside the water column; simplifications in multibeam sounders array structure and processing are also to consider.

6:00

5pUWe7. Adaptive removal of a known interference and its scattered energy. Yung P. Lee (SAIC, 4001 N. Fairfax Dr, Suite 175, mail stop 1-11-1, Arlington, VA 22203, USA, yung.p.lee@saic.com), William Lee (Duke University, Department of Electrical and Computer Engineering, Durham, NC 27708, USA, wwl3@duke.edu)

In underwater passive processing, the nearby fast moving loud surface ships produce nonstationary interference field and limit most sonar performance. After 9/11, AIS (automatic identify system) is required for ships larger than 60's, providing ground truth of positions, speeds, and other ancillary information for the nearby surface ships. This study investigates adaptive removal of a known loud signal using the ground truth information and acoustic propagation modeling. In addition, scattered energy away from the loud signal is identified through a delay correlation processing and is also removed. In a segment of MAPEX2k experiment conducted by SACLANT Centre off the west coast of Italy on November 28, 2000, the research vessel ALLIANCE towed an array at 2 m/s passing by a moored controlled source. The controlled source transmitted a loud 1-s 150-500 Hz LFM sequence every 15 s. In this setting, the fixed controlled source generated a nonstationary interference field received on the towed array. With the knowledge of the array tow track and source position the performance of removing the known interference and its scattered energy in a real shallow water environment is examined.

Session 5pUWf

Underwater Acoustics and ECUA: Sound Generation and Attenuation

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Contributed Papers

4:40

5pUWf1. Neural classification of ships hydroacoustic signatures. Andrzej Zak (Polish Naval Academy, Smidowicza 69, 81-103 Gdynia, Poland, a.zak@amw.gdynia.pl)

The paper presents method of classification of hydroacoustic signatures generated by moving ship. Classification is a procedure in which individual items are placed into groups based on quantitative information on one or more characteristics inherent in the items. In this paper the hydroacoustics signals classification is understood as the process of automatically recognition what kind of object is generating acoustics signals on the basis of individual information included in generated sounds. Hydroacoustics signal classification is a difficult task and it is still an active research area. Automatic signal classification works based on the premise that sounds emitted by object to the environment are unique for that object. However this task has been challenged by the highly variant of input signals. The paper includes discussion about unique of sound generated by moving ships. To solve problem of hydroacoustic signatures classification the Kohonen networks which belongs to group of self organizing networks where chosen. Hydroacoustic signals were acquired on the Polish Navy Range during the complex ship measurement. At the end the results of classification of underwater noises made by ship were presented.

5:00

5pUWf2. Underwater acoustic generation with narrow and broadband lasers. Theodore G. Jones (U.S. Naval Research Laboratory, Code 6795, 4555 Overlook Ave. SW, Washington, DC 20375, USA, ted.jones@nrl.navy.mil), Melissa K. Hornstein (U.S. Naval Research Laboratory, Code 6795, 4555 Overlook Ave. SW, Washington, DC 20375, USA, melissa.hornstein@nrl.navy.mil), Antonio C. Ting (U.S. Naval Research Laboratory, Code 6795, 4555 Overlook Ave. SW, Washington, DC 20375, USA, antonio.ting@nrl.navy.mil), Zachary W. Wilkes (Research Support Instruments, Inc., 4325 Forbes Blvd, Ste B, Lanham, MD 20706, USA, zachary.wilkes@nrl.navy.mil), Dennis A. Lindwall (U.S. Naval Research Laboratory-Stennis, Marine Geosciences Division, Code 7432, Stennis Space Center, Stennis, MS 39529, USA, lindwall@nrlssc.navy.mil)

Underwater laser acoustic sources, generated by intense pulsed lasers on above-water and underwater platforms, are under investigation. In a novel configuration, a tailored intense broadband laser pulse can be designed to propagate many meters underwater and compress at a predetermined remote location. Controlled compression of these optical pulses is governed by a combination of optical group velocity dispersion (GVD) and nonlinear Kerr self-focusing, resulting in photoionization, localized heating, and shock generation. Recent and ongoing experiments include near-field acoustic source characterization using lens-focused pulses of a broadband 400 nm Ti: sapphire laser, as well as 1064 nm and 532 nm narrowband YAG laser pulses. Also, the nonlinear optical Kerr index of water at 400 nm was precisely measured. Acoustic source characterization includes measurements of photoacoustic energy conversion efficiency, acoustic power spectrum, and directivity. Experimental results will be presented, and laser sources and techniques for underwater acoustic generation will be compared. [This work is supported by the U.S. Office of Naval Research.]

5:20

5pUWf3. Recent progress on the theoretical modeling of underwater acoustics induced by sonic booms. Johnson C. Wang (The Aerospace Corporation, System Planning and Engineering, El Segundo, CA 90245-4691, USA, Johnson.C.Wang@aero.org), Charles P. Griffice (The Aerospace Corporation, System Planning and Engineering, El Segundo, CA 90245-4691, USA, Charles.P.Griffice@aero.org), Adam M. Fincham (University of Southern, Aerospace & Mechanical Engineering, 854 W Downey way, Los Angeles, CA 90089-1191, USA, afincham@usc .edu), John R. Edwards (U. S. Air Force, Air Force Space Command, 483 N. Aviation Blvd., El Segundo, CA 90245-2808, USA, John.Edwards @LOSANGELES.AF.MIL), Adel A. Hashad (U. S. Air Force, Air Force Space Command, 483 N. Aviation Blvd., El Segundo, CA 90245-2808, USA, Adel.Hashad@LOSANGELES.AF.MIL)

This paper, review and status in nature, consists of three parts: (1) The salient nature and results of the published papers from an ocean sonic boom (OSB) project will be reviewed including theoretical and experimental studies of the wavy surface effect on the underwater acoustics, a three-dimensional theory of underwater acoustics and the underwater acoustics induced by a sonic boom traveling at hyper-velocity speeds. (2) New unpublished results will be reported including studies of sonic boom underwater overpressures affected by ocean finite depth and ocean stratification (variable sound speed). These studies are possible due to the derivation of a semi-similar transformation for the underwater acoustics governing equations and the application of high performance computers. (3) The preliminary results of the work-in-progress will also be discussed including three-dimensional extensions of former professor H. K. Cheng's two-dimensional wavy surface theory and laboratory verification.

5:40

5pUWf4. Influence of a resonance changer on the sound radiation of a submarine. Sascha Merz (University of New South Wales, School of Mechanical and Manufacturing Engineering, 2052 Kensington, Australia, sascha.merz@gmx.net), Roger Kinns (University of New South Wales, School of Mechanical and Manufacturing Engineering, 2052 Kensington, Australia, roger.kinns@aol.com), Nicole J. Kessissoglou (University of New South Wales, 2052 Sydney, Australia, n.kessissoglou@unsw.edu.au)

In order to reduce the excitation of the submarine hull through the shaft, a vibration attenuation system, called a resonance changer, can be implemented in the propeller/propulsion system. The effectiveness of such a system in reducing the low frequency sound radiation characteristics of a submarine is investigated. Only sound radiation due to fluctuating propeller forces, which are generated by the operation of the propeller in a nonuniform wake, is considered. These fluctuating forces are transmitted to the submarine hull through the fluid, as well as through the propeller shaft. Both

types of excitation cause hull vibration and sound radiation. The accordion modes of the pressure hull, are particularly efficient sound radiators. Parameters for the resonance changer system have been optimised previously by considering only excitation of the hull through the shaft. It is shown that the effectiveness of the resonance changer at different frequencies is modified significantly in a typical full-scale implementation, due to the sound radiation from the propeller. The effect on performance is increased further when the vibration of the propeller itself is taken into account. Therefore overall optimisation of any resonance changer system requires a comprehensive model. Some of the principal effects are explored in this paper.

6:00

5pUWf5. The research on measuring the coefficient of sound absorption in turbid seawater. Qi Li (Harbin Engineering University, College of Underwater Acoustic Engineering, 150001 Harbin, China, chizhafengyun1979@126.com), Yongwei Liu (Harbin Engineering University, College of Underwater Acoustic Engineering, 150001 Harbin, China, chizhafengyun1979@126.com)

When naval mine-hunting sonars and side-scan surveying sonars operating at 20~60 kHz are working in shallow coastal waters, the visco-thermal absorption of sound by suspended mud in the water may greatly decrease their detection properties. This kind of water is also characterized as turbid seawater. However, the research on measuring the coefficient of sound absorption in turbid seawater has little been done. The main difference between turbid seawater and clear seawater is that there are mud particles suspending in turbid seawater. It is also the main reason why sound absorption in turbid seawater is greater than that in clear seawater. In the paper, the coefficient of sound absorption at 20~60 kHz in turbid seawater has been measured by the reverberation technique. Results demonstrate that if the concentration of the mud is bellow 0.11 kg/m^3, the mud in turbid seawater doesn't cause additional absorption. When the concentration of the mud is between 0.14 and 0.49 kg/m^3, the coefficient of sound absorption in turbid seawater is as twice at least as that in clear seawater.