



## Speaker comfort and increase of voice level in lecture rooms

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## Session 4aAAa

## Architectural Acoustics: Architectural Acoustics Potpourri

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## Contributed Papers

8:00

**4aAAa1. Irregular sound absorbers work better.** Bernard Sapoval (Ecole Polytechnique, PMC-Polytechnique Route de Saclay, 91128 Palaiseau, France, bernard.sapoval@polytechnique.edu), Anna Rozanova-Pierrat (Ecole Polytechnique, PMC-Polytechnique Route de Saclay, 91128 Palaiseau, France, anna.rozanova-pierrat@polytechnique.edu), Simon Félix (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, simon.felix@univ-lemans.fr), Marcel Filoche (Ecole Polytechnique, PMC-Polytechnique Route de Saclay, 91128 Palaiseau, France, marcel.filoche@polytechnique.edu)

The diffraction and absorption of waves by a system with both absorbing properties and irregular geometry is an open physical problem. In the same time, irregular absorbers have been shown to be extremely efficient (1). A more reachable and closely related goal is the understanding of wave oscillations in confined systems containing an absorbing material with an irregular shape. From the theoretical point of view, the difficulty lies in the fact that part of the propagation occurs in a lossy material for which the wave operator is non-hermitian. It is found here that, in resonators containing an irregular shaped absorbent material, there appears a new type of localization. This phenomenon, that we call "astride" localization, describes the fact that these modes exist in both the lossless and the lossy regions. They are then both lossy and well coupled to sources in the air. A numerical computation of the time decay of acoustic energy shows that indeed sound absorbing devices work better when presenting a very irregular shape and that this is directly linked due to the existence of astride localisation. (1) Fractal Wall, product of Colas Inc., French patent N0-203404; U.S. patent 10/508,119.

8:20

**4aAAa2. A brief review on micro-perforated sound absorbers.** Christian Nocke (Akustikbüro Oldenburg, Katharinenstr. 10, 26121 Oldenburg, Germany, nocke@akustikbuero-oldenburg.de), Catja Hilge (Akustikbüro Oldenburg, Katharinenstr. 10, 26121 Oldenburg, Germany, hilge@akustikbuero-oldenburg.de), Jean-Marc Scherrer (Normalu S.A.S, Route du Sipes, 68680 Kembs, France, jmscherrer@barrisol.com)

The theory and design of microperforated panel sound-absorbing constructions have been introduced by D.-Y. Maa in 1975. Since then many variations of micro-perforated sound absorbing devices and materials have been introduced. Materials that have been used to be micro-perforated have been metal, wood, plastics and many others. A survey of different applications of micro-perforation will be presented. Examples shown in more detail are the development of stretched foils and metal as micro-perforated panels. Results of sound absorption measurements of various assemblies for normal and statistical incidence of sound will be shown and compared with theoretical predictions. Finally the potential of the micro-perforated sound absorber will be discussed.

8:40

**4aAAa3. The vibration sound absorption theory of soft materials.** Xin An Zhang (Xi'an Polytechnic University, 134#, Key Laboratory, 19 South Jin Hua Road, 710048 Xi'an, China, zxafafa@yahoo.com.cn)

During the past 2 years, a new theory has been established for soft materials that the vibration of materials brings the sound absorption, regardless whether they have pores in them or not. This theory is totally different from Classical theory, such as the Rayleigh model and the Zwicker and Kosten theory. Firstly, an empirical sound absorption coefficients formula of fibrous materials was found. Secondly, the theory sound absorption formula of thin fiber layers was obtained by the vibration sound absorption analysis and the applying of conservation law of energy. This formula agrees well with the empirical sound absorption formula of fibrous materials (mentioned above). Based on this achievement and applying classical laws of conservation of momentum and conservation of energy, the sound absorption theory formula of membrane (diaphragm) was also obtained, which have been well justified and also agree with the practice. This paper will review and discuss the main point about the vibration sound absorption theory and its establishment.

9:00

**4aAAa4. The influence of absorption on statistical distribution of free path lengths in rooms.** Dragana Sumarac Pavlovic (Faculty of Electrical Engineering, Bulevar Kralja Aleksandra 73, 11000 Belgrade, Serby, dsumarac@etf.bg.ac.yu), Miomir Mijic (Faculty of Electrical Engineering, Bulevar Kralja Aleksandra 73, 11000 Belgrade, Serby, emijic@etf.bg.ac.yu)

The ray tracing method in sound field simulation in a room allows the calculation of the free path lengths distribution (FPL) by superimposing all rays paths generated in the analysis. In room acoustics the statistical theory of sound field defined this distribution as an indicator of room geometrical properties which are independent of absorption. Accordingly, some commercial softwares for ray tracing simulation allow user to calculate that global distribution of FPL only. This paper is concerned with the analysis of the changes in the shape of FPL distribution as a consequence of nonuniform arrangement of absorption in room. Particularly is analysed the impact of absorptive auditorium in different global forms of the halls where all other interior surfaces are acoustically hard.

9:20

**4aAAa5. The influence of geometrical features of rooms on their acoustic response - insight based on measurements in physical models.** Miomir Mijic (Faculty of Electrical Engineering, Bulevar Kralja Aleksandra 73, 11000 Belgrade, Serby, emijic@etf.bg.ac.yu), Dragana Sumarac Pavlovic (Faculty of Electrical Engineering, Bulevar Kralja Aleksandra 73, 11000 Belgrade, Serby, dsumarac@etf.bg.ac.yu)

In a previous paper by the same authors, an analysis of the influence of geometrical features of rooms on their acoustic response was presented (Acta Acustica, Vol 93, 2007, 1012-1026). The analysis was based on ray tracing simulation. The results have shown that geometrical characteristics of rooms do influence reverberation time, and this influence is realised by the structure of sound energy paths through the room. It was concluded that

room geometry influences reverberation time both at macro and micro levels, and this influence cannot be separated. To validate these conclusions in a real sound field, additional measurements have been conducted in specially prepared physical models of rooms. In several models of different shapes selected from previous results, scaled 1:10, the changes of scattering were introduced by appropriate modifications of interior surfaces. The results have confirmed the previous study conclusions.

9:40

**4aAAa6. Room acoustics prediction based on multiple linear regressions and artificial neural networks.** Maria Ribeiro (FEUP/DEC, Rua Dr. Roberto Frias, 4200-465 Porto, Portugal, [mrbeiro@fe.up.pt](mailto:mrbeiro@fe.up.pt)), Fernando Martins (FEUP/DEC, Rua Dr. Roberto Frias, 4200-465 Porto, Portugal, [fgm@fe.up.pt](mailto:fgm@fe.up.pt))

Room acoustic quality is known to be dependent on several objective variables that are expected to be well correlated with subjective impressions of the room acoustics as judged by musical experts. Although subjects have different preferences and overall judgments are based in different criteria, it can be said that listeners in their subjective evaluation would give preference to reverberance, clarity, intimacy or spaciousness attributes. In this study, reverberance and clarity, expressed respectively by T30/EDT and C80 objective values, were predicted by multiple linear regression and artificial neural networks using normalized original data and principal components as dependent variables. The results obtained by these approaches were compared with predicted values using a computer simulation program based on the physics of ray-tracing and with measured data. Room acoustical quality was also evaluated based on preferred values as suggested by some researchers.

10:00-10:20 Break

10:20

**4aAAa7. Minimum BRIR grid resolution for dynamic binaural synthesis.** Alexander Lindau (Department of Audio Communication, Technical University of Berlin, Sekr. EN-08, Einsteinufer 17c, 10587 Berlin, Germany, [alexander.lindau@tu-berlin.de](mailto:alexander.lindau@tu-berlin.de)), Hans-Joachim Maempel (Department of Audio Communication, Technical University of Berlin, Sekr. EN-08, Einsteinufer 17c, 10587 Berlin, Germany, [hans-joachim.maempel@tu-berlin.de](mailto:hans-joachim.maempel@tu-berlin.de)), Stefan Weinzierl (Department of Audio Communication, Technical University of Berlin, Sekr. EN-08, Einsteinufer 17c, 10587 Berlin, Germany, [stefan.weinzierl@tu-berlin.de](mailto:stefan.weinzierl@tu-berlin.de))

Binaural synthesis of acoustical environments is based on binaural room impulse responses (BRIRs) measured with a certain grid of spherical coordinates separated by angles of typically between 1° and 15°. The resulting spatial resolution defines the size of the BRIR database as well as the duration of its measurement. Perceptual evaluations of datasets with a different spatial resolution using HRTFs (anechoic case) have been reported from. Most of these studies use the localization performance of listeners as a criterion; a more sensitive measurement for slight degradations in audio quality can be expected from a criterion-free comparison of datasets with different angular grids. Thus, to determine the minimum grid resolution required for dynamic binaural synthesis a listening test was performed. Following an adaptive 3AFC procedure, the spatial resolution of a recorded BRIR dataset was gradually lowered from a maximum of 1°/1° resolution until audible artefacts were introduced. This was done for a sound source located at 0°/0° presented with dynamical auralization in two degrees of freedom. To test for interaction effects the thresholds were derived independently for azimuth and elevation. The datasets used were acquired in an anechoic environment and in two rooms of different size and reverberation time.

10:40

**4aAAa8. Optimum Room Acoustic Comfort<sup>TM</sup> (RAC<sup>TM</sup>) can be achieved by using a selection of appropriate acoustic descriptors.** Carsten Svensson (Saint-Gobain Ecophon AB, Box 500, SE-260 61 Hyl-linge, Sweden, [carsten.svensson@ecophon.se](mailto:carsten.svensson@ecophon.se)), Erling Nilsson (Saint-Gobain Ecophon AB, Box 500, SE-260 61 Hyl-linge, Sweden, [erling.nilsson@ecophon.se](mailto:erling.nilsson@ecophon.se))

In order to create an optimum Room Acoustic Comfort<sup>TM</sup> (RAC<sup>TM</sup>) in rooms it is important to consider a variety of different acoustic descriptors. These descriptors must match and facilitate for wanted human qualities such as ability to concentrate, reduced stress, clear speech etc. In this process it is important to consider the people, what they do (the activity) and what room they will be in. Today, when designing ordinary rooms from an acoustic perspective, mainly reverberation time (T20) is utilised - both in practice but also in building regulation and standards. Reverberation time (T20) only describe the later part of the decay curve, and therefore only partly mirror the wanted acoustic reality. Thus, based upon a large number of acoustic measurements, we suggest a "cocktail" of acoustic descriptors for ordinary rooms in buildings like schools, offices, health care premises etc. These descriptors have to cover both early and late decay, sound levels and speech quality. Our suggestions are Speech Clarity (C50), Speech Transmission Index (STI), Early Decay Time (EDT), Reverberation Time (T20) and Strength (G). Moreover, in open and long spaces we also suggest the acoustic descriptors Rate of Spatial Decay (DL2) and Excess of Sound Pressure Level (DLf).

11:00

**4aAAa9. Distribution of Speech Intelligibility Metrics in Classrooms with Varied Signal to Noise (S/N) Ratios.** Nurgun Bayazit Tamer (Istanbul Technical University, Taskisla Taksim Istanbul, 34437 Istanbul, Turkey, [nurgun@itu.edu.tr](mailto:nurgun@itu.edu.tr))

In a classroom to provide adequate speech intelligibility is vital especially when young children are concerned. Room acoustical characteristics of classrooms like reverberation times and background noise mainly define the speech intelligibility in classrooms. Excessive background noise deteriorates the signal to noise ratio (S/N) and leads to reduction in learning efficiency. An extensive measurements study is being used in Istanbul Elementary Schools as part of a project. This paper explains the results of the investigation of the influence of varying signal to noise ratios on different speech intelligibility metrics. Background noise includes external noises such as outdoor traffic noise, noise from playground or noise from adjacent classrooms. STI, RASTI, Alcon(%) values of 20 different classrooms in 20 different schools are measured in 9 different positions. The measurements were repeated in each classroom while windows were open and closed respectively. The object of the work was to systematically study the influence of the S/N ratio variations and reverberation time on the different speech intelligibility metrics. Finally to elucidate the effects of different absorption treatments achieving recommended reverberations on the measured speech metrics are also discussed.

11:20

**4aAAa10. Speaker comfort and increase of voice level in lecture rooms.** Jonas Brunsog (Dept. of Acoustic Technology, Technical University of Denmark, Building 352, DK 2800 Lyngby, Denmark, [jbr@oersted.dtu.dk](mailto:jbr@oersted.dtu.dk)), Anders C. Gade (Dept. of Acoustic Technology, Technical University of Denmark, Building 352, DK 2800 Lyngby, Denmark, [acg@oersted.dtu.dk](mailto:acg@oersted.dtu.dk)), Gaspar Payà Bellester (C/ Sènia 1, 1er. C.P., 03640 Monòver Alacant, Spain, [kansbarpb@gmail.com](mailto:kansbarpb@gmail.com)), Lilian Reig Calbo (Alacant, Spain, [xusketa@hotmail.com](mailto:xusketa@hotmail.com))

Teachers often suffer health problems or tension related to their voice. These problems may be related to there working environment, including room acoustics of the lecture rooms which forces them to stress their voices. The present paper describes a first effort in finding relationships between the objectively measurable parameters of the rooms and the objective voice power produced by speakers. In rooms with different sizes, reverberation time and other physical attributes, the sound power levels produced by six speakers where measured while giving a short lecture. Relevant room acoustic parameters were also measured in the rooms and subjective impressions from about 20 persons who had experience talking in these rooms were collected as well. Analysis of the data revealed significant differences in the sound power produced by the speaker in the different rooms. It was also found that these changes were mainly related to the size of the room and to the gain or support produced by the room. To describe this quality, a new room acoustic quantity called 'room gain' is proposed.

11:40

**4aAa11. Measuring the mixing time in auditoria.** Guillaume Defrance (Institut Jean le Rond d'Alembert, Laboratoire d'Acoustique Musicale, 11, rue de Lourmel, 75015 Paris, France, defrance@lam.jussieu.fr), Jean-Dominique Polack (Institut Jean le Rond d'Alembert, Laboratoire d'Acoustique Musicale, 11, rue de Lourmel, 75015 Paris, France, polack@ccr.jussieu.fr)

The goal of prediction, in Room Acoustics, is to synthesize the impulse responses (IRs) of a hall, in order to derive acoustic indices or to allow auralization. The process assumes the hall to be a time invariant linear

system. Furthermore, the IR is known to behave stochastically when the sound field becomes diffuse, that is, after a certain time called mixing time. This study aims at characterizing the IR mixing time. Three methods are presented for visualizing and detecting the time evolution of the IR behaviour. The first one highlights the transition from early reflections to diffuse sound field by monitoring the phase evolution versus time. The two others exploit the gaussian distribution of pressure in a diffuse sound field, when the IR becomes statistical. These methods are applied to measurements, carried out in Salle Pleyel, and confirm the simple relationship found earlier between mixing time and volume.

THURSDAY MORNING, 3 JULY 2008

ROOM 243, 8:00 TO 9:20 A.M.

### Session 4aAAb

## Architectural Acoustics: Prediction Methods in Building Acoustics II

Berndt Zeitler, Cochair

*NRC - Institute for Research in Construction, 1200 Montreal Road, Building M-27, Ottawa, ON K1A 0R6, Canada*

Catherine Guigou-Carter, Cochair

*CSTB, 24, rue Joseph Fourier, Saint Martin D'Hères, 38400, France*

### Invited Papers

8:00

**4aAAb1. Prediction method for the acoustic performance of permanent form systems.** Jean-Baptiste Chene (CSTB, 84, Avenue Jean-Jaurès, 77447 Marne-la-Vallée Cedex 2, France, jean-baptiste.chene@cstb.fr), Catherine Guigou-Carter (CSTB, 24, rue Joseph Fourier, 38400 Saint Martin D'Hères, France, catherine.guigou@cstb.fr)

Permanent form systems considered in this paper combine a layer of porous or fibrous material onto which a concrete layer is poured. Such systems allow the fulfillment of French thermal regulations. However, their acoustic performance is quite limited. Indeed, the acoustic performance of the concrete layer is usually reduced by the presence of the porous or fibrous layer. The modeling of such multi-layered structures submitted to acoustic excitation is discussed in this work. The behavior of such a system is investigated by using a wave approach based prediction tool. The porous layer is modeled following Biot's theory. The effect of the metallic anchors that connect the fiber and the concrete layers is also investigated. The acoustic performance of such systems is studied both experimentally and analytically. A parametric study is performed to identify the most determinant parameters. The model is thus used to obtain insight into the behavior of such systems in order to develop solutions that result in improved acoustic performance.

8:20

**4aAAb2. The directivity of the forced radiation of sound from panels and openings including the shadow zone.** John L. Davy (RMIT University, Applied Physics, GPO Box 2476V, 3001 Melbourne, Victoria, Australia, john.davy@rmit.edu.au)

This paper presents a method for calculating the directivity of the radiation of sound from a two dimensional panel or opening, whose vibration is forced by the incidence of sound from the other side. The directivity of the radiation depends on the angular distribution of the incident sound energy. For panels or openings in the wall of a room, the angular distribution of the incident sound energy is predicted using a physical model which depends on the sound absorption coefficient of the room surfaces. For an opening at the end of a duct, the sound absorption coefficient model is used in conjunction with a model for the directivity of the sound source in the duct. For angles of radiation approaching 90 degrees to the normal to the panel or opening, the effect of the diffraction by the panel or opening, or by the finite baffle in which the panel or opening is mounted, is included. A simple empirical model has been developed to predict the diffraction of sound into the shadow zone when the angle of radiation is greater than 90 degrees to the normal to the panel or opening. The method is compared with published experimental results.

4a THU. AM

## Contributed Papers

8:40

**4aAAb3. Numerical study of sound transmission loss using an indirect boundary element method.** Matthew Cassidy (Queen's University Belfast, School of Mechanical and Aerospace Engineering, Ashby Building, Stranmillis Road, BT9 5AH Belfast, UK, mcassidy06@qub.ac.uk), Richard K. Cooper (Queen's University Belfast, School of Mechanical and Aerospace Engineering, Ashby Building, Stranmillis Road, BT9 5AH Belfast, UK, R.Cooper@qub.ac.uk), Richard Gault (Queen's University Belfast, School of Mechanical and Aerospace Engineering, Ashby Building, Stranmillis Road, BT9 5AH Belfast, UK, r.gault@qub.ac.uk), Jian Wang (Queen's University Belfast, School of Mechanical and Aerospace Engineering, Ashby Building, Stranmillis Road, BT9 5AH Belfast, UK, j.wang@qub.ac.uk)

The purpose of this study was to simulate transmission loss tests at the acoustic facilities of FG Wilson, Larne, UK. A hemi-anechoic chamber

adjoins a reverberation room via a transmission plug where canopy panel sections are mounted for testing. Boundary element methods in LMS Virtual .Lab are used in conjunction with a baffle model to simulate the test facilities for transmission loss. On one side of this wall the reverberation room is modelled as a diffuse field using a series of defined plane waves, and on the other the hemi-anechoic chamber is represented as a free field. Experiments were carried out on a steel plate and lead sheet following the ISO 15186 standard for measurement of sound insulation using sound intensity. Source room sound pressure levels were recorded with a microphone, and an intensity probe was used to map the sound intensity field on the receiving side. Transmission loss for a frequency range was calculated as stated in the standard and compared with the results for the computational analysis. Comparison of the computational simulation with the experimental yielded a sufficient agreement.

## Invited Paper

9:00

**4aAAb4. Prediction of the Sound Transmission Loss of Multi-layered Small Sized Elements.** Stefan Schoenwald (Eindhoven University of Technology, Den Dolech 2, BWK/BPS, 5600 MB Eindhoven, Netherlands, s.schoenwald@tue.nl), Eddy Gerretsen (TNO Science and Industry, Stieljesweg 1, 2628CK Delft, Netherlands, eddy.gerretsen@tno.nl), Heiko J. Martin (Eindhoven University of Technology, Den Dolech 2, BWK/BPS, 5600 MB Eindhoven, Netherlands, h.j.martin@tue.nl)

In this paper an improved method for the prediction of the sound transmission loss of multilayered finite structures, like glazing will be presented. The sound transmission loss of an infinite structure is predicted with a common transfer matrix as a function of the angle of the incident sound wave. Then Villiot's spatial windowing method is applied to take into account the finiteness of the element. Usually an ideal diffuse distribution of the incident sound power is assumed and the prediction results are integrated over all angles of incidence. The obtained prediction results tend to underestimate sound transmission loss due to the dominance of the small values for grazing incidence. Often simple ad-hoc corrections are used for improvement, like Beranek's field incidence, that fail for multilayered structures. Kang suggests that the incident sound power on a surface of a room generally is Gaussian distributed on the angle of incidence and introduces a weighting function for the integration of the prediction results over the angles of incidence. New in this paper is that spatial windowing as well as a Gaussian distributed sound power is considered for the prediction of the transmission loss. The results of the prediction are validated by experiment.

THURSDAY MORNING, 3 JULY 2008

ROOM 243, 10:00 A.M. TO 7:00 P.M.

## Session 4aAAc

### Architectural Acoustics: Measuring Methods and Uncertainty in Building Acoustics I

Brandon Tinianow, Cochair

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Werner Scholl, Cochair

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## Contributed Papers

10:00

**4aAAc1. A new technique for the measurement of the normal incidence absorption coefficient using an impedance tube and a single microphone with fixed position.** Cedric Vuye (Hogeschool Antwerpen, Dept. of Industrial Sciences, Paardenmarkt 92, BE-2000 Antwerpen, Belgium, c.vuye@ha.be), Steve Vanlanduit (Vrije Universiteit Brussel, Acoustics and Vibration Research Group, Dept. of Mechanical Engineering, Pleinlaan 2, BE-1050 Brussels, Belgium, Steve.Vanlanduit@vub.ac.be), Karl Van Nieuwenhuyse (Vrije Universiteit Brussel, Acoustics and Vibration Research Group, Dept. of Mechanical Engineering, Pleinlaan 2, BE-1050 Brussels, Belgium, karl.van.nieuwenhuyse@skynet.be), Patrick Guillaume (Vrije Universiteit Brussel, Acoustics and Vibration Research

Group, Dept. of Mechanical Engineering, Pleinlaan 2, BE-1050 Brussels, Belgium, Patrick.Guillaume@vub.ac.be)

The normal incidence absorption coefficient of acoustic materials can be measured inside an impedance tube with different settings for the microphone(s). The two most widespread techniques are the standing wave method using a probe microphone and side-mounted two-microphone techniques. Errors that can occur here are related to phase mismatch between the two microphones and the knowledge of the exact locations of the (acoustic centre of the) microphones and test sample. These problems have (partially) been solved by, for example, calibrating the microphones by swapping them and calculating a calibration transfer function or by using one microphone techniques. In this article we will present a novel technique



which will also avoid the need for the knowledge of the exact microphone location by continuously moving the sample under test. Different results will be presented and compared to the traditional techniques.

10:20

**4aAAc2. New method for measuring sound absorption coefficients in an industrial hall.** Joël Ducourneau (Faculté de Pharmacie de Nancy, Université Henri Poincaré, 5, rue Albert Lebrun, BP 80403, 54001 Nancy, France, joel.ducourneau@pharma.uhp-nancy.fr), Vincent Planeau (Institut National de Recherche et de Sécurité (INRS), Ave. de Bourgogne, B.P. 27, F-54501 Vandoeuvre Cedex, France, vincent.planeau@secav.fr), Jacques Chatillon (Institut National de Recherche et de Sécurité (INRS), Ave. de Bourgogne, B.P. 27, F-54501 Vandoeuvre Cedex, France, jacques.chatillon@inrs.fr), Armand Nejade (Institut National de Recherche et de Sécurité (INRS), Ave. de Bourgogne, B.P. 27, F-54501 Vandoeuvre Cedex, France, armand.nejade@inrs.fr)

### Invited Paper

10:40

**4aAAc3. Measurement uncertainty of the sound absorption coefficient.** Anna Izewska (Building Research Institute, Filtrowa Str.1, 00-611 Warsaw, Poland, a.izewska@itb.pl)

The standard ISO/IEC 17025:2005 on the competence of testing and calibration laboratories requires that these laboratories shall apply procedures for estimating the uncertainty of their measurement results. One of the possibility is to evaluate the budget of uncertainty, taking into account all components that contribute significant uncertainty to the final result. In case of the sound absorption coefficient measurement, carried out according to the standard EN ISO 354:2003, the overall uncertainty is first of all influenced by the reverberation times  $T_1$ ,  $T_2$  and the power attenuation coefficients  $m_1$  and  $m_2$ , calculated according to the ISO 9613-1 standard and representing the climatic conditions in the reverberation room. In spite of very little difference between the values  $m_1$  and  $m_2$  representing the change of climatic conditions (usually, it is the case in laboratory), exponential form of the coefficient's function causes that the uncertainty of measurement results increase with frequency very fast. Particularly for the high frequencies, the values of uncertainty are so important that the evaluation of the sound absorption coefficient is practically not possible.

### Contributed Papers

11:00

**4aAAc4. A comparison of impulse-like sources to be used in reverberation time measurements.** Marko Horvat (Faculty of EE and Computing, Unska 3, Department of Electroacoustics, HR-10000 Zagreb, Croatia, marko.horvat@fer.hr), Kristian Jambrosic (Faculty of EE and Computing, Unska 3, Department of Electroacoustics, HR-10000 Zagreb, Croatia, kristian.jambrosic@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 an extensive ongoing research on reverberation time measurements, efforts have been made to find an impulse-like source that will comply with certain demands concerning its spectral content and sound pressure levels it can provide. Although the relevant standards state that such kind of sources should be avoided due to the lack of measurement repeatability, their use has proved to be necessary due to inability of conventional omni-directional loudspeakers to provide adequate sound pressure levels, especially at the low end of the frequency range of interest, thereby ensuring sufficient dynamic range. Therefore, the emphasis of this investigation has been made on peak sound pressure levels and the amount of low frequency content each source is able to provide. The investigated sources include 6 mm and 8 mm pistols, firecrackers with different amount of explosive compound and explosive mixtures of acetylene gas.

11:20

**4aAAc5. An Application of Time-Reversal Acoustics to Sound Insulation Measurements in Buildings.** Doheon Lee (University of Sydney, Faculty of Architecture, Design and Planning, NSW 2006 Sydney, Australia, dlee7117@mail.usyd.edu.au), Densil Cabrera (University of Sydney, Faculty of Architecture, Design and Planning, NSW 2006 Sydney, Australia, densil@usyd.edu.au)

This paper considers the possible application of time reversal acoustics (TRA) to airborne sound insulation measurements in buildings. In TRA, an

Predicting the sound pressure level at a workplace requires in-situ characterization of the facings. This work describes a new method for the measurement of the sound absorption coefficient of flat panels present in industrial halls. In such room, it is necessary to separate the echo coming from the studied panel from the others due to the entire reverberation. This separation has been achieved in space by an acoustic array and in time by an impulse sound source. The array processing uses a multipolar weighting to achieve a directivity constant with frequency and with attenuated side lobes. This weighting requires a limited number of microphones. The impulse source has been designed using the inverse impulse response from the emission system (equalizer, amplifier, loudspeaker). This inverse filtering technique allows equalizing the response of the emission system in order to radiate very short pulses. Sound absorption coefficient of several flat facings have been evaluated by mean of this new device in a semi-anechoic chamber and in an industrial hall designed for testing. The results show a good agreement with others techniques except at low frequencies for which the array length is too small and the absorption coefficients too low.

array of transducers is set up to form a time-reversal 'mirror' or 'cavity'. Using this array, the sound radiated from an initial source is collected and refocused spatially and temporally, thereby being reproduced at the initial source position with high signal-to-noise ratio (S/N). Most previous studies of TRA have been conducted underwater with ultrasonic sound sources, with only a few in the audible range in real buildings. The technique is best suited to non-dissipative systems, raising the question of whether any advantage could exist for transmission between rooms. This study applies TRA experimentally in the audible range using maximum length sequence signals for transmission between two rooms. Comparison is made between conventional measurements (with and without impulse response deconvolution) and TRA in terms of effective S/N and apparent level difference between the rooms. Substantially greater S/N is achieved using TRA, but the interpretation of measurements is not straightforward, and the technique is much more demanding than conventional measurements.

11:40

**4aAAc6. Measuring Sound Insulation using Deconvolution Techniques.** Constant C. Hak (Technische Universiteit Eindhoven, De Rondom 10, 5612 AP Eindhoven, Netherlands, c.hak@inter.nl.net), Nicole H. Van Hout (LeVel Acoustics BV, De Rondom 10, 5612 AP Eindhoven, Netherlands, n.h.a.m.v.hout@tue.nl), Heiko J. Martin (Eindhoven University of Technology, Den Dolech 2, BWK/BPS, 5600 MB Eindhoven, Netherlands, h.j.martin@tue.nl)

To compare the acoustic performance of a building element with the given sound insulation requirements, measurements need to be done. Generally, a broadband noise source is used according to international standards. This method does not always work in practice due to high sound insulation values or high background noise levels. It is very inconvenient from a practical point of view or even impossible to perform an accurate sound insulation measurement for all frequency bands. A solution to this problem can be found in deconvolution techniques using mls or sweep signals. It is possible to increase the signal to noise ratio with these techniques by averaging mea-

measurements and spreading out the spectral sound energy in time. As a result an efficient use of available sound power is possible. In a laboratory is investigated how to use mls or sweep signals as a source signal and deconvolution as a measurement technique to obtain the sound insulation under noisy conditions.

12:00

**4aAAc7. Characterising a washing machine as a structure borne sound source on a lightweight floor.** Matthias Lievens (Institute of Technical Acoustics, Neustraße 50, 52056 Aachen, Germany, mli@akustik.rwth-aachen.de)

The transfer of structure borne sound power depends on the mobility of the source and the receiver. If source and receiver are coupled through multiple points, the interaction between those points has to be accounted for. The force of a washing machine injected into a lightweight wooden floor is analysed to develop a simple measurement procedure for similar multiple point structure borne sound sources. A complete mobility model will be compared with a simplified model based on a reduced mobility matrix. The importance of different matrix components will be determined. Receiver structures used in real buildings will be investigated.

### *Invited Paper*

12:20

**4aAAc8. On the use of scaled models in building acoustics.** Volker Wittstock (Physikalisch-Technische Bundesanstalt, Bundesallee 100, 38116 Braunschweig, Germany, volker.wittstock@ptb.de), Martin Schmelzer (Physikalisch-Technische Bundesanstalt, PTB, Working group 1.71 'Building Acoustics', Bundesallee 100, 38118 Braunschweig, Germany, martin.schmelzer@ptb.de), Christoph Kling (Physikalisch-Technische Bundesanstalt, PTB, Working group 1.71 'Building Acoustics', Bundesallee 100, 38118 Braunschweig, Germany, christoph.kling@ptb.de)

Experimental studies of physical effects in building acoustics are usually time consuming and expensive. This is mainly caused by the building costs but also by the experimental effort. It is thus desirable to have another method for the investigation of basic effects in building acoustics. Building acoustic problems are characterized by the interaction between airborne and structure-borne sound fields. It is therefore possible to use scaled models when both sound fields are treated correctly. This means that the wavelengths in the airborne and in the structure-borne sound fields have to be scaled in the same way. With a scaling factor of typically 1:8, the costs can be reduced drastically and nearly all model parameters can be changed separately. Due to these advantages, this technique is used at PTB's building acoustics group. This contribution gives an overview on the physical background of scaled models, reports on validation experiments and on several applications e.g. investigations of the influence of temperature and static pressure, damping effects, geometry influence on the sound insulation of walls, the measurement of the flanking transmission of walls and the measurement of suspended ceilings.

12:40-1:40 Lunch Break

### *Contributed Paper*

1:40

**4aAAc9. Influence of the source orientation on the measurement of acoustic parameters in a large reverberant cathedral.** Miguel Arana (Public University of Navarre, Physics Department. Campus de Arrosadia, s/n, 31006 Pamplona, Spain, marana@unavarra.es), Ricardo San Martin (Public University of Navarre, Physics Department. Campus de Arrosadia, s/n, 31006 Pamplona, Spain, ricardo.sanmartin@unavarra.es), Maria Luisa San Martin (Public University of Navarre, Physics Department. Campus de Arrosadia, s/n, 31006 Pamplona, Spain, sanmartin@unavarra.es)

ISO 3382 standard describes both definitions and measurement procedures of different acoustic parameters derived from the room impulse

response. Regarding to sound sources, most of the commercial dodecahedron loudspeakers comply with the maximum allowed directional deviations of the source specified in the standard. However, the influence of its specific orientation may affect the results obtained on some parameters more than their subjective just noticeable difference-jnd- at least in rooms with no high reverberation times. An interesting aim is to study such influence in function of the liveliness of the room. A detailed measurement set is been carrying out in a reverberant place (Cathedral of Tudela, Spain) with the objective to analyze the influence of the source's orientation-apart from its acoustic characterization. In addition to dodecahedron loudspeakers, pseudo-impulsive sources are been used in order to compare results from a statistical point of view. Results obtained will be compared with those obtained in several concert and theater rooms.

### *Invited Papers*

2:00

**4aAAc10. Measurement of reverberation time with rotating microphone in test chamber and its problems.** Hiroshi Sato (National Institute of Advanced Industrial Science and Technology, Central 6, 1-1-1 Higashi, 305-8566 Tsukuba, Japan, sato.hiro@aist.go.jp), Junichi Yoshimura (Kobayasi Institute of Physical Research, 3-20-41 Higashi-Motomachi, Kokubunji, 185-0022 Tokyo, Japan, yoshi@kobayasi-riken.or.jp), Satoshi Sugie (Kobayasi Institute of Physical Research, 3-20-41 Higashi-Motomachi, Kokubunji, 185-0022 Tokyo, Japan, sugie@kobayasi-riken.or.jp), Takashi Koga (Kajima Technical Research Institute, 2-19-1 Tobitakyu, Chofu, 182-0036 Tokyo, Japan, tkoga@kajima.com), Emi Toyoda (Kobayasi Institute of Physical Research, 3-20-41 Higashi-Motomachi, Kokubunji, 185-0022 Tokyo, Japan, toyoda@kobayasi-riken.or.jp), Jongkwan Ryu (National Institute of Advanced Industrial Science and Technology, Central 6, 1-1-1 Higashi, 305-8566 Tsukuba, Japan, jongkwan-ryu@aist.go.jp)

When measurement of reverberation time is done in a test chamber to evaluate acoustical property of materials, spatial averaging of reverberation time should be done. Using microphone rotator is recognized as one of the tool to do spatial averaging. This study compares between three methods of spatial averaging of reverberation times measured in small rectangular test chamber (3m x 4m x 5m). The first method is averaging reverberation time measured at 5 of fixed position used as standard positions for testing, the second is at 72 fixed positions on the circle of microphone rotator, and the third is with microphone rotator (64 s/rotation). The result of comparison between three method reveals that reverberation time measured by rotating microphone has more scatter than and presents different reverberation time from those measured by other method especially at lower frequency bands. Simulation of microphone rotation with

the decay curves measured at 72 fixed positions suggests that spatial distribution of steady state sound pressure level, rotation speed of microphone, and reverberation time of test chamber are key factors of errors. As a conclusion, the strict guideline for measurement of reverberation time with microphone rotator should be presented to minimize errors.

2:20

**4aAAc11. Uncertainty in building acoustics.** Werner Scholl (Physikalisch-Technische Bundesanstalt, Bundesallee 100, 38116 Braunschweig, Germany, werner.scholl@ptb.de), Volker Wittstock (Physikalisch-Technische Bundesanstalt, Bundesallee 100, 38116 Braunschweig, Germany, volker.wittstock@ptb.de)

In many countries, legal requirements exist with respect to the acoustical performance of buildings and building elements. Therefore information about the uncertainty of measured or predicted building acoustic properties is urgently needed. The complexity of the problem becomes obvious taking sound reduction index R as an example: R represents the ratio of incoming and transmitted sound power of a building element. For practical reasons, the direct measurement of R is replaced by a spatiotemporally averaged sound pressure level difference in two limited rooms, adjusted by the absorption of the receiving room. In doing so, unwanted influences occur like modal effects, flanking transmission, structural power exchange between laboratory and specimen, deviation from ideal diffuse sound fields with unknown consequences etc. For economic reasons, often only one sample is tested and declared 'typical' for the whole family of products without regarding their spread. As a consequence, the uncertainty most often is felt to be too large to decide about the compliance with regulations but cannot be quantified. PTB in Germany has investigated the problem for the German authorities by calculation, evaluation of Round-Robin-tests und large measurement series in model houses. The results are presented in the talk.

### Contributed Paper

2:40

**4aAAc12. Uncertainty evaluation in field measurements of airborne sound insulation.** Ranny L.X. Michalski (Inmetro / CNPq, Av. N. S. das Graças, 50, Xerém, Duque de Caxias, 25250-020 RJ, Brazil, ranny\_xavier@yahoo.com.br), Daiana Ferreira (Inmetro / CNPq, Av. N. S. das Graças, 50, Xerém, Duque de Caxias, 25250-020 RJ, Brazil, dpferreira@inmetro.gov.br), Marco Nabuco (Inmetro / CNPq, Av. N. S. das Graças, 50, Xerém, Duque de Caxias, 25250-020 RJ, Brazil, nabuco@inmetro.gov.br), Paulo Massarani (Inmetro / CNPq, Av. N. S. das Graças, 50, Xerém, Duque de Caxias, 25250-020 RJ, Brazil, pmmassarani@inmetro.gov.br)

The Brazilian Committee of Civil Engineering presented a set of standards concerning the evaluation of the performance of several topics for buildings up to five floors. The acoustic performance is one of them. The standards are in approval process and measurements in real buildings will be

necessary. Different professionals using different equipment will emit certificates establishing which levels of insulation a certain flat provides and its uncertainties. The expanded measurement uncertainty can provide the basis to compare different measurement results for a same building. The international Guide to the Expression of Uncertainty in Measurement, the ISO GUM, is the document that specifies how to determine and evaluate the uncertainty of a measurement result. The standards concerning sound insulation measurements are ISO 140 and ISO 18233. Uncertainty estimates are available only for the classical technique described in ISO 140, based in repeatability and reproducibility tests performed in laboratories. Field measurements present some characteristics that can contaminate the results, as time variance. Several independent measurements were carried out in a one flat building using ISO 18233 specifications and the ISO GUM was applied to obtain the uncertainty for measurement results of airborne sound insulation between rooms in situ.

### Invited Paper

3:00

**4aAAc13. Uncertainty of Receiving Space Volume in Field Measurements of Transmission Loss Under ASTM E336-05.** Jonah Sacks (Acentech, 33 Moulton Street, Cambridge, MA 02138, USA, jsacks@acentech.com)

As acoustical consultants, we are frequently asked by our clients to measure the sound isolating performance of constructions in the field. While it is often preferable to report "system-level" performance ratings such as Noise Isolation Class, there are compelling benefits both to us and to our clients to measuring and reporting "specimen-level" performance ratings such as Apparent Sound Transmission Class, defined by ASTM E336-05 (and E413-04). The accuracy of such ratings depends on accurate assessment of the amount of acoustical absorption present in receiving spaces at the time of testing, arrived at by means of reverberation time measurements and physical measurement of receiving spaces. When a receiving space is irregular in shape, one may feel pressed to use creative judgment to estimate its effective volume, and such judgments can have large impacts on the reported results. We will discuss the challenges of measuring apparent transmission loss in the field, and the compelling reasons to conduct these measurements despite the challenges.

### Contributed Paper

3:20

**4aAAc14. Field measurements of acoustic performance in buildings: a Round Robin Test.** Fabio Scamoni (Construction Technologies Institute of Italian National Research Council, Viale Lombardia, 49, 20098 San Giuliano Milanese (MI), Italy, Fabio.Scamoni@itc.cnr.it), Maurizio Bassanino (ARPA Lombardy - Air and Physical Agents, 3/1, Viale F. Restelli, 20124 Milan, Italy, M.BASSANINO@arpalombardia.it), Giuseppe Bruno (Lombardy Region - Environmental Quality, Via Pola, 14, 20124 Milan, Italy, giuseppe\_bruno@regione.lombardia.it), Giovanni Zambon (Department of Environmental Sciences of the University of

Milano - Bicocca, Piazza della Scienza, 1, 20126 Milan, Italy, giovanni.zambon@unimib.it)

This paper presents the experimental results of a round robin test performed on the same building by different teams working with three independent bodies: a research body, ITC-CNR, a university laboratory DISAT and the Regional Agency for Environment Protection of Lombardy, ARPA. A partition wall (the airborne sound insulation between rooms), a floor (the impact sound insulation between rooms) and a façade (the insulation of the façade against outdoor sound) were tested, using the measurement methods



given in the relevant parts of the ISO 140 series. The accuracy of acoustic measurement methods in buildings depends on many factors. In this round robin test, the situation (varying from straightforward rectangular rooms to half open spaces with all kind of shapes) and the construction details were kept fix. The analysis was aimed at investigating the influence of the oper-

ating condition of the equipment (repeatable settings) and the reproducibility, using different types of acoustic instrumentation and varying the method (choice of source and microphone positions). In particular the tolerance limits in the verification of the requirements of tender specifications were investigated.

### 3:40-5:00 Posters

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

### *Invited Papers*

5:00

**4aAAc15. Handling of uncertainties for CE marking concerning Sound Transmission Loss of glazings.** Marc Rehfeld (Saint Gobain Glass, CRDC, B.P. 40103, 60777 Thourrotte Cedex, France, marc.rehfeld@saint-gobain.com)

A joint working group between CEN TC 126 (building acoustics) and CEN TC 129 (glass in building) has been created to handle the redaction of a test code, (give rules for CE marking) and particularly handle uncertainty problems. A round robin has been organised, at which 23 European labs have participated, and two configuration of double glazings have been tested: one with two monolithic glass components, the other with one monolithic and one highly damped laminate component. The paper will present the specifications and main results of this round robin, the questions that occurred concerning the possible ways to decrease uncertainty values, as well as the conclusions and decisions of the working group.

5:20

**4aAAc16. Repeatability and reproducibility of field noise isolation testing.** John Loverde (Veneklasen Associates, 1711 Sixteenth Street, Santa Monica, CA 90404, USA, jloverde@veneklasen.com), Wayland Dong (Veneklasen Associates, 1711 Sixteenth Street, Santa Monica, CA 90404, USA, wdong@veneklasen.com)

The reliability and precision of test measurements and methods are generally described in terms of repeatability and reproducibility. ASTM standards define and quantify these terms for noise isolation test methods in both laboratory and field conditions. Understanding these as reproducibility and repeatability of the measurement method, the authors extend the concept to the reproducibility and repeatability of a wall or floor/ceiling assembly design. Multiple instances of a floor/ceiling assembly on a multi-family residential project built by the same contractors is an example of design repeatability, while the same assembly design constructed on different projects is an example of design reproducibility. In a previous paper [LoVerde and Dong, J. Acoust. Soc. Am. 122, 2955 (2007)], definitions were suggested for field repeatability and reproducibility for Field Impact Insulation Class testing. Test data is presented to quantify the field repeatability and reproducibility of several assembly designs, which are compared to laboratory values. Field and laboratory repeatability and reproducibility of airborne noise isolation for a partition assembly are also examined.

5:40

**4aAAc17. Analysis of uncertainty in building acoustic predictions using Monte-Carlo methods.** Ralph T. Muehleisen (Illinois Institute of Technology, Civil, Architectural and Environmental Engineering, 3201 S. Dearborn St., Room 228, Chicago, IL 60616, USA, muehleisen@iit.edu)

One of the most common tasks in architectural acoustics is the prediction of the acoustic performance of some aspect of a building, such as the background sound level of a room, the reverberation time of an enclosure, or the sound transmission of a wall construction. The accuracy of a prediction is not only dependent upon the computation model, but also upon the accuracy of the data of the model. Because of the complicated, non-linear interaction of various inputs, assessing the accuracy of a prediction can be difficult. One way to provide more accurate predictors and estimate the error in the prediction of complicated, multiple input systems is to utilize the Monte-Carlo method. In this talk, the application of Monte-Carlo methods to building acoustic predictions is presented.

### *Contributed Papers*

6:00

**4aAAc18. Diffuseness and sound field distribution at room boundaries.** Jean-Daniel Chazot (Université de Technologie de Compiègne, Lab. Roberval UMR 6253, Dept. Acoustique, BP 60319, 60203 Compiègne, France, jean-daniel.chazot@utc.fr), Jean-Louis Guyader (INSA de Lyon - LVA, Bâtiment St. Exupéry, 25 bis avenue Jean Capelle, F-69621 Villeurbanne Cedex, France, jean-louis.guyader@insa-lyon.fr)

In the present work, a classical modal analysis is used up to medium frequencies to study the sound field distribution, and its diffuseness, particularly at boundaries. Due to intensification zones at boundaries, the diffuse field distribution at room boundaries can not be assimilated to the

distribution inside the room. Moreover, diffuseness at room boundaries, that is of interest for sound insulation measurement, is usually only related to an incidence angle while inside the room volume several descriptors such as a correlation function and the spatial uniformity are necessary to characterize a diffuse field. In this paper, we present a new descriptor adapted to characterize the sound field diffusivity at boundaries. This descriptor is called Boundary Diffuse Field Index. Its averaged value over a specific surface can be related to a limit incidence angle, and its standard deviation can be related to the spatial distribution over the surface. Finally, thanks to this descriptor, Sabine's assumptions of diffuse sound field are also evaluated in this study.

**4aAAc19. Previous results on the evaluation of the influence of sound level meter case in diffuse field.** Alfonso R. Molares (University of Vigo, E.T.S.I de Telecomunicación, Rúa Maxwell s/n, 36310 Vigo, Spain, amolares@gts.tsc.uvigo.es), Manuel A. Sobreira-Seoane (University of Vigo, E.T.S.I de Telecomunicación, Rúa Maxwell s/n, 36310 Vigo, Spain, msobre@gts.tsc.uvigo.es)

In order to evaluate the influence of the sound level meter case on real measurements, two different case geometries are tested using Boundary Element Method. Solving the coupled structural-acoustic problem by means of an iterative procedure, their correction curves are obtained for various angles of incidence and for diffuse field. The deviation due to the case is obtained by comparison against the result of the reference microphone. The results reveal that the influence of the case can be relevant at the middle frequency range with deviations that can exceed 0.4 dB under free-field conditions for normal incidence. At the time of writing this abstract the available results show that the deviation for diffuse field at middle frequencies can reach 0.2 dB, although further research is needed to evaluate the deviation at higher frequencies.

**4aAAc20. Uncertainty analysis in acoustical modeling of room.** Dominik Mleczko (Romera 17, 33-300 Nowy Sacz, Poland, dmleczko@agh.edu.pl)

For a long time there is a need in industry of acoustical modeling of rooms. It is necessary for new production room design, machine exchange, renovation or enlargement of production rooms, change in a production profile or acoustical room adaptation for acoustical work conditions improvement. In such cases modeling quality is essential and thanks to uncertainty analysis it is possible to quantitatively estimate the effect that input parameters value variation has on model behavior. The article presents general rules for sound pressure level prediction uncertainty calculation in a room. By partial uncertainty calculation analysis of input parameters influence on uncertainty prediction an effort was taken to find parameters with biggest influence on the prediction process. As an example an industrial production room is presented which was modeled to predict noise level on a work stands after it was expanded.

THURSDAY MORNING, 3 JULY 2008

ROOM 342B, 8:00 TO 10:20 A.M.

### Session 4aABa

## Animal Bioacoustics, Psychological and Physiological Acoustics, and ECUA: Auditory Brainstem Response and Behavior Correlation II

Elizabeth Brittan-Powell, Cochair

*Dept of Psychology, University of Maryland, College Park, MD 20742, USA*

Alexander Y. Supin, Cochair

*Institute of Ecology and Evolution, 33 Leninsky Prospect, Moscow, 119071, Russian Federation*

### Invited Papers

8:00

**4aABa1. Evoked-potential study of hearing directivity and sound-receiving apertures in dolphins.** Vladimir V. Popov (Institute of Ecology and Evolution, 33 Leninsky Prospect, 119071 Moscow, Russian Federation, popov\_vl@sevin.ru), Alexander Y. Supin (Institute of Ecology and Evolution, 33 Leninsky Prospect, 119071 Moscow, Russian Federation, alex\_supin@mail.ru)

Positions of sound-receiving apertures were searched-for in bottlenose dolphins using the ABR technique. The receiving-area position was computed basing on ABR delays at various sound-source positions. Two acoustic apertures were revealed in such a way: for frequencies of 32 kHz and higher, the receiving area was located near the proximal part of the lower jaw (the mandibular acoustic window); for lower frequencies, the receiving area was located near the tympanic bulla. In another experimental series, AEP thresholds to near-field stimuli were measured with transducer positioning next to various points of the dolphin's head. Again, at stimulus frequencies of 32 kHz and higher, the lowest threshold area was next to the mandibular acoustic window; at lower frequencies, the lowest threshold area was next to the bulla. The conclusion is that dolphins have at least two acoustic apertures differing in their frequency sensitivity. Directional sensitivity of these two apertures was investigated by measuring ABR thresholds at different frequencies and different sound source positions. At higher frequencies, the best-sensitivity direction estimated by ABR thresholds was near the head midline, at lower frequencies the best-sensitivity direction deviated laterally. These data were interpreted as indicating different axis directions of the two receiving apertures.

8:20

**4aABa2. Tuning curves derived from auditory brainstem responses point to a defect in outer hair cells of hypothyroid mice.** Edward Walsh (Boys Town National Research Hospital, 555 North 30th Street, Omaha, NE 68132, USA, walsh@boystown.org), Megan Korte (Boys Town National Research Hospital, 555 North 30th Street, Omaha, NE 68132, USA, kortem@boystown.org), Joann McGee (Boys Town National Research Hospital, 555 North 30th Street, Omaha, NE 68132, USA, mcgee@boystown.org)

Based on an analysis of ABR derived tuning curves, recent reports suggest that the mechanics of passive transduction in hypothyroid mice, although delayed developmentally, eventually become indistinguishable from normal animals, whereas the mechanics of active transduction remain grossly abnormal throughout life, raising the possibility that the outer hair cell system is at least partially responsible for abnormalities observed in mutant animals. Moreover, results of in vitro studies have shown that although OHCs are electro-

motile, they appear unable to withstand extensive voltage excursions in the whole cell voltage clamp environment; i.e., they appear fragile. These data led us to question the structural integrity of the OHC lateral wall in hypothyroid mice. Preliminary studies suggest that the cortical cytoskeleton is abnormal in at least some regions of the OHC lateral wall and preliminary confocal immunofluorescence images of the constituent proteins, f-actin and  $\alpha$ -II spectrin, suggest that spectrin is either absent or expressed in very low levels in the cytoskeleton of OHCs harvested from the hypothyroid progeny of Tshr mutant dams. We suggest that a cytoskeletal defect involving the f-actin cross-linking protein, spectrin, might compromise the efficient transfer of force along the long axis of OHCs, effectively diminishing the power of active amplification.

## Contributed Papers

8:40

**4aABa3. Electrophysiological and behavioral measures of temporary threshold shift in a bottlenose dolphin (*Tursiops truncatus*).** James J. Finneran (US Navy Marine Mammal Program, Space and Naval Warfare Systems Center, 53560 Hull St., Code 71510, San Diego, CA 92152, USA, james.finneran@navy.mil), Carolyn E. Schlundt (EDO Professional Services, 3276 Rosecrans St., San Diego, CA 92110, USA, carolyn.melka@edocorp.com), Brian K. Branstetter (US Navy Marine Mammal Program, Space and Naval Warfare Systems Center, 53560 Hull St., Code 71510, San Diego, CA 92152, USA, branstet@hawaii.edu), Randall L. Dear (Science Applications International Corporation, 4065 Hancock St., San Diego, CA 92110, USA, RANDALL.L.DEAR@saic.com)

Auditory evoked potentials are being increasingly applied to more advanced studies of marine mammal hearing, such as frequency selectivity, temporal processing, and temporary threshold shift (TTS). In this study, both behavioral and electrophysiological techniques were used to measure TTS in a bottlenose dolphin exposed to 20-kHz tones. Behavioral hearing thresholds were estimated using a modified staircase procedure and a whistle response. Electrophysiological thresholds were assessed using the multiple auditory steady-state response. Evoked potential stimuli consisted of seven frequency-modulated tones having carrier frequencies from 10-70 kHz and unique modulation rates. Tones were simultaneously presented and the evoked response at each modulation rate independently tracked to test hearing at all seven frequencies simultaneously. The behavioral and evoked response data both showed frequency-dependent patterns of TTS, with the largest shifts at 30 kHz; however, TTS measured using evoked potentials (up to 40-45 dB) was always larger than that observed behaviorally (19-33 dB). This discrepancy may be the result of the evoked response input-output function, which can be represented as the sum of two processes, a low threshold, saturating process and a higher threshold linear process, that react and recover to fatigue at different rates.

9:00

**4aABa4. Directional sensitivity and hearing pathways in the beluga whale, *Delphinapterus leucas*.** Aude Pacini (University of Hawaii, Hawaii Institute of Marine Biology, P.O. Box 1106, Kailua, HI 96734, USA, aude@hawaii.edu), Paul E. Nachtigall (University of Hawaii, Hawaii Institute of Marine Biology, P.O. Box 1106, Kailua, HI 96734, USA, nachtiga@hawaii.edu), T. Aran Mooney (University of Hawaii, Hawaii Institute of Marine Biology, P.O. Box 1106, Kailua, HI 96734, USA, mooneyt@hawaii.edu), Manuel Castellote (L'Oceanografic, C/. Junta de Murs i Valls, s/n, 46013 Valencia, Spain, mcastellote@oceanografic.org), Kristen A. Taylor (University of Hawaii, Hawaii Institute of Marine Biology, P.O. Box 1106, Kailua, HI 96734, USA, kristent@hawaii.edu), José-Antonio Esteban (Research Department, Parques Reunidos Valencia S. A. L'Oceanogràfic, Ciudad de las Artes y las Ciencias, 46013 Valencia, Spain, investigacion@oceanografic.org)

Although much variation exists in jaw morphology among species, odontocetes are believed to receive sound primarily through the pan bone region of the lower jaw. In order to further examine this jaw hearing hypothesis, we tested the head receiving sensitivity and directional hearing of a beluga whale, *Delphinapterus leucas*. Hearing measurements were conducted with a 9-yr-old female beluga using the auditory evoked potential technique. A preliminary audiogram indicated that the subject had very sensitive hearing (45-55dB from 32-80 kHz) and heard up to 128 kHz. The pathway investigation used a piezo-electric transducer to present the click stimuli, whereas the hearing directivity was measured in the far field, also using broadband clicks. Like the bottlenose dolphin, the subject had a region of high sensitivity around the pan bone region (78 dB), however, unlike the

bottlenose dolphin, we found that the region of maximum sensitivity was at the tip of the lower jaw (75dB). The subject also proved to have highly directional hearing. This study supports the shaded receiver hypothesis and also shows that hearing pathway variations appear to exist among odontocete species and are at least partially dependent on head morphology.

9:20

**4aABa5. Evoked potential and behavioral hearing thresholds in nine bottlenose dolphins (*Tursiops truncatus*).** Carolyn E. Schlundt (EDO Professional Services, 3276 Rosecrans St., San Diego, CA 92110, USA, carolyn.melka@edocorp.com), James J. Finneran (US Navy Marine Mammal Program, Space and Naval Warfare Systems Center, 53560 Hull St., Code 71510, San Diego, CA 92152, USA, james.finneran@navy.mil), Brian K. Branstetter (US Navy Marine Mammal Program, Space and Naval Warfare Systems Center, 53560 Hull St., Code 71510, San Diego, CA 92152, USA, branstet@hawaii.edu), Randall L. Dear (Science Applications International Corporation, 4065 Hancock St., San Diego, CA 92110, USA, RANDALL.L.DEAR@saic.com), Dorian S. Houser (Biomimetica, 7951 Shantung Dr., Santee, CA 92071, USA, dhouser@spawar.navy.mil), Erica Hernandez (University of Southern Mississippi, Department of Psychology, 118 College Dr., Box 5025, Hattiesburg, MS 39406, USA, erica.hernandez@yahoo.com)

Auditory evoked potentials are popular for assessing hearing in marine mammals because they do not require the same access to and training of animals as behavioral methods; however, within-subject comparisons of behavioral and electrophysiological thresholds are still necessary to benchmark evoked potential results against the more universally accepted behavioral data. In this study, auditory thresholds were measured in nine dolphins using both behavioral and electrophysiological methods. Subjects included eight males and one female, ages 21-43 yrs. Some had a full-range of hearing and others exhibited high-frequency hearing loss. Tests were conducted in-air, in San Diego Bay, and/or in quiet pools. Hearing test stimuli included frequency-modulated, amplitude-modulated, or pure tones projected in the direct field or via a "jawphone" contact transducer. Comparisons reveal good agreement between the evoked potential and behavioral methods, particularly when the testing environments, stimulus delivery methods, and stimulus waveforms are similar. The results show that evoked potential thresholds obtained in a variety of conditions provide reasonable approximations to underwater sensitivity, especially with respect to the shape of the audiogram and the upper limit of hearing.

9:40

**4aABa6. Interactions of frequency components of multi-component envelope following response in a beluga.** Alexander Y. Supin (Institute of Ecology and Evolution, 33 Leninsky Prospect, 119071 Moscow, Russian Federation, alex\_supin@mail.ru), Vladimir V. Popov (Institute of Ecology and Evolution, 33 Leninsky Prospect, 119071 Moscow, Russian Federation, popov\_vl@sevin.ru)

The envelope-following response (EFR) in odontocetes is composed of overlapping ABRs produced by each cycle of rhythmic sound stimulus at a rate of a few hundred cycles per sec. It has been shown recently (Finneran et al., JASA 2007, 121: 1775) that a complex stimulus consisting of a few carriers modulated by different rates produces a complex EFR composed of components reproducing all the modulation rates. Using this technique, interactions between different components of complex EFR were investigated in a beluga whale *Delphinapterus leucas*. When all carriers of the complex stimulus were equalized by SPL, the interaction depended on both the SL of carriers (their level relative the threshold) and inter-carrier frequency spacing. Addition of components of higher SL (at low-threshold

frequencies) dramatically reduced the amplitudes of other EFR components, especially at short frequency spacing. The leading factor of this reduction was decreasing of the modulation depth of each of the carriers when the overall power of the stimulus increased by additional components. The threshold estimate of each carrier little depended on the number of components; however precision of threshold determination fell down with increasing the number of stimulus components due to the reduced amplitudes of EFR components.

10:00

**4aABa7. Acoustic communication in *Panthera tigris*: A study of tiger vocalization and auditory receptivity revisited.** Edward Walsh (Boys Town National Research Hospital, 555 North 30th Street, Omaha, NE 68132, USA, walsh@boystown.org), Douglas L. Armstrong (Henry Doorly Zoo, 3701 S. 10th, Omaha, NE 68107, USA, douga@omahazoo.com), Julie Napier (Henry Doorly Zoo, 3701 S. 10th, Omaha, NE 68107, USA, julie.napier@omahazoo.com), Lee G. Simmons (Henry Doorly Zoo, 3701 S. 10th, Omaha, NE 68107, USA, lsimmons@omahazoo.com), Megan Korte (Boys Town National Research Hospital, 555 North 30th Street, Omaha, NE 68132, USA, kortem@boystown.org), Joann McGee (Boys Town National Research Hospital, 555 North 30th Street, Omaha, NE 68132, USA, mcgee@boystown.org)

Preliminary findings reported at the 145th meeting of the Society suggested that confrontational tiger roars contain energy in the infrasonic portion of the electromagnetic spectrum. This discovery generally supported the proposition that free ranging individuals may take advantage of this capability to communicate with widely dispersed conspecifics inhabiting large territories in the wild. Preliminary ABR findings indirectly supported this view suggesting that although tigers are most sensitive to acoustic events containing energy in the 0.3 to 0.5 kHz band, they are most likely able to detect acoustic events in the near-infrasonic and infrasonic range based on the assumption that felid audiograms exhibit uniform shapes. In this study, the spectral content of territorial and confrontational roars was analyzed and relevant features of ABR based threshold-frequency curves were considered in relation to the acoustical properties of both roar types. Unlike the confrontational roar, infrasonic energy was not detected in the territorial roar; however, like the confrontational roar, peak acoustic power was detected in a frequency band centered on ~ 0.3 kHz. In addition, ABR recordings acquired in a double walled sound attenuating chamber recently installed at the Henry Doorly Zoo suggest that acoustic sensitivity is significantly underestimated under "field" conditions.

THURSDAY MORNING, 3 JULY 2008

ROOM 342B, 10:40 A.M. TO 1:00 P.M.

## Session 4aABb

### Animal Bioacoustics: Sound Production and Reception in Amphibious Marine Mammals

Jason Mulsow, Cochair

*UCSC Institute of Marine Sciences, Long Marine Lab - University of California, 100 Shaffer Road, Santa Cruz, CA 95060, USA*

Ronald J. Schusterman, Cochair

*UCSC Institute of Marine Sciences, Long Marine Lab - University of California, 100 Shaffer Road, Santa Cruz, CA 95060, USA*

Ian Boyd, Cochair

*Sea Mammal Research Unit, Gatty Marine Laboratory, University of St Andrews, St Andrews, KY16 8LB, UK*

### Invited Papers

10:40

**4aABb1. Vocal Learning in Pinnipeds: A Model System for Human Speech Evolution.** William T. Fitch (Centre for Social Learning and Cognitive Evolution, University of St. Andrews, School of Psychology, KY16 9JP St. Andrews, Fife, UK, wtsf@st-andrews.ac.uk), Ronald J. Schusterman (UCSC Institute of Marine Sciences, Long Marine Lab - University of California, 100 Shaffer Road, Santa Cruz, CA 95060, USA, rjschust@ucsc.edu), Colleen Reichmuth (UCSC Institute of Marine Sciences, Long Marine Lab - University of California, 100 Shaffer Road, Santa Cruz, CA 95060, USA, coll@ucsc.edu), Marija Spasikova (Centre for Social Learning and Cognitive Evolution, University of St. Andrews, School of Psychology, KY16 9JP St. Andrews, Fife, UK, marija.spasikova@gmail.com), Daniel Mietchen (Structural Brain Mapping Group, University of Jena, PF 07737 Jena, Germany, daniel.mietchen@googlemail.com)

Vocal learning is limited to a small subset of vertebrates: including birds (songbirds, parrots, hummingbirds), and mammals (humans, cetaceans, pinnipeds, and probably elephants and bats). Intriguingly, in most of these species, vocal production involves functionally or mechanistically novel systems: the avian syrinx, the nasal bursae in odontocetes, and ultrasonic echolocation in bats. The novel neural circuitry that evolved to control these systems may provide a "preadaptation" for vocal learning. Only two known vocal learners - humans and seals - definitely use the standard vertebrate vocal production system (larynx and vocal tract). Our studies of vocal production in harbour seals verify a surprisingly human-like vocal production system, and the critical question remaining is what neural mechanisms underlie the control of the *Phoca* vocal system, and whether such "human" specializations as direct cortico-ambiguall connections are also found in seals. Pinniped investigations also open the door to molecular exploration of the genetic bases for neural innovations. In addition to these mechanistic questions, we discuss evolutionary questions. What functional forces might select for vocal learning in aquatic mammals? One hypothesis is that selection for control of breathing may provide a preadaptation for enhanced cortical vocal control, explaining its prevalence in marine mammals.



11:00

**4aABb2. Sound production by pinnipeds can be modified by contingency learning.** Ronald J. Schusterman (UCSC Institute of Marine Sciences, Long Marine Lab - University of California, 100 Shaffer Road, Santa Cruz, CA 95060, USA, rjschust@ucsc.edu), Colleen Reichmuth (UCSC Institute of Marine Sciences, Long Marine Lab - University of California, 100 Shaffer Road, Santa Cruz, CA 95060, USA, coll@ucsc.edu)

In contrast to terrestrial mammals, pinnipeds (seals, sea lions and walruses) have remarkable flexibility in the ways that they can learn to use and modify their amphibious sound emissions. The experiments that we will describe are drawn from captive studies which show that changes in sound production can occur as a result of contingency learning, using food as positive reinforcement. A range of specialized physiological and anatomical adaptations appear to play a critical role in controlling sound production in pinnipeds. These adaptations include breath-holding and buoyancy mechanisms, as well as fine muscular control of the mouth, lips and tongue that may be used primarily in feeding. The manipulation and modulation of air flow through these components of the vocal tract and associated super-laryngeal filters appears to be susceptible to some of the same reinforcing consequences that are routinely used to establish reliable control over motor behaviors, such as flipper waving, in operant conditioning contexts.

11:20

**4aABb3. Inertial and cochlear constraints for high-frequency hearing in phocid and otariid pinnipeds.** Sirpa Nummela (University of Helsinki, Department of Biological and Environmental Sciences, PO Box 65 (Viikinkaari 1), FI-00014 Helsinki, Finland, snummela@fastmail.fm), Simo Hemilä (University of Helsinki, Department of Biological and Environmental Sciences, PO Box 65 (Viikinkaari 1), FI-00014 Helsinki, Finland, simo.hemila@welho.com), Annalisa Berta (San Diego State University, Biology Department, LS 250, San Diego, CA 92182-4614, USA, aberta@sciences.sdsu.edu), Tom Reuter (University of Helsinki, Department of Biological and Environmental Sciences, PO Box 65 (Viikinkaari 1), FI-00014 Helsinki, Finland, tom.reuter@helsinki.fi)

In air-borne hearing, mammals rely on sound transmission through the tympanic membrane and middle ear ossicles between the surrounding air and the cochlea. The high-frequency hearing limit (HFHL) is determined by the ossicular inertia, and also by the cochlear sensitivity. Due to coevolution, the sensitivity ranges of the middle and inner ear structures generally overlap, and the roles of inertial and cochlear constraints for the HFHL are difficult to discern. For studying this question we considered anatomical and experimental data for two phocid and two otariid pinnipeds. While any detailed mechanism for pinniped underwater hearing remains unclear, an underwater HFHL exceeding that in air is possible. Published in-air and underwater audiograms provide an opportunity for comparing the roles of ossicular mass inertia and cochlear sensitivity in HFHL. Phocid ossicles are very heavy, and their inertia explains the lower HFHLs in air - according to underwater audiograms the phocid cochlea is sensitive to higher frequencies. Otariids have normalized mammalian ossicles, and their inertia should allow underwater hearing at higher frequencies than in air. However, the HFHL is approximately equal in air and water for otariids, hence their underwater HFHL is apparently set by the cochlea alone.

11:40

**4aABb4. Evoked potential audiometry in sea lions.** Jason Mulsow (UCSC Institute of Marine Sciences, Long Marine Lab - University of California, 100 Shaffer Road, Santa Cruz, CA 95060, USA, jmulow@ucsc.edu), Colleen Reichmuth (UCSC Institute of Marine Sciences, Long Marine Lab - University of California, 100 Shaffer Road, Santa Cruz, CA 95060, USA, coll@ucsc.edu)

Auditory sensitivity in the otariid pinnipeds (sea lions and fur seals) has traditionally been studied using a relatively small number of trained subjects and psychophysical techniques. Recent refinement of auditory evoked potential techniques with odontocete cetaceans has elevated interest in adapting these methods for sea lion subjects, with the goal of increasing sample size and efficiency in audiometric studies. To date, several basic electrophysiological characteristics of the California sea lion (*Zalophus californianus*) auditory system have been described, and these findings have allowed for the development of more advanced techniques in investigations of sea lion hearing. Most notable is the recording of the envelope following response (EFR) evoked by narrow-band, sinusoidally amplitude-modulated tones. This method can provide significant advantages in the detection of low-amplitude electrophysiological signals in noise using Fourier analysis and objective statistical detection of responses. Currently, EFR audiometry in the California sea lion and Steller sea lion (*Eumetopias jubatus*) is proving to be a promising method for rapidly assessing the variation of hearing capabilities among individuals, including the detection of hearing loss.

12:00

**4aABb5. Air and bone conduction evoked potential audiometry in the northern elephant seal.** Dorian S. Houser (Biomimetica, 7951 Shantung Dr., Santee, CA 92071, USA, dhouser@spawar.navy.mil), Daniel Crocker (Sonoma State University, Department of Biology, 1801 East Cotati Avenue, Rohnert Park, CA 94928, USA, crocker@sonoma.edu), James J. Finneran (US Navy Marine Mammal Program, Space and Naval Warfare Systems Center, 53560 Hull St., Code 71510, San Diego, CA 92152, USA, james.finneran@navy.mil)

Elephant seals (*Mirounga angustirostris*) are the largest and most aquatic of the pinnipeds, spending up to eight months of the year at sea diving to depths as great as 1600 m. The pinna is absent in the elephant seal and the middle ear cavity and auditory canal are lined with a cavernous tissue, both of which are likely adaptations to deep diving. Elephant seals demonstrate a greater sensitivity to low frequency sounds than do other pinnipeds and an overall greater sensitivity to underwater sound than to airborne sound. The relative importance of sound conduction pathways in the elephant seal is undetermined, although it has been speculated that bone conduction pathways are important to underwater hearing in this species. To compare the sensitivity of the elephant seal to both air and bone conducted stimuli, auditory evoked responses were recorded in seals exposed to signals presented through headphones and via a bone vibrator. In comparison to airborne stimuli, bone conduction methods provide an opportunity to more effectively study sensitivity to low frequency sounds, but are challenged by a lack of reference equivalent threshold sound pressure levels. Future efforts should compare bone conduction and direct field audiometry results obtained within the same individual.

**4aABb6. Preliminary Results of a Behavioral Audiometric Study of the Polar Bear.** Ann E. Bowles (Hubbs-Sea World Research Institute, 2595 Ingraham Street, San Diego, CA 92109, USA, abowles@hswri.com), Megan A. Owen (Conservation and Research for Endangered Species, Zoological Society of San Diego, PO Box 120551, San Diego, CA 92112-0551, USA, mowen@sandiegozoo.org), Samuel L. Denes (Hubbs-Sea World Research Institute, 2595 Ingraham Street, San Diego, CA 92109, USA, sdenes@hswri.org), Stefanie K. Graves (Hubbs-Sea World Research Institute, 2595 Ingraham Street, San Diego, CA 92109, USA, sgraves@hswri.org), Jennifer L. Keating (Hubbs-Sea World Research Institute, 2595 Ingraham Street, San Diego, CA 92109, USA, jenniferlkeating@hotmail.com)

The hearing of polar bears is of great interest because little is known about hearing of large terrestrial carnivores, they are amphibious, their predatory habits differ from most bears, and there is an increasing need for data to manage anthropogenic noise in maternal denning habitat. Behavioral auditory thresholds were collected from two female polar bears at the San Diego Zoo (ZSSD) in 2006-2007, and are now underway with two females and a male at SeaWorld San Diego (SWSD). Thresholds were measured at 19 frequencies between 125 Hz and 31.5 kHz using shaped 500 ms tones, a 'go/no-go' response protocol, and staircase presentation order with catch trials. Holding areas in both facilities were sound-isolated to the extent practicable. Threshold measurements were limited by background noise below 5 kHz, but sensitivity could be measured to below 0 dB at higher frequencies. To date, the bears have detected sounds down to the noise floor from 125 Hz to about 14 kHz. Their sensitivity declines rapidly above 20 kHz. The results suggest that their auditory threshold functions are narrower or shifted to lower frequencies than those of small carnivores. [Supported by Polar Bears International, ZSSD, SWSD and the author's organizations]

**4aABb7. Variation in pup vocalisations and mother-pup behaviour between harp seal whelping patches: effects of climate or geography?** Ilse Catharina Van Opzeeland (Alfred Wegener Institute, P.O. Box 120161, 27515 Bremerhaven, Germany, Ilse.Van.Opzeeland@awi.de), Peter J. Corkeron (US NOAA, Northeast Fisheries Science Center, 166 Water Street, Woods Hole, MA 02543, USA, peter.corkeron@gmail.com), Denise Risch (US NOAA, Northeast Fisheries Science Center, 166 Water Street, Woods Hole, MA 02543, USA, drisch@whsun1.wh.who.edu), Gary B. Stenson (Dept of Fisheries and Oceans, P.O. Box 5667, St John's, NL A1C5X1, Canada, StensonG@DFO-MPO.GC.CA), Sofie Van Parijs (US NOAA, Northeast Fisheries Science Center, 166 Water Street, Woods Hole, MA 02543, USA, sofie.vanparijs@noaa.gov)

Harp seals breed in pack-ice, a substrate which can vary substantially between whelping patches depending on differing environmental and oceanographic conditions. This study demonstrates clear site differences in pup vocalizations and mother pup behaviour between Northeast (Greenland Sea) and Northwest (Canadian Front) Atlantic harp seal populations. Classification trees showed a distinctive split between Front and Greenland Sea pup vocalisations. No clear sex differentiation in vocalizations was present for pups at the Front; 42% (n = 12) of male and 38% (n = 13) of female calls could be attributed to a given individual. In the Greenland Sea, 55% (n = 42) of female vocalisations were attributed to individuals compared with only 8% for males (n = 47). In addition behavioural observations of mother pup pairs were conducted (Front, n = 58; Greenland Sea, n = 78). Greenland Sea pups were found to nurse more, and were more alert than Front pups. Female attendance patterns also differed between sites: females at the Front were more likely to attend their pups than those in the Greenland Sea. This marked difference in female presence between sites could have several origins such as variability in ice conditions, predation pressure, or female condition.

## Session 4aAO

**Acoustical Oceanography, Signal Processing in Acoustics, and ECUA: Adjoint Modeling for Geoacoustic Inversion**

Paul Hursky, Cochair

*HLS Research, Inc., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA 92037, USA*

Jean-Pierre Hermand, Cochair

*Université libre de Bruxelles (U.L.B.) - Environmental hydroacoustics lab, av. Franklin D. Roosevelt 50, CP 194/5, Bruxelles, 1050, Belgium***Invited Papers**

11:00

**4aAO1. Validation of adjoint-generated environmental gradients for the acoustic monitoring of a shallow water area.** Matthias Meyer (Royal Netherlands Naval College (NLDA) - REA group, PO Box 10000, 1780 Den Helder, Netherlands, mmeyer@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), Mohamed Berrada (Laboratoire d'Océanographie et du Climat - Expérimentation et Approches Numériques, Université Pierre et Marie Curie, Tour 45-55 - 5ème étage - 4, place Jussieu, 75005 Paris, France, mohamed.berrada@locean-ipsl.upmc.fr), Mark Asch (Université de Picardie Jules Verne, LAMFA (CNRS UMR 6140), 33 Rue Saint Leu, 80039 Amiens, France, mark.asch@u-picardie.fr)

In the framework of the recent Maritime Rapid Environmental Assessment sea trial MREA07/BP'07 [Le Gac&Hermand, 2007] that was conducted in the same area south of the island of Elba as the earlier Yellow Shark trial (YS94), this paper examines the original YS94 acoustic data and the recent MREA07 oceanographic data to demonstrate adjoint-based acoustic monitoring of environmental parameters in Mediterranean shallow waters. First, adjoint-generated environmental gradients are validated for the application in geoacoustic inversion where the bottom acoustic parameters of the YS94 layered seabed are determined from the long-range waterborne propagation of a multi-frequency signal. Then, for the application in ocean acoustic tomography, the temporal variability of the MREA07/BP'07 oceanographic data is analyzed in terms of empirical orthogonal functions and the adjoint-based inversion scheme is used to track the time-varying sound speed profile of the experimental transect.

11:20

**4aAO2. Variational assimilation of simulated ocean acoustic tomography data in an ocean model.** Elisabeth Remy (Mercator-Ocean, Parc Technologique du Canal, 8-10 rue Hermès, 31520 Ramonville Saint Agne, France, eremy@mercator-ocean.fr), Fabienne Gaillard (LPO-IFREMER, BP 70, 29280 Plouzane, France, fabienne.gaillard@ifremer.fr), Jacques Verron (Laboratoire des Écoulements Géophysiques et Industriels (LEGI), BP 53, 38041 Grenoble Cedex 9, France, verron@hmg.inpg.fr)

In the concept of large scale observing system for the ocean, ocean acoustic tomography is an original tool to monitor the ocean interior. Analysis of tomographic travel time using inversion gives an estimate of the temporal evolution of the heat content along the observed sections, an important quantity to monitor the ocean climate evolution. At lower scales than the cell size defined by the observational array, it is not possible to estimate the ocean temperature field without using other sources of information. A possible approach is to combine the tomographic observations with a numerical dynamical ocean model to obtain a complete description consistent with the data on a given time interval. We propose to explore a variational method using the adjoint technic to assimilate those integral data. We studied the case of a basin scale observational array, as the one deployed in the Mediterranean sea for the Thetis 2 experiment. Only travel time anomalies due to the sea water properties are considered. The ability of tomographic data to constrain the ocean model circulation is evaluated using simulated observations with a model solution. This approach called twin experiments, allows to compare the result after assimilation with the "true" solution.

**Contributed Papers**

11:40

**4aAO3. Probabilistic PCA and Ocean Acoustic Tomography Inversion with an Adjoint Method.** Mohamed Berrada (Laboratoire d'Océanographie et du Climat - Expérimentation et Approches Numériques, Université Pierre et Marie Curie, Tour 45-55 - 5ème étage - 4, place Jussieu, 75005 Paris, France, mohamed.berrada@locean-ipsl.upmc.fr), Fouad Badran (Laboratoire CEDRIC, Conservatoire National des Arts et Métiers, 292, rue Saint Martin, 75003 Paris, France, badran@cnam.fr), Sylvie Thiria (Laboratoire d'Océanographie et du Climat - Expérimentation et Approches Numériques, Université Pierre et Marie Curie, Tour 45-55 - 5ème étage - 4, place Jussieu, 75005 Paris, France, Sylvie.Thiria@locean-ipsl.upmc.fr)

We present an Ocean Acoustic Tomography (OAT) inversion in a shallow water environment. The idea is to determine the celerity  $c(z)$ ,  $z$  is depth, knowing the acoustic pressures caused by a multiple frequencies source and collected by a sparse receiver array. The variational approach minimizes a cost function which measures the adequacy between the measurements and their forward model equivalent. This method introduces also a regularisation term in the form  $(c(z)-c_b)^T B^{-1} (c(z)-c_b)$ , which supposes that  $c(z)$  follows an a priori normal law. To circumvent the problem of estimating  $B^{-1}$ , we propose to model the celerity vectors by a probabilistic PCA. In contrast to the methods which use PCA as a regularization method and filter the useful information, we take a sufficient number of axes which allow the modelization of useful information and filter only the noise. The probabilistic PCA intro-

duces a reduced number of non correlated latent variables  $\eta$  which act as new control parameters introduced in the cost function. This new regularization term, expressed as  $\eta^T \eta$ , reduces the optimization computation time. In the following we apply the probabilistic PCA to an OAT problem, and present the results obtained when performing twin experiments.

12:00

**4aAO4. Geoacoustic Adjoint-Based Inversion via the Parabolic Equation.** John S. Papadakis (Institute of Applied & Computational Mathematics, Foundation for Research and Technology-Hellas (FORTH), Vassilika Vouton, P.O. Box 1385, GR-71110 Heraklion, Greece, johnsp@iacm.forth.gr), 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), Evangelia T. Flouri (Institute of Applied & Computational Mathematics, Foundation for Research and Technology-Hellas (FORTH), Vassilika Vouton, P.O. Box 1385, GR-71110 Heraklion, Greece, flouri@iacm.forth.gr), Matthias Meyer (Royal Netherlands Naval College (NLDA) - REA group, PO Box 10000, 1780 Den Helder, Netherlands, mmeyer@ulb.ac.be)

In this paper an analytic method is exhibited for recovering the acoustic parameters of the sub-bottom region in the ocean. The acoustic propagation problem is modeled via the wide angle parabolic equation and the bottom boundary condition used is in the form of a Neumann to Dirichlet or Dirichlet to Neumann map. The sub-bottom region is assumed homogeneous

or horizontally stratified with homogeneous layers. The inversion is modeled as an optimal control problem, and the solution is based on the adjoint method. Several cost functions are introduced which make use of the relative amplitude of the observed complex field. The method is applied to several test cases and satisfactory convergence of the inversion scheme is exhibited.

12:20

**4aAO5. Adjoint modeling for Acoustic Inversion based on an Adjoint Parabolic Equation.** John S. Papadakis (Institute of Applied & Computational Mathematics, Foundation for Research and Technology-Hellas (FORTH), Vassilika Vouton, P.O. Box 1385, GR-71110 Heraklion, Greece, johnsp@iacm.forth.gr), Evangelia T. Flouri (Institute of Applied & Computational Mathematics, Foundation for Research and Technology-Hellas (FORTH), Vassilika Vouton, P.O. Box 1385, GR-71110 Heraklion, Greece, flouri@iacm.forth.gr)

In this paper the acoustic propagation problem is modeled by the wide angle parabolic equation and the bottom boundary condition is in the form of a Neumann to Dirichlet map. We formulate the inversion as an optimal control problem, the control parameters being the sound speed in the water and the sound speed, density and attenuation in the bottom. Using the adjoint operator to the wide angle parabolic equation, an inversion scheme is derived for the acoustic parameters of the water column and the bottom region. The optimization approach used is the method of steepest decent. Several test cases are exhibited.

THURSDAY MORNING, 3 JULY 2008

ROOM 352B, 8:00 A.M. TO 12:20 P.M.

## Session 4aBB

### Biomedical Ultrasound/Bioresponse to Vibration: Theoretical and Computational Models of Ultrasonic Propagation in Bones I

James G. Miller, Cochair

*Washington University, Physics Box 1105, 1 Brookings Drive, St. Louis, MO 63130, USA*

Pascal Laugier, Cochair

*Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, Paris, 75006, France*

### Invited Papers

8:00

**4aBB1. Recent developments in modeling and measuring scattering from trabecular bone.** Keith A. Wear (U.S. Food and Drug Administration, Center for Devices and Radiological Health, 10903 New Hampshire Ave, Bldg 62, Rm 3108, Silver Spring, MD 20993, USA, keith.wear@fda.hhs.gov)

Laboratories at the US Food and Drug Administration, the University of Paris and elsewhere have investigated scattering from trabecular bone. Recent analysis of data from 23 human femur samples in vitro suggests that the Faran Cylinder Model and the Weak Scattering Model accurately predict frequency dependence of backscatter coefficient. Shear mode conversion of incident longitudinal waves may be a significant source of scattering loss. Other recent work involves the effect of errors of attenuation measurements on backscatter coefficient estimates. Backscatter measurements must be compensated for attenuation in order to estimate backscatter coefficient. However, attenuation is often overestimated because it is often measured using phase sensitive receivers that exhibit phase cancellation artifacts. Recent analysis of data from 16 human calcaneus samples in vitro suggests that backscatter coefficient estimates that are based on phase sensitive attenuation compensation tend to overestimate 1) average magnitude of backscatter coefficient at 500 kHz by a factor of about  $1.6 \pm 0.3$  (mean  $\pm$  standard deviation) and 2) average exponent (n) of frequency dependence by about  $0.34 \pm 0.12$  (where backscatter coefficient is assumed to be proportional to frequency to the nth power).



**4aBB2. The fast wave propagation in bovine cancellous bone-experiments and simulation.** Mami Matsukawa (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp), Katsunori Mizuno (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, dtg0151@mail4.doshisha.ac.jp), Yoshiki Nagatani (Nara Medical University, 840 Shijo-cho, 634-8522 Kashihara, Japan, naramed-u@nagatani.ne.jp)

Cancellous bone is comprised of a complicated network of trabeculae and has strong anisotropy and inhomogeneity. In the cancellous bone, two types of longitudinal waves, fast and slow waves, are observed when the waves propagate parallel to the trabeculae direction. Paying attention to the wave front of observed waves, we have experimentally made clear the effect of anisotropy on the fast wave speeds and shown interesting relation between the mean trabeculae length and wave speeds (Mizuno et. al., IEEE Trans., UFFC, accepted), making use of the microstructural indices of the measured bone. We then compared the experimental results of fast waves with the simulation studies, using the three dimensional X-ray CT data and the Finite-Difference Time-Domain (FDTD) method. In spite of the lack of attenuation effects in the simulation, we can find interesting correlation between the fast wave speeds obtained by experiments and simulation. In addition, the characteristic attenuation behaviors of fast wave were found in both experiments and simulations. Attenuation of fast wave is always higher in the initial state of propagation, regardless of propagation direction and samples.

### Contributed Papers

8:40

**4aBB3. Experimental confirmation of negative dispersion and Bayesian inversion of multimode propagation in a bone-mimicking phantom.** Adam Q. Bauer (Washington University, Physics Box 1105, 1 Brookings Drive, St. Louis, MO 63130, USA, abauer@hbar.wustl.edu), Christian C. Anderson (Washington University, Physics Box 1105, 1 Brookings Drive, St. Louis, MO 63130, USA, canderson@wustl.edu), Karen R. Marutyan (Department of Radiology, Washington University, 1 Brookings Drive, St. Louis, MO 63130, USA, karenmarutyan@gmail.com), G Larry Bretthorst (Department of Radiology, Washington University, 1 Brookings Drive, St. Louis, MO 63130, USA, gbretthorst@wustl.edu), Keith A. Wear (U.S. Food and Drug Administration, Center for Devices and Radiological Health, 10903 New Hampshire Ave, Bldg 62, Rm 3108, Silver Spring, MD 20993, USA, keith.wear@fda.hhs.gov), Mark R. Holland (Washington University, Physics Box 1105, 1 Brookings Drive, St. Louis, MO 63130, USA, mrh@wuphys.wustl.edu), James G. Miller (Washington University, Physics Box 1105, 1 Brookings Drive, St. Louis, MO 63130, USA, james.g.miller@wustl.edu)

Previously we demonstrated using numerical simulations that negative dispersion observed in bone can result from the interference of two propagating modes, each of which exhibits positive dispersion, consistent with the Kramers-Kronig predictions. [J. Acoust. Soc. Am. 120, EL55-61 (2006)] One goal of the present study was to demonstrate this negative dispersion experimentally using the simplest example of a bone-mimicking phantom that is capable of producing two such interfering modes. An additional goal was to establish that, with the experimental data serving as input to a Bayesian approach to the inverse problem [J. Acoust. Soc. Am. 121, EL8-15 (2007)], reliable estimates of the underlying properties of the bone-mimicking phantom could be obtained from the measured signals. The phantom consisted of a flat and parallel Plexiglas<sup>TM</sup> plate into which a step discontinuity was milled. The phase velocity and attenuation coefficient (3 to 7 MHz) of the phantom were measured with a 0.25-inch piezoelectric receiver and calculated using both broadband and narrowband data. Negative dispersion was observed at specific spatial locations near the step where the attenuation coefficient rose approximately linearly with frequency. Results demonstrate that interference between modes can result in negative dispersion and that Bayesian inversion can yield underlying material properties.

9:00

**4aBB4. Modeling of "anomalous" velocity dispersion in trabecular bone: effect of multiple scattering and of viscous absorption.** Guillaume Haiat (CNRS, Laboratoire de Recherches Orthopédiques, 10, Avenue de Verdun, 75010 Paris, France, haiat@univ-paris12.fr), Alain Lhémy (Commissariat à l'Energie Atomique, Bat 611, LIST, CEA Saclay, 91191 Gif-sur-Yvette, France, alain.lhemery@cea.fr), Frederic Padilla (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de

l'Ecole de Médecine, 75006 Paris, France, Frederic.Padilla@lip.bhdc.jussieu.fr), Pascal Laugier (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, laugier@lip.bhdc.jussieu.fr), Salah Naili (Université Paris 12, B2OA, 61, Avenue du Général de Gaulle, 94010 Créteil, France, naili@univ-paris12.fr)

Speed of sound is now widely used in the clinic to assess bone strength and susceptibility to fracture. Therefore, much effort has been spent on the understanding of ultrasonic wave propagation in trabecular bone, which is an attenuating composite material in which negative values of velocity dispersion have been measured, but remain poorly explained physically. In this work, the physical determinants of velocity dispersion are described with a 2D homogenization model of the wave propagation in trabecular bone. The medium is assumed to be constituted of infinite viscoelastic cylinders (trabeculae) immersed in a saturating viscoelastic matrix (marrow). The coupling between multiple scattering and absorption phenomena allows the computation of phase velocity and of dispersion as a function of bone properties. Negative values of velocity dispersion are predicted, in good agreement with experimental results obtained in phantoms mimicking trabecular bone. In trabecular bone, mostly negative but also positive values of velocity dispersion are predicted, which spans within the range of values measured experimentally. Scattering effects are responsible for the negative values of dispersion whereas the frequency dependence of the attenuation coefficient in bone marrow and/or in the trabeculae is shown to induce an increase of dispersion.

9:20

**4aBB5. Effects of elastic properties on the wave propagation in cancellous bones - a simulation study -.** Yoshiki Nagatani (Nara Medical University, 840 Shijo-cho, 634-8522 Kashihara, Japan, naramed-u@nagatani.ne.jp), Takashi Saeki (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, dth0915@mail4.doshisha.ac.jp), Mami Matsukawa (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp), Takefumi Sakaguchi (Nara Medical University, 840 Shijo-cho, 634-8522 Kashihara, Japan, t-saka@naramed-u.ac.jp), Hiroshi Hosoi (Nara Medical University, 840 Shijo-cho, 634-8522 Kashihara, Japan, hosoi@naramed-u.ac.jp)

For diagnosing osteoporosis, ultrasonic systems are considered a powerful tool, because ultrasonic waves strongly depend on the elasticity and structure of cancellous bones. We have reported the separation of longitudinal waves into fast and slow waves, a phenomenon that is strongly connected to the alignment of bone trabeculae. In order to understand this complicated wave propagation, we have simulated the wave propagations with the finite-difference time-domain (FDTD) method using three-dimensional X-ray CT images of actual cancellous bones. In this simulation, the effects of elastic properties in the solid portions are important. One idea is to adopt the experimentally observed ultrasonic properties of cortical bone. However, we should remind the possible problems like the elasticity difference between cancellous and cortical bones, individual differences, and anisotropy.

In this study, then, we have investigated the influences of elastic properties, Poisson's ratio, and densities of solid portion (trabeculae). As a result, we confirmed that the small changes give strong influences on not only the wave speed but also the amplitudes of fast and slow waves. The influence on the amplitudes seems to come from the changes of acoustic impedance of trabeculae. The results show the importance of elastic properties in the simulation.

9:40

#### 4aBB6. Wave propagation in cancellous bone in terms of Biot's theory.

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The paper is focused on modelling of wave propagation in cancellous bones using Biot's theory. Almost all required input mechanical and structural parameters for 31 pure femoral trabecular bone specimens were measured individually. Then frequency dependent wave parameters predicted by the model were compared with the results of ultrasonic tests performed on the same specimens. To compare the predictions to measurements, additional interactions of the plane harmonic wave with the slab of cancellous was considered. The most important finding is the significant contribution of the fluid/bone and bone/fluid boundaries on the global attenuation loss. The corrected values of attenuation coefficient are of the same order of magnitude compared to measured values. The theoretical results exhibit higher attenuation of fast wave compared to that of the slow wave in good agreement with experimental observations. Moreover the amplitude ratio of simulated time domain signals of both longitudinal waves (accordingly to the Biot's model with boundary corrections), are of the same order of magnitude compared to the amplitude ratio of experimental time records. However, an analysis in the frequency domain shows that the frequency content of the simulated pulses of the fast and slow wave differs from that observed in the experiments.

10:00

#### 4aBB7. Experimental and numerical investigation of ultrasonic transmission through the skull bone and associated temperature rise.

Mathieu Pernot (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, mathieu.pernot@espci.fr), Emmanuel Bossy (Laboratoire Photons et Matière, ESPCI/CNRS, 10 rue Vauquelin, 75231 Paris Cedex 05, France, emmanuel.bossy@espci.fr), Marie Muller (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, marie.muller@espci.fr), Christine Boué (Laboratoire Photons et Matière, ESPCI/CNRS, 10 rue Vauquelin, 75231 Paris Cedex 05, France, christine.boue@espci.fr), Jean-François Aubry (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, jean-francois.aubry@espci.fr), Mickael Tanter (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, michael.tanter@espci.fr), Mathias Fink (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, mathias.fink@espci.fr), Albert-Claude Boccara (Laboratoire Photons et Matière, ESPCI/CNRS, 10 rue Vauquelin, 75231 Paris Cedex 05, France, boccara@optique.espci.fr)

The feasibility of transcranial high-intensity focused-ultrasound (HIFU) therapy within the brain relies on the ability to transmit ultrasound through the skull bone at relatively high ultrasound power. Absorption of ultrasound through the skull bone may cause important temperature rises, and is therefore an important parameter to control. Ultrasonic measurements have shown that the ultrasound beam undergoes a significant attenuation when propagating through the skull, with values on the order of 10 to 20 dB/cm/MHz. To predict temperature rise from such values, it is fundamental to weigh the relative role of absorption to the total ultrasonic attenuation

(scattering + absorption + specular reflection). In this work, two types of numerical simulations and experiments are performed to investigate this relative role. Through-transmission of 1 MHz ultrasound was performed numerically using a 3D Finite-Difference Time-Domain (FDTD) algorithm coupled to a 3D bone model obtained from high-resolution synchrotron microtomography, and compared to experimental measurements obtained with the same bone sample. Temperature rises were numerically simulated using the 3D bone model coupled to the heat equation, and compared to infrared thermography obtained experimentally while high-intensity ultrasound was propagating through the sample.

10:20-10:40 Break

10:40

#### 4aBB8. Ultrasound simulation in the distal radius using clinical high-resolution CT images.

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The overall objective of this research is to develop an ultrasonic method for non-invasive assessment of the distal radius. The specific objective of this study was to examine the propagation of ultrasound through the distal radius and determine the relationships between bone mass and architecture and ultrasound parameters. Twenty-six high-resolution peripheral-CT clinical images were obtained from a set of subjects that were part of a larger study on secondary osteoporosis. A single mid-section binary slice from each image was selected and used in the 2D simulation of an ultrasound wave propagating from the anterior to the posterior surfaces of each radius. Mass and architectural parameters associated with each radius, including total bone mass, volume fraction, trabecular number, and trabecular thickness were computed. Ultrasound parameters, including net time delay (NTD), broadband ultrasound attenuation (BUA), and ultrasound velocity (UV) were also evaluated. Significant correlations were found between NTD and total bone mass ( $R^2 = 0.92$ ), BUA and trabecular number ( $R^2 = 0.78$ ), and UV and trabecular bone volume fraction ( $R^2 = 0.82$ ). The study shows that ultrasound measurements are correlated with bone mass and architecture at the distal radius, and thus ultrasound may prove useful as a method for non-invasive assessment of osteoporosis and fracture risk.

11:00

#### 4aBB9. Guided ultrasound wave propagation in cortical bone with microstructure using the gradient elasticity theory.

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Ultrasonic characterization of bone has been largely based on the linear theory of classical elasticity. However, the classical theory cannot adequately describe the mechanical behavior of materials with microstructure. In such materials, the stress state has to be defined in a nonlocal manner by employing theories, such as those proposed by Cosserat brothers, Mindlin (gradient elastic theory) and Eringen. In this study, we adopt the simplest form of gradient theory (Mindlin Form II) to model the cortical bone's microstructural effects in a macroscopic framework. The frequency characteristic equations are analytically derived for a bone plate (4mm thick, density 1.5 g/cm<sup>3</sup>, bulk longitudinal velocity 4107 m/s). The plate is assumed free of stresses, as in the classical Lamb problem, and free of double stresses. The volumetric strain gradient energy coefficient,  $g$  (a measure of internal length), is equal to  $10^{-4}$  and  $10^{-5}$  m, i.e. of the order of the osteons size. The velocity dispersion curves of guided waves are numerically obtained using root-finding techniques and compared with those of the Lamb waves. It is shown that microstructure affects mode dispersion by inducing both material and geometrical dispersion. In conclusion, bone models with microstructure can contribute to the interpretation of in vivo measurements.

#### 11:20

**4aBB10. Investigation of the porous network as a determinant of the overall stiffness of cortical bone: Mori-Tanaka model vs. ultrasound propagation.** Cécile Baron (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, norabelic@yahoo.fr), Quentin Grimal (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, quentin.grimal@lip.bhdc.jussieu.fr), Maryline Talmant (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, talmant@lip.bhdc.jussieu.fr), Pascal Laugier (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, laugier@lip.bhdc.jussieu.fr)

Assessing the effect of porosity on stiffness in cortical bone remains an important issue that has already been addressed with several models. The originality of the present work is to compare two models of cortical bone: one uses a realistic porous network (voxel 20 microns) reconstructed from synchrotron radiation tomography; the other considers cylindrical pores aligned in a single direction. In the first case, overall elastic properties are evaluated indirectly by means of finite difference time domain simulation of ultrasound bulk wave propagation at 1 MHz. In the second model, effective elasticity is calculated by means of a Mori-Tanaka scheme based on Eshelby solution for cylindrical inclusions with ellipsoidal cross section. Overall properties were evaluated with the two methods for 18 porosity values, each corresponding to a reconstructed bone volume. The diagonal stiffness coefficients of the overall bone material estimated with the two methods compared well. Results for the stiffness coefficient in the longitudinal bone direction are indistinguishable, which indicates that the detailed geometry and distribution of the pores have a negligible effect on the longitudinal stiffness. For the other stiffness coefficients, the Mori-Tanaka method slightly overestimates the stiffness compared to the wave propagation evaluation.

#### 11:40

**4aBB11. Numerical modelling and in-vitro studies of ultrasound signal loss across fractures in cortical bone mimics.** Victor F. Humphrey (Institute of Sound and Vibration, Univ. of Southampton, University Road, Highfield, SO17 1BJ Southampton, UK, vh@isvr.soton.ac.uk), Simon P. Dodd (University of Bath, School of Management, Claverton Down, BA2 7AY Bath, UK, S.Dodd@bath.ac.uk), Sabina Gheduzzi (University of Bath, Centre for Orthopaedic Biomechanics, Dept. of Mechanical Engineering, BA2 7AY Bath, UK, S.Gheduzzi@bath.ac.uk), James L. Cunningham (University of Bath, Centre for Orthopaedic Biomechanics, Dept. of Mechanical Engineering, BA2 7AY Bath, UK, J.L.Cunningham@bath.ac.uk), Anthony W. Miles (University of Bath, Centre for Orthopaedic Biomechanics, Dept. of Mechanical Engineering, BA2 7AY Bath, UK, A.W.Miles@bath.ac.uk)

The propagation of 200 kHz ultrasonic waves along cortical bone mimics and across a simulated fracture has been investigated using a Finite Difference numerical model. The first arrival signal (FAS) transit time and peak amplitude have been calculated as a function of range at 200kHz in order to help understand the factors that determine the propagation across a fracture. The variation in the amplitude of the first peak of the reradiated wave is studied as a function of the gap width and shape. The results compare well with experimental measurements made in vitro using an axial transmission technique on cortical bone mimics and bovine cortical bone samples. The effects of various stages of the healing process have also been considered by introducing different fracture geometries into the plate model. Changing the geometry to an external callus with different mechanical properties causes the signal loss across the fracture to reduce significantly. The most significant changes are observed to occur from the initial inflammatory stage to the formation of a callus and in the remodelling stage after a significant reduction in the size of the callus has taken place.

#### 12:00

**4aBB12. The role of bone marrow on acoustic properties of cancellous bone - finite difference time domain modelling study.** Antti S. Kallioniemi (University of Kuopio, POB 1627, 70211 Kuopio, Finland, anti.kallioniemi@uku.fi), Juha Töyräs (Kuopio University Hospital, POB 1777, 70211 Kuopio, Finland, Juha.Toyras@kuh.fi), Mikko Hakulinen (University of Kuopio, POB 1627, 70211 Kuopio, Finland, Mikko.Hakulinen@uku.fi), Jukka Jurvelin (University of Kuopio, POB 1627, 70211 Kuopio, Finland, Jukka.Jurvelin@uku.fi)

Quantitative ultrasound (US) parameters are related to structure and properties of cancellous bone. The effect of bone marrow on US propagation, i.e. absorption and scattering, is still poorly understood. However, substitution of fatty marrow with water is known to significantly affect the US parameters. The present study investigates the role of marrow on US parameters, using microtomography based 3D-finite difference time domain (FDTD) modelling. Eleven human cancellous bone samples were analysed with a micro-CT system (SkyScan 1072) to determine microstructure and morphology. Wave 3000 Pro software (Cyberlogic Inc.) was used for simulations. Models were created to simulate experimental US measurement geometry with focused 1MHz transducers. Simulations were repeated before and after replacing the marrow with water. The voxel size of the simulation mesh significantly affected sample structure and simulations. US attenuation and speed decreased and increased, respectively, when marrow was replaced with water ( $p < 0.01$ ). US reflection at sample surface and backscattering from internal structures increased ( $p < 0.01$ ) when marrow was replaced with water. Contribution of bone marrow was stronger in samples with low bone volume fraction. This implicates that inter-individual differences in the composition of marrow may significantly affect measured ultrasound parameters, especially when investigating osteoporotic bone with low density.



## Session 4aEA

## Engineering Acoustics: Sound Fields I

Raymond Kirby, Chair

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## Contributed Papers

8:00

**4aEA1. A virtual headphone based on wave field synthesis.** Klaus Laumann (Institut für Rundfunktechnik GmbH, Floriansmühlstraße 60, 80939 München, Germany, laumann@irt.de), Günther Theile (Institut für Rundfunktechnik GmbH, Floriansmühlstraße 60, 80939 München, Germany, theile@irt.de), Hugo Fastl (AG Technische Akustik, MMK, TU München, Arcisstr. 21, 80333 München, Germany, fastl@mmk.ei.tum.de)

The term "virtual headphone" refers to specially designed loudspeaker systems aiming for transmission characteristics equal to real headphones. Particularly of interest is the exact pre-filtering of the speaker signals to compensate the effect of head related transfer functions (HRTFs) between loudspeakers and ear canal. These so-called "HRTF inverse filters" are dependent on geometrical conditions and so they have to be updated with every head movement. In order to avoid problematic adaptive HRTF inverse filtering, the real loudspeakers are replaced by focussed sources generated according to the principles of Wave Field Synthesis (WFS). Head tracking controlled adjustment of driving functions allows easy source movement and thus fixed source positions in relation to the listener's ears, providing stable virtual headphone reproduction. A single static HRTF inverse filter network can be used. It is designed to ensure precise headphone equalization according to ITU-R BS.708 and offers accurate reproduction of e.g. binaural signals. A pilot study with a circular WFS array built in a panel above the listener's head has verified the functional capability of this concept.

8:20

**4aEA2. Modelling sound propagation in a waveguide containing multiple obstacles.** Raymond Kirby (Brunel University, School of Engineering and Design, Uxbridge, UB8 3PH Middlesex, UK, ray.kirby@brunel.ac.uk)

Acoustic waveguides often include relatively short area changes and/or complex non-uniform obstacles. Understanding the propagation of sound within such waveguides requires a detailed knowledge of the scattering of sound at each obstacle and how these obstacles interact with one another. Mathematically modelling sound propagation in waveguides containing multiple non-uniform obstacles is challenging, especially if one assumes that the waveguide is relatively large. Accordingly, a computationally efficient hybrid numerical method is presented here that uses the standard finite element method to model non-uniform obstacles, and maps this onto a wave-based modal solution that is used for uniform duct sections only. The hybrid method has the advantage, moreover, of removing the need to numerically enforce a non-reflecting boundary condition downstream of the obstacles, which is often encountered in studies that rely solely on the standard finite element method. In this way, transmission loss predictions for relatively large ducts and multiple obstacles may be generated efficiently, and predictions are presented here for two cylinders placed in a two-dimensional waveguide.

8:40

**4aEA3. Sound generation by airflow in a pipe having a small internal cavity.** Ulf Kristiansen (Acoustics group, Norwegian University of Science and Technology, O.S. Bragstads plass 2B, N-7491 Trondheim, Norway, ulf.kristiansen@iet.ntnu.no), Børge Nygård (Acoustics group, Norwegian

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Gas flow through corrugated pipes is known to excite strong acoustic resonances within the pipe. In an attempt to better understand the flow acoustic phenomena involved, we have investigated experimentally a short pipe (0.6m long and 0.04m diameter) having a single small (5mm long, 2.5mm deep) circumferential cavity. It was found that if placed close to the pipe's inflow end, strong acoustic resonances were generated. The experimental results were compared to a model based on describing-function theory. The model involves two transfer functions, one associated with the pipe resonator, and the other the shear layer above the cavity. These are combined to a feedback system. This model gives the frequencies generated and the acoustic pressure levels (to within a constant) for different flow velocities. Reasonable agreement was obtained between the experimental results and the model predictions.

9:00

**4aEA4. Multi-modal acoustic propagation in pipes with arbitrary defects: theory and experiments.** Raymond Kirby (Brunel University, School of Engineering and Design, Uxbridge, UB8 3PH Middlesex, UK, ray.kirby@brunel.ac.uk), Kirill V. Horoshenkov (University of Bradford, School of Engineering, Design and Technology, BD7 1DP Bradford, UK, k.horoshenkov@Bradford.ac.uk), Tareq Bin Ali (University of Bradford, School of Engineering, Design and Technology, BD7 1DP Bradford, UK, m.t.binali@Bradford.ac.uk)

Underground sewer systems are prone to flooding incidents caused by obstructions such as sediment deposits and wall deterioration. An efficient method for identifying and characterising these obstructions involves measuring the amplitudes of the reflected and transmitted acoustic normal modes excited by a point source in the sewer pipe. However, the behaviour of higher order modes in relatively large pipe work is often difficult to predict and interpret. In order to provide a greater physical insight into the measured data and to guide future experimental work, theoretical predictions have been developed and validated. The presented theoretical work is based on a finite element method and a mode matching technique. In this paper the predicted and measured sound fields are analysed for up to four acoustic modes reflected from two different obstacles (axisymmetric and non-axisymmetric) deposited in a 150 mm diameter uPVC pipe.

9:20

**4aEA5. Benchmarking for acoustic simulation software.** Alfonso R. Molares (University of Vigo, E.T.S.I de Telecomunicación, Rúa Maxwell s/n, 36310 Vigo, Spain, amolares@gts.tsc.uvigo.es), Manuel A. Sobreira-Seoane (University of Vigo, E.T.S.I de Telecomunicación, Rúa Maxwell s/n, 36310 Vigo, Spain, msobre@gts.tsc.uvigo.es)

The validation of acoustic simulation software is still an obscure and imprecise matter. Validation studies of commercial implementations are rarely provided by vendors which are reluctant to show their product weakness. On the contrary in open academic implementations validation studies are one of the keys to make them used, rather than marketing. However, they demand from users a high level of knowledge and imply, generally, so long calculation times that make them impractical for common industry purposes. The aim of this study is to contribute to clarify this point setting a simple benchmark to measure the accuracy and performance of different software pack-



ages for sound field calculations. The benchmark is presented by its application with two widely used commercial implementations of finite element method and with an open-source implementation of boundary element method developed at the Southern Denmark University. The validation is performed against analytical formulae and also against experimental results. In order to study the balance between accuracy and computational cost the results are finally presented in terms of relative error versus calculation time.

9:40

**4aEA6. Spatialized additive synthesis.** Charles Verron (Orange Labs, 2 avenue Pierre Marzin, 22307 Lannion, France, charles.verron@orange-ftgroup.com), Mitsuko Aramaki (CNRS - INCM and Université de Provence, 31, chemin Joseph Aiguier, 13402 Marseille, France, aramaki@lma.cnrs-mrs.fr), Richard Kronland-Martinet (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, kronland@lma.cnrs-mrs.fr), Grégory Pallone (Orange Labs, 2 avenue Pierre Marzin, 22307 Lannion, France, gregory.pallone@orange-ftgroup.com)

In virtual auditory environments, a spatialized sound source is typically simulated in two stages: first a "dry" monophonic signal is recorded or synthesized, and then spatial attributes (directivity, width and position) are applied by specific signal processing algorithms. In this paper, a unified analysis/spatialization/synthesis system is presented. It is based on the spectral modeling framework that analyses/synthesizes sounds as a combination of time-varying sinusoidal, noisy and transient contributions. The proposed system takes advantage of this representation to allow intrinsic parametric sound transformations, such as spatial distribution of sinusoids or diffusion of the noisy contribution around the listener. It integrates timbre and spatial parameters at the same level of sound generation, so as to enhance control capability and computational performance.

10:00

**4aEA7. Notes on the reproduction of moving virtual sound sources.** Jens Ahrens (Deutsche Telekom Laboratories, Ernst-Reuter-Platz 7, 10587 Berlin, Germany, jens.ahrens@telekom.de), Sascha Spors (Deutsche Telekom Laboratories, Ernst-Reuter-Platz 7, 10587 Berlin, Germany, sascha.spors@telekom.de)

Wave field synthesis and higher-order Ambisonics are two representatives of sound reproduction systems that are based upon the concept of physical recreation of a desired sound field. Conventional implementations of such systems typically reproduce moving virtual sound sources as a concatenation of static source positions that change over time. This approach introduces various artifacts which are reported in the literature to be strongly audible. It was recently shown by the authors that the explicit consideration of the physical properties of the sound field of moving sources in the reproduction algorithm avoids these artifacts. It thus allows for the accurate reproduction of the Doppler Effect. In practical implementations of such sound field reproduction systems unavoidable artifacts arise. These are mainly a consequence of sampling and truncation of the loudspeaker distribution and appear both for static and moving virtual sound sources. For static sources, they are well documented in the literature. We revisit these investigations and point out the particularities of these artifacts with respect to the time-variant property of the reproduced sound field.

10:20-10:40 Break

10:40

**4aEA8. Loudspeakers simulation of sound environments for the car industry.** Benoît Gauduin (Genesis S.A., Bâtiment Gérard Mégie, Domaine du Petit Arbois - BP 69, 13545 Aix-en-Provence Cedex 4, France, benoit.gauduin@genesis.fr), Sylvain Hourcade (Genesis S.A., Bâtiment Gérard Mégie, Domaine du Petit Arbois - BP 69, 13545 Aix-en-Provence

Cedex 4, France, sylvain.hourcade@genesis.fr), Nathalie Le Hir (Renault, TCR/AVA 163, Technocentre, 78288 Guyancourt Cedex, France, nathalie.le-hir@renault.com), Gaël Guyader (Renault, TCR/AVA 163, Technocentre, 78288 Guyancourt Cedex, France, gael.guyader@renault.com)

Car manufacturers are strongly interested in sound quality: they need to understand the expectations of their customers in terms of engine sound signature, door closure noise, etc. ... In order to allow engineering teams of research departments to perform psychoacoustics tests under controlled and realistic conditions, GENESIS supplies high fidelity listening 3D simulators on loudspeakers. Based on binaural recordings, these simulators, called transaural, are calibrated to obtain the best fidelity in terms of spatial and spectral components. In this presentation, we will discuss the requirements regarding sound system and room installation and present the precision reachable in terms of frequency response. Measurements and results achieved on such systems will be shown, with a particular focus on the two systems built for RENAULT and GENESIS. Finally, in order to perform equivalent psychoacoustics tests on both RENAULT and GENESIS sites, we will develop the procedure of comparison and validation of both transaural systems.

11:00

**4aEA9. Development of a simple and accurate approximation method for the Gaussian beam expansion technique.** Wei Liu (Institute of Acoustics, Chinese Academy of Sciences, Bei-Si-Huan-Xi Road, 100080 Beijing, China, liuwei9811@gmail.com), Peifeng Ji (Institute of Acoustics, Chinese Academy of Sciences, Bei-Si-Huan-Xi Road, 100080 Beijing, China, jipeifeng@mail.ioa.ac.cn), Jun Yang (Institute of Acoustics, Chinese Academy of Sciences, Bei-Si-Huan-Xi Road, 100080 Beijing, China, junyang.ioa@gmail.com)

The calculation of the sound field can be greatly simplified by using the Gaussian beam expansion technique. The source distribution function is expressed as the superposition of a small number of Gaussian functions, and the expansion coefficients could be obtained by minimizing an object function in the spatial or k-space domain. In this paper, a fast algorithm is developed to determine the Gaussian function coefficients for a more accurate approximation. Two-stage procedures are employed in the proposed method. Firstly, two real coefficients are estimated by a simple search approach, and then the least mean square (LMS) algorithm is adopted for determining the optimal expansion coefficients. Finally, the presented method is evaluated in the case of calculation of sound fields radiated from a piston and a rectangular planar source. Simulation results show that, compared with the previous approaches, the developed scheme is simple to implement with high accuracy.

11:20

**4aEA10. Determination of Condition for Fastest Negative Group Velocities of Lamb-Type Waves under each Density Ratio of Solid and Liquid Layers.** Kojiro Nishimiya (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, nishimiya@aclab.esys.tsukuba.ac.jp), Koichi Mizutani (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, mizutani@esys.tsukuba.ac.jp), Naoto Wakatsuki (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, wakatsuki@iit.tsukuba.ac.jp), Ken Yamamoto (Kansai Univ., 3-3-35 Yamate-cho, 564-8680 Suita, Japan, ken@ipcku.kansai-u.ac.jp)

Lamb-type waves are coupling modes of leaky Lamb waves on a layer structure. The Lamb-type waves have complicated propagation characteristics more than ordinary Lamb waves on a uniform elastic plate. In the characteristics, we examine the negative group velocities. Generally, the negative group velocities of Lamb waves are slower than positive group velocities under the same condition. If the negative group velocities are applied to fabricating some new application, it is desired that the speeds of negative group velocities are comparable to those of positive group velocities. Consequently, we aim to obtain the faster negative group velocities. Lamb-type waves show more discriminative characteristics in negative group velocities than ordinary Lamb waves. In this research, we consider the Lamb-type waves in a solid/liquid/solid structure. It is de-

scribed the conditions for obtaining the fastest negative group velocities of Lamb-type waves. The conditions, which are the acoustical impedance ratio, are expressed as the function of the density ratio of solid and liquid layers. These results are verified by numerical calculations.

11:40

**4aEA11. Acoustic characterization of thin polymer layers for Love mode surface acoustic waveguide.** Laurent Robert (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, laurent.robert@femto-st.fr), Lamia El Fissi (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, lamia.elfissi@femto-st.fr), Jean-Michel Friedt (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, jmfriedt@femto-st.fr), Frederic Cherioux (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, frederic.cherioux@femto-st.fr), Sylvain Ballandras (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, sylvain.ballandras@femto-st.fr)

We investigate the use of thin (1-10  $\mu\text{m}$ ) polymer films as guiding layer for Love mode surface acoustic wave sensors. Beyond the great gravimetric sensitivity provided by the polymer guiding layer resulting from the low acoustic velocity, the use of photoresists provide economical means of depositing guiding layers of optimal thicknesses compared to inorganic layer deposition processes (typically PECVD deposition of silicon dioxide lasting several hours). The limit of very thick ( $> 100 \mu\text{m}$ ) layers provides means of propagating interface waves mostly insensitive to the environment (package-less sensors) whose properties only vary through modification of the bulk properties of the polymer. We here analyse the evolution of the properties of the guiding layer in terms of acoustic velocity and losses as a function of time (solvent evaporation following photoresist spin coating) and temperature (typical baking steps). The polymer films is deposited on AT-cut quartz patterned with interdigitated transducers for generating 40  $\mu\text{m}$ -wavelength shear waves converted to a guided Love mode in a delay line configuration. We complete the experimental results with data interpretation using a model of acoustic wave propagation yielding quantitative results including viscosity and density out of the velocity and insertion loss measurements, both for the guided Love mode and interface layers

12:00

**4aEA12. From frequency to time domain: Signal features and physical characteristics for resonant acoustical systems.** Samuel Rodriguez (Renault - Laboratoire PHASE, Centre Technique Renault - CTL L16 1 29, 1, allée Cornuel, 91510 Lardy, France, rodriguez.samuel@yahoo.fr), Vincent Gibiat (Université Paul Sabatier, PHASE, 118, route de Narbonne, 31062 Toulouse cedex 9, France, gibiat@cict.fr), Stephane Guilain (Renault - Laboratoire PHASE, Centre Technique Renault - CTL L16 1 29, 1, allée Cornuel, 91510 Lardy, France, stephane.guilain@renault.com), Alain Lefebvre (Renault - Laboratoire PHASE, Centre Technique Renault - CTL L16 1 29, 1, allée Cornuel, 91510 Lardy, France, alain.a.lefebvre@renault.com)

The determination of the impulse response or the reflection function of an acoustical system from data expressed in the frequency domain is not immediate. Signal processing from frequency domain to time domain should involve phenomena of large amplitude as oscillations known as "ripple" that does not correspond to any physical phenomenon. The "ripple phenomenon" will be analyzed from both a signal processing and a physical point of view with the help of simple duct acoustic examples. A map designed as a new time-frequency tool helps us to show that it cannot be removed in most cases without the use of processing techniques involving modifications in the computed signal. This work has been developed for the automotive research, but can be applied to musical acoustics or to any field connected with time domain exploration of acoustic cues.

12:20

**4aEA13. Extension of Optimal Source Distribution principle.** Takashi Takeuchi (OPSODIS Limited, c/o ISVR, University of Southampton, Highfield, SO17 1BJ Southampton, UK, tt@isvr.soton.ac.uk)

Binaural reproduction over loudspeaker requires system inversion which is often referred to as cross-talk cancellation. Such process is the major factor to degrade the quality of 3D sound reproduction but Optimal Source Distribution (OSD) provides simple and effective loudspeaker design principle and signal processing which enables lossless crosstalk cancellation process. OSD takes advantage of its physical property where in-phase and out-of-phase components of the binaural reproduction process are balanced, hence the bulk of the crosstalk cancellation is achieved by its loudspeaker design principle and natural interference in the sound field. It is also shown that the advantage of OSD is further enhanced by separating in-phase and out-of-phase components through its loudspeaker design.

**Session 4aEDa****Education in Acoustics: Acoustics in the Public School Science Classrooms**

Uwe J. Hansen, Cochair

*Indiana State University, 64 Heritage Dr, Terre Haute, IN 47803, USA*

Malte Kob, Cochair

*RWTH Aachen, Dept. of Phoniatrics, Pedaudiology, and Communication Disorders, Pauwelsstr. 30, Aachen, 52074, Germany****Invited Papers*****8:00**

**4aEDa1. Acoustics in the public school classroom.** Uwe J Hansen (Indiana State University, 64 Heritage Dr, Terre Haute, IN 47803, USA, u-hansen@indstate.edu), Corinne Darvennes (Tennessee Tech Univ., Dept. of Mechanical Engineering, PO Box 5014, Cookeville, TN 38501, USA, cdarvennes@ntech.edu)

As mentioned in a DAGA 07 paper 1, physical science preparation in American schools often leaves something to be desired. In recent years the ASA Committee on Education in Acoustics has made an effort to participate in finding relief for that problem. Three approaches have been of some influence. 1. Teacher workshops; 2. Hands-on student sessions. 3. Secondary school curriculum input. Teacher workshops have emphasized music as a vehicle to introduce science in the elementary classroom. Hands-on student sessions have included about 20 acoustics experiments of varying degrees of sophistication for students both in high school physics classes and in elementary general science classes. Secondary curriculum input has included both, development of laboratory experiments in acoustics, and exposure to relatively low cost educational versions of computational software. Examples of teacher workshop content, a number of hands-on experiments, and some finite element calculations will be discussed. 1Musik: Zugang zur Wissenschaft in der Grundschule. Uwe J. Hansen, DAGA 2007, Stuttgart pp. 171-172.

**8:20**

**4aEDa2. Math and science partnership program in the Upper Cumberland districts of Tennessee.** Corinne Darvennes (Tennessee Tech Univ., Dept. of Mechanical Engineering, PO Box 5014, Cookeville, TN 38501, USA, cdarvennes@ntech.edu)

The Mathematics and Science Partnership (MSP) program by the US Department of Education "is intended to increase the academic achievement of students in mathematics and science by enhancing the content knowledge and teaching skills of classroom teachers. Partnerships between high-need school districts and the science, technology, engineering, and mathematics (STEM) faculty in institutions of higher education are at the core of these improvement efforts." This paper will present efforts in the Upper Cumberland districts of Tennessee to introduce engineering applications to Math and Science teachers in grades 8-12. The science teachers recruited represented the disciplines of Chemistry, Physics, and Physical Science. The 3-year program consisted of summer institutes, as well as bi-monthly Saturday workshops. An overview of the program will be discussed. It will include background information about the Upper Cumberland region, Tennessee content standards related to acoustics, working with teachers, some of the hands-on activities that were used during summer workshops, and the equipment that was provided to the teachers.

**8:40**

**4aEDa3. Teaching Communication Acoustics and Physiology at the girl's day.** Malte Kob (RWTH Aachen, Dept. of Phoniatrics, Pedaudiology, and Communication Disorders, Pauwelsstr. 30, 52074 Aachen, Germany, mkob@ukaachen.de)

The girl's day is an annual event for young female pupils who wish to have their own experiences during a day in a potential future work place. In our department a team of 19 phoniatricians, speech therapists, phoneticians, engineers, audiometrists and assistants works on diagnosis and therapy of disorders of voice, speech, hearing and swallowing. For several years, our department has been visited by a group of 16 to 20 pupils who are guided in small groups through a sequence of stations with hands-on exercises in the field of voice and hearing acoustics. The experiments include the measurement of a voice range profile, recording and visualisation of voice signals, subjective and objective assessment of voice disorders, and listening tests. The experiences with organisation and feedback of the girl's days are reported.

## Contributed Paper

9:00

**4aEDa4. Investigating musical sound as a model for the scientific process.** Andrew Piacsek (Central Washington University, Department of Physics, 400 E. University Way, Ellensburg, WA 98926, USA, piacsek@cwu.edu)

In recent years, science pedagogy at all levels has embraced active student learning, in which students are engaged in the process of discovery, rather than passively receiving information. One of the benefits of active learning in a science curriculum is that students have an opportunity to emulate scientists in their approach to producing knowledge. It is desirable that students at all levels, including those in primary and secondary schools, develop an understanding of (or at least an appreciation for) the scientific pro-

cess, in addition to learning science content. Presented here is a description of a musical acoustics curriculum used at Central Washington University for non-science students, with emphasis on how the structure of the curriculum and the active-learning elements contribute to accomplishing the objectives of conceptual learning, problem-solving ability, and scientific thinking. An essential feature of this curriculum is the combination of guided and open scientific investigations by teams of students. Portions of this college curriculum have been successfully adapted to the high school classroom and to audiences of children as young as ten years of age. Because musical acoustics incorporates many fundamental topics in physics and engineering, and it is appealing and relevant to students of all ages, this topic can be a significant asset to any science curriculum.

## Invited Paper

9:20

**4aEDa5. Acoustics Modules Developed in the IIT Research Experience for Teachers Program.** Ralph T. Muehleisen (Illinois Institute of Technology, Civil, Architectural and Environmental Engineering, 3201 S. Dearborn St., Room 228, Chicago, IL 60616, USA, muehleisen@iit.edu)

From 2004-2008, the Illinois Institute of Technology (IIT) received funding from the US National Science Foundation (NSF) to institute a program to bring K-12 (primary and secondary school) teachers into University research labs for a seven week summer program where teachers work with an IIT faculty Research Mentor and develop an educational module for their students based on their mentor's research. The module's goal is to introduce students to engineering design concepts, utilizing scientific inquiry techniques, and incorporating an ethics component and a design project. Additionally, the modules must be linked to the Illinois State Board of Education Learning Standards. In 2005, a teacher developed a module for teaching high school algebra, geometry, and physics using musical acoustics, culminating with a woodwind instrument design project. In 2006, a teacher developed a module for a 3rd grade science class that explores the basics of sound and hearing, culminating in a noise control design project. This paper will present and discuss these two acoustic teaching modules. This project was supported by NSF grant EEC-0502174.

9:40-10:00 Break

## Contributed Papers

10:00

**4aEDa6. Inquiring activities on the acoustic phenomena at the classroom using sound card in personal computer.** Young H. Kim (Korea Science Academy, 111 Backyangkwanmoonro, Busanjin-ku, 614-822 Busan, Republic of Korea, youngkim@paran.com)

Inquiring activities on acoustic phenomena have been carried out in the classroom of a high school for highly gifted children. Instead of expensive instruments such as function generators and oscilloscopes, sound cards, installed in a personal computer, were employed for the generation and detection of sound. The stereo function of the sound card offered two sound sources, so that phenomena of interference and beats can be realized in the classroom. The record function of the sound card offered detection of sound, permitting frequency spectrum analysis of sounds from two tuning forks or the sound from a moving fork. Using sound card, a lot of acoustic phenomena can be demonstrated in the classroom. In addition, sound from Rijke tube, which is a typical thermacoustic phenomena, was analyzed by using a sound card. Pop-up sound of a wine bottle and breaking of wine glasses, which are related to resonance and standing waves, were also investigated. Curiosity of students was greatly increased through a series of inquiring activities with sound cards, so that they were completely absorbed in research on acoustics.

10:20

**4aEDa7. Ur[ban]sonate: Echoes of twentieth-century sound art in the urban elementary classroom.** Kevin N. Summers (101 Woodbine Ave., Apt. 305, Syracuse, NY 13206, USA, knsummers@hotmail.com), Jason E. Summers (ARiA Consulting LLC, 1222 4th St. SW, Washington, DC 20024, USA, jesummers@alumni.kenyon.edu)

The sonic arts provide a variety of cross-disciplinary entry points into the traditional lower-elementary curriculum. The exploration of sound poetry provides students with a workspace to deconstruct the connection between written and spoken language; phonemes; phonics; voiced, unvoiced,

stressed and unstressed letter sounds and blends. Kurt Schwitters's "Ursonate," the music and scores of John Cage, various other sound art works and Deep Listening (TM) techniques provide a bridge to the study of acoustic sound and serve to validate/contextualize students' desires to creatively explore physical phonemes such as the relationship of pitch to frequency of vibration. After creating symbolic representations of sounds, writing sound poems and creating musical instruments, students synthesize learned material by scoring and directing the performance of original sound art pieces.

10:40

**4aEDa8. Acoustics in the partial deaf student school music classrooms.** Filiz Bal Kocyyigit (Karabuk University, 232. Str. No: 5/5 Ilkbahar Mah., Cankaya, 06550 Ankara, Turkey, filizbkocyyigit@yahoo.com), Kubra Sevim X. Gulec (Karabuk University Fine Arts and Design Faculty, Safranbolu, 78600 Karabuk, Turkey, kubsev@gmail.com)

Music schools need some special acoustical design for good education. If this is also for children with disabilities, the effect is more important. Research is about musical education for partially hearing disabled children. At this point architectural acoustic design acquires importance. This is not only in the shape of the room but also in using covering materials and other solutions. The question is whether architectural acoustics solutions can affect the music education of partially hearing disabled children or not. Here the first discussion is whether we need some additional components of good classroom acoustics for partially hearing disabled children in the American National Standards Institute Standard S12.60 2002 for classroom acoustics, and the cost impact of the Standard. Childhood hearing loss is a widespread problem with significant impact, an invisible condition resulting in communication problems that can ultimately interfere with learning and social development. Included are audio files that illustrate that even a mild hearing loss can have a significant impact on a child's ability to understand the teacher. Especially children with a partial hearing loss need a more lively room to amplify the instrument sound for improved hearing capacity.

4a THU. AM



11:00

**4aEDa9. The energy flow for a spherical acoustic lens: ray vs. wave methods.** Cleon E. Dean (Physics Department, Georgia Southern University, P. O. Box 8031, Statesboro, GA 30460-8031, USA, cdean@GeorgiaSouthern.edu), James P. Braselton (Department of Mathematical Sciences, Georgia Southern University, P.O.B. 8093, Statesboro, GA 30460-8093, USA, jbraselton@GeorgiaSouthern.edu)

A simple classroom demonstration consists of a weather balloon filled with carbon dioxide, a sound source, and a microphone. Since the speed of

sound is slower in carbon dioxide than in air at room temperature and pressure, the balloon acts as a positive spherical acoustic lens. The accuracy of ray methods in locating the acoustic focus versus a full blown wave solution approach is probed. This problem presents particular difficulties if the sound source lies in the near field region. The sound emitter is treated as a dipole source equivalent to a rigid oscillating sphere of small size and amplitude of motion relative to the scatterer. The energy flux around the balloon is visualized by both ray methods and by the acoustic Poynting vector field. The geometrical ray results and the acoustic Poynting vector field resulting from the wave solution are compared.

THURSDAY MORNING, 3 JULY 2008

ROOM 202/203, 11:20 TO 12:20 P.M.

### Session 4aEDb

#### Education in Acoustics: Take 5's

Uwe J Hansen, Cochair

*Indiana State University, 64 Heritage Dr, Terre Haute, IN 47803, USA*

Malte Kob, Cochair

*RWTH Aachen, Dept. of Phoniatrics, Pedaudiology, and Communication Disorders, Pauwelsstr. 30, Aachen, 52074, Germany*

Thomas D. Rossing, Cochair

*Stanford University, CCRMA, Department of Music, Stanford, CA 94305, 26464 Taaffe Rd, Los Altos Hills, CA 94022, USA*

For a Take-Five session no abstract is required. We invite you to bring your favorite acoustics teaching ideas. Choose from the following: short demonstrations, teaching devices, or videos. The intent is to share teaching ideas with your colleagues. If you can, bring a brief, descriptive handout with enough copies for distribution. Spontaneous inspirations are also welcome. You sign up at the door for a five-minute slot before the session starts. If you have more than one demo, sign up for non-consecutive slots.

## Session 4aMU

## Musical Acoustics: Virtual Musical Instruments I

Julius O. Smith, Cochair

Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180,  
USA

Antoine Chaigne, Cochair

ENSTA, Chemin de la Hunière, Palaiseau, 91761, France

## Invited Papers

8:00

**4aMU1. Interacting with virtual musical instruments at the junction nodes.** Cumhur Erkut (Helsinki University of Technology (TKK), Lab. Acoustics and Audio Signal Processing, P.O. Box 3000, FI-02015 TKK Espoo, Finland, Cumhur.Erkut@tkk.fi), Antti Jylhä (Helsinki University of Technology (TKK), Lab. Acoustics and Audio Signal Processing, P.O. Box 3000, FI-02015 TKK Espoo, Finland, antti.jylha@tkk.fi), Matti Karjalainen (Helsinki University of Technology (TKK), Lab. Acoustics and Audio Signal Processing, P.O. Box 3000, FI-02015 TKK Espoo, Finland, matti.karjalainen@tkk.fi)

Sound synthesis by block-based physical modeling of musical instruments separates the tasks of component modeling and managing their interactions. The components are the exciters or the resonators, and their interactions are managed by explicit interaction blocks, which are obtained from the physical continuity and energy conservation rules. Well-known examples of the interactors include the wave- digital adaptors and the digital waveguide scattering junctions. When the virtual instruments need to be interfaced to the outside environment with sensors and actuators for bidirectional interaction, it is advantageous to reformulate the interactors to accept and provide signal inputs and outputs, respectively. In this contribution, we refer to these elements as nodes, introduce different types of nodes, and discuss their interconnection.

8:20

**4aMU2. Block based physical modeling for virtual musical instruments.** Rudolf Rabenstein (University Erlangen-Nürnberg, Cauerstr.7, D-91058 Erlangen, Germany, rabe@LNT.de)

A variety of methods for physical modeling sound synthesis has been developed so far, mostly for single sound objects like strings, plates, etc. However, complex virtual musical instruments require not only advanced models but also methods for combining one or more sound objects with excitation mechanisms and resonating structures. This presentation shows how to derive modeling blocks from basic physical laws and how to connect them in a physically meaningful way. The first step is to establish a physical model of a dynamical structure in terms of potential and flow variables, like force and velocity. It is important to observe also the boundary conditions because firstly they shape the spectrum of the vibrating structure and thus the timbre of the sound and secondly they determine the exchange of energy with the environment. The second step is the discretization of the physical model. This procedure is shown for the functional transformation method, which delivers discrete-time models with direct access to the parameters of the physical model and which reproduces the original sound spectrum. In the last step, the resulting modeling blocks are connected by scattering elements which reflect energy back into the model or transmit it to neighboring blocks.

8:40

**4aMU3. Cymbal Synthesis.** Stefan Bilbao (University of Edinburgh, Room 7306B, JCMB, King's Bldgs., Mayfield Rd., EH9 3JZ Edinburgh, UK, sbilbao@staffmail.ed.ac.uk)

Time domain sound synthesis based on a physical model of the cymbal presents special problems, due to the need for a strongly nonlinear model of shell vibration. When standard numerical methods such as finite difference schemes are employed, various computational issues arise; among these are numerical stability, a proper treatment of numerical boundary conditions, which are nontrivial at the free edge and center of the cymbal, and the extra concern of working in polar coordinates. Coupling with mallet and bow models, possibilities for increased computational efficiency using spectral methods, and general strengths and weaknesses of difference methods in this context will be discussed.

9:00

**4aMU4. Passive admittance synthesis for sound synthesis applications.** Balázs Bank (University of Verona, Ca' Vignal 2, strada le Grazie 15, 37134 Verona, Italy, bank@mit.bme.hu), Matti Karjalainen (Helsinki University of Technology (TKK), Lab. Acoustics and Audio Signal Processing, P.O. Box 3000, FI-02015 TKK Espoo, Finland, matti.karjalainen@tkk.fi)

In physics-based sound synthesis, it is in general possible to incorporate a mechanical or acoustical admittance/impedance in the form of a digital filter. Examples include modeling of the termination of a string or a tube. However, when digital filters are fitted to measured admittance or impedance data, care has to be taken that the resulting filter corresponds to a passive mechanical or acoustical

system, otherwise the stability of the instrument model is at risk. This paper presents a simple method for designing inherently passive admittance or impedance filters. The admittance/impedance is composed as a linear combination of positive real (PR) functions, and the weights are determined by a constrained least squares optimization. The resulting filter is a parallel set of second-order sections. For wave-based modeling, such as digital waveguides (DWGs) or wave digital filters (WDFs), the admittance/impedance is converted to a reflectance filter. The parallel filter structure is retained during conversion. As an example, a guitar model based on DWG approach is presented, using mechanical admittance measurements of a guitar bridge behavior. The model is implemented as an efficient real-time sound synthesis algorithm.

9:20

**4aMU5. Modal parameter estimation for shape-changing geometric objects.** Cynthia Maxwell (Center for New Media and Audio Technologies, 1750 Arch Street, Berkeley, CA 94720, USA, cynthia@code404.com)

As a novel advancement in interactive sound synthesis, we would like to change the shape of a finite element model of an instrument and hear how the sound changes in real-time. Traditional modal synthesis methods require computing a new eigendecomposition for each geometric change -- a costly computation using today's hardware. However, by using the modes computed for one geometry to estimate modal frequencies for other nearby geometries, we can instantly hear the effects of changing the instrument shape on the sound produced. In this talk, we describe the process of estimating resonant frequencies of an instrument by combining information about the modes of similar instruments. We also propose a method for transferring the modal information from one finite element mesh to another. This method is used in situations where severe modifications to the object's geometry distort the finite element mesh and require an entirely new mesh to be created. We also describe the balance between computational speed and numerical accuracy of the computed resonances.

9:40

**4aMU6. Applications of passivity theory to the active control of acoustic musical instruments.** Edgar Berdahl (Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180, USA, eberdahl@ccrma.stanford.edu), Guenter Niemeyer (Stanford Univ., Mech. Eng., Bldg. 530, Stanford, CA 94305, USA, gunter.niemeyer@stanford.edu), Julius O. Smith (Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180, USA, jos@ccrma.stanford.edu)

The dynamic behavior of any acoustic musical instrument can be modified by closing a feedback loop around even a single sensor and actuator. The ultimate goal is to make the acoustics of the instrument programmable by way of a digital feedback controller, while the instrument retains its tangible form. In this talk, we describe a class of controllers that are applicable to passive acoustic musical instruments, and we present sound examples from laboratory experiments on a vibrating string. First, we briefly introduce positive real functions. Next, we design positive real controllers allowing the quality factor and resonant frequency of instrument modes to be individually controlled. Because positive real controllers are passive, they are stable if the instrument is passive. This means that neither a full instrument model nor complete state measurements are required. Finally, we describe a class of simple passive nonlinear controllers that can emulate various kinds of friction, stiffening and softening springs, etc. Passivity of these controllers follows from the local passivity of the controller components. Controller parameters may often be tweaked so that the controllers are no longer passive but still perform useful functions, such as bowing emulation.

10:00-10:20 Break

## Contributed Papers

10:20

**4aMU7. Vibrating-String Coupling Estimation from Recorded Tones.** Nelson Lee (Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180, USA, nalee@stanford.edu), Julius O. Smith (Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180, USA, jos@ccrma.stanford.edu)

Coupling of vibrational polarizations in a single string, for an instrument such as the acoustic guitar, produces psychoacoustically significant effects such as beating and two-stage decay (Weinreich, JASA v62 n6). Previous considerations of string coupling phenomena appear not to have addressed the practical problem of calibrating computational models based on recorded tones. In this work, we take a data-driven approach using measured data from a vibrating string from an acoustic guitar, the motion of the string in two orthogonal planes, and formulate a regularized least-squares problem for computing the coupling between the measurements. Such a formulation ensures that the resulting coupling is physically admissible, in that the resulting coupling factors do not generate energy, and are easily found as the problem is convex. Well-studied algorithms for solving convex problems, such as interior-point and gradient descent methods can be used and are widely available in the form of open-source libraries.

10:40

**4aMU8. Sound synthesis of circular plates by finite differences.** Kevin Arcas (ENSTA, Chemin de la Hunière, 91761 Palaiseau, France, arcas@ensta.fr), Antoine Chaigne (ENSTA, Chemin de la Hunière, 91761 Palaiseau, France, antoine.chaigne@ensta.fr), Stefan Bilbao (University of Edinburgh, Room 7306B, JCMB, King's Bldgs., Mayfield Rd., EH9 3JZ Edinburgh, UK, sbilbao@staffmail.ed.ac.uk)

This paper shows a method for simulating linear flexural vibrations of circular plates by finite differences (FD) for the purpose of sound synthesis. The vibrations are assumed to follow the Kirchhoff-Love model. In order to solve the continuous problem numerically, the equations are approximated in space and time by FD methods. Two schemes are presented and compared; depending on the coordinate system used for the grid, rectangular and polar, respectively. Cartesian FD are not easily adaptable to circular boundary conditions and generic conservative boundary conditions cannot be found. On the contrary, polar FD allow to find well-adapted conservative boundary conditions. With a polar grid, the distance between consecutive gridpoints decreases from the edge to the center. As a consequence the stability of the algorithm depends on the minimum radius of the grid, where this distance is the smallest. Because of this highly restrictive stability condition, numerical dispersion is high and the high-frequency content of the spectrum is badly reproduced. To avoid this problem an implicit polar FD

scheme has been developed which yields simulations with acceptable numerical dispersion. The accuracy of the algorithm is estimated by computing the ratio between numerical and analytic eigenfrequencies in a simple case.

11:00

**4aMU9. Digital algorithm for sound synthesis: Realism and complexity for creativity.** Joël Bensoam (IRCAM, 1, place Igor Stravinsky, 75004 Paris, France, bensoam@ircam.fr)

Modalys, a sound synthesis software developed at Ircam for research and musical applications, make it possible to build virtual instruments based on physical models in order to obtain the broadest range of expressive variations in the instrument in response to intuitive controls. An instrument, as a complex structure, is described by the mechanical/acoustical interactions of its components (strings, tubes, soundboard, 3D FEM objects...). New research has recently been carried out in two directions: On one hand, a generic digital algorithm able to treat a large class of non linear interaction models (lip-reed, contact and friction, aero-acoustics jets, etc) was built. Based on a Gauss-Siedel strategy, this algorithm is used indifferently regardless of the modeled structure's complexity and guarantees convergence and robustness of the whole sound synthesis. On the other hand, efforts are made to infer from measurements the physical parameters needed for the sound synthesis (geometric or gesture parameters for example). Due to the lack of a complete analytical formulation (digital algorithm), derivative based methods to solve inverse problem (Newton, gradient, etc) are not allowed. Evolution Strategy (ES), especially for multiobjective optimization, are then investigated.

11:20

**4aMU10. Application of Volterra series to simulate dynamics of a Reissner beam.** David Roze (IRCAM, 1, place Igor Stravinsky, 75004 Paris, France, david.roze@ircam.fr), Thomas Hélie (IRCAM, 1, place Igor Stravinsky, 75004 Paris, France, thomas.helie@ircam.fr), Joël Bensoam (IRCAM, 1, place Igor Stravinsky, 75004 Paris, France, bensoam@ircam.fr)

Sound synthesis of strings needs the use of nonlinear models to provide realistic sounds according to the amplitude and the method of excitation (pluck, bow...). In order to include coupling between the different degrees of freedom, the Reissner beam model will be used to simulate nonlinear dynamics of a string. Expression of equilibrium of Reissner beams using Lie groups and algebra allows us to write an exact, simple formulation including nonlinearities due to large strains. This work aims to apply Volterra series method to this formulation in order to perform simulation of a Reissner beam at a given order of nonlinearity. Volterra series requires a rewriting of the model with well-defined boundary and initial conditions and interconnection laws of the series adapted to the variables (vector and matrix). Once the linear part isolated and studied (calculation of the Green function, and therefore of the first kernel), nonlinear terms must be organized to define a recurrent relation and solve the kernel equation. Finally, it can be possible to identify a structure of simulation using instantaneous sums and products of outputs of linear filters.

11:40

**4aMU11. Singing Integrals or wind instruments modeling using Boundary Integral Equations.** Umberto Iemma (University Roma Tre, via vasca navale 79, 00146 Rome, Italy, u.iemma@uniroma3.it)

The paper deals with the modeling of woodwind musical instruments using a Boundary Integral Equation (BIE) formulation. Specifically, the BIE is used to model the acoustic response of the instrument bore, and is numerically solved by means of a Boundary Element Method (BEM). The latter takes advantage of an analytical solution for the calculation of the BEM coefficients, thus allowing for the representation of the problem as an open-domain problem. This peculiarity avoids the use of approximated boundary conditions at the open end of the pipe. The formulation is used to: evaluate the input impedance of the resonating air column; identify the frequency dependent transfer function relating the inflow of the instrument with the signal measured at a specified location (the Reed-To-Microphone transfer function); extend the analysis to a realistic performing environment to obtain the "Reed-To-Listener" transfer function. Standard techniques are used to take into account the interaction of the bore with the nonlinear exciting device. Numerical results are obtained for single-reed instruments in terms of tuning properties, convergence of solution, directivity patterns, and simple synthesized sounds. Issues related to the possibility of real-time simulations are briefly addressed. Specifically, the identification of digital filters from the calculated transfer functions is investigated, and some preliminary numerical result included.

12:00

**4aMU12. An approach and technique for acoustic modelling of contact.** Matthias Rath (Technische Universität Berlin, Deutsche Telekom Laboratories, Quality and Usability Lab, Ernst-Reuter-Platz 7, 10587 Berlin, Germany, matthias.rath@telekom.de)

Vibration of solid objects can often be modelled by modal description, which exists when certain operators in a differential equation are linear. Discrete-time algorithms can represent the behaviour of "modal objects" without artefacts, *exact* (with precision of finite computer architecture), if based on analytical solution, *not* numeric approximation. In particular, the energy associated with the state of the system is then preserved and the algorithm stable. A scenario of two objects interacting only during contact, however, is non-linear: a non-zero linear function cannot be zero in a half-space of the system's state-space. Existing computational models are based on *approximate*, numeric solution and cannot guaranty stability in situations with longer contact phases, such as rolling or sliding. General principle of our approach is: although any possible term for the interaction force must be non-linear, it may be *piecewise-linear*, and during each linear phase the whole system (of interacting solids) may be simulated by modal description. The question of "switching" between the different phases is here critical but may be solved in a way that assures control over the system's energy. Our implemented new contact model is stable in any condition and overcomes artefacts found with previous techniques.



**Session 4aNSa****Noise and EURONOISE: EU Projects for Aircraft Noise Reduction II**

Dominique Collin, Chair

*Snecma - Safran Group, Centre de Villaroche, Direction R&T / UC, Moissy Cramayel, 77550, France****Invited Papers*****8:00**

**4aNSa1. European Aircraft Noise Research Network (X3-Noise).** Dominique Collin (Snecma - Safran Group, Centre de Villaroche, Direction R&T / UC, 77550 Moissy Cramayel, France, dominique.collin@snecma.fr)

The X3-NOISE Coordination Action, through its network structure and comprehensive workplan involving expert groups, scientific workshops, stakeholder seminars and a common information system, addresses the aircraft noise challenges set by the ACARE 2020 Vision. To this end, X3-Noise undertakes the elaboration and coordination of research strategies, the dissemination of results and the integration of European research activities in the field of air transport related to noise. Over 4 years, the project involves 32 partners from 20 countries (FR, UK, ES, NL, BE, DE, SE, IT, PL, HE, HU, CZ, LT, IE, PT, RO, CH, UA, EG, BR), combining the complementary skills and expertise of industry partners, SMEs, university and research establishments to cover the whole field of interest. The international co-operation aspects of the research agenda to be developed through the project activity are further reinforced by the participation of 3 partners from Ukraine, Egypt and Brazil acting as Focal Points at Regional level.

**8:20**

**4aNSa2. The EU FP6 research project PROBAND - Objectives and first results.** Lars Enghardt (DLR - German Aerospace Center, Mueller-Breslau-Str. 8, 10623 Berlin, Germany, lars.enghardt@dlr.de)

Fan broadband noise is a major aircraft noise challenge now and in the future will be even more important. Novel low-noise engine architectures, such as ultra-high-bypass-ratio engines and lower-speed fans, can help address jet noise and fan tone noise, but previous EC-funded programmes have shown they are unlikely to reduce significantly fan broadband noise without improved understanding of the source mechanisms. PROBAND will accomplish a major technical leap in providing industry with an improved understanding of the broadband noise source mechanisms, with validated broadband noise prediction methods, and with low fan broadband noise concepts. PROBAND will exploit the noise technology and methodology acquired in EC-funded projects and national programmes, to develop methods to allow the design of a fan system that will generate sufficiently low broadband noise to meet the EU noise level targets. This will be achieved by: 1. Developing a better understanding of broadband noise generation mechanisms using advanced experimental and computational techniques. 2. Developing and validating improved prediction methods using conventional computational fluid dynamics, and integrating them into industrial codes. 3. Exploring new prediction strategies using advanced computational techniques 4. Developing low broadband fan noise concepts.

**8:40**

**4aNSa3. SILENCE(R): A major step towards aircraft noise reduction.** Eugène J. Kors (Snecma, Etablissement de Villaroche Sud - UE, Rond-point René Ravaud - Réau, 77550 Moissy-Cramayel cedex, France, eugene.kors@snecma.fr)

From 2001 up to 2007, SILENCE(R) has focused on the development of aircraft noise reduction technologies. As an European Union project coordinated by Snecma, SILENCE(R) brought together some 50 companies (including Airbus, Rolls-Royce, MTU Aero Engines and Snecma), research centers and universities. The overall budget was 112 million euros. Combined with innovative low-noise operational procedures studied at the same time, SILENCE(R) has achieved an impressive 5 dB noise reduction. This meets the medium-term objective of the European Commission's R&D Framework Programs, and marks a significant advance towards ACARE's research goal of a 10 dB reduction in aircraft noise by 2020. SILENCE(R) carried out successful tests of more than 35 prototypes to check 10 noise reduction technology concepts. These included several advanced low noise fan rotors, as well as components for a complete low-noise nacelle (negatively scarfed intake, "squid" nozzle fitted with high frequency liner), flight tested on an Airbus A320. Flight tests were also carried out on an Airbus A340 with landing gear fitted with aerodynamic fairings.

9:00

**4aNSa4. TIMPAN - Technologies to IMProve Airframe Noise.** Stephane Perrin Decroux (Airbus France S.A.S, 316, route de Bayonne BP M0112/4, 31060 Toulouse, France, stephane.perrin-decroux@airbus.com)

TIMPAN is a 3-year Strategic Targeted Research Project of the 6th European Framework Programme, Priority "Aeronautics and Space", launched in 2006. It addresses the community noise reduction objective for commercial aircraft by focusing on main airframe noise sources: landing gear and high-lift devices - responsible, on recent aircraft, for about half of the total noise in approach situation. TIMPAN addresses both sources, within 3 main tasks: 1) Landing gear activity: investigation of both innovative low noise technologies on bluff body structures and the improvement of advanced low noise main gear design, as outcome from previous EC Technology Platform SILENCE(R). 2) High lift device activity: study of both innovative concepts based on flow control technologies and mid-term noise reduction solutions as absorptive wing leading edge treatments and high-lift settings optimization through computational aero-acoustic methods. 3) Technology evaluation: aims to prepare the exploitation phase by evaluating the noise reduction technologies under consideration in terms of noise benefit, integration, cost and performance. TIMPAN brings together 14 actors from the European aeronautics industry including aircraft manufacturers (Airbus and Dassault Aviation), landing gear manufacturers (Messier-Dowty and Messier Bugatti), key research institutes (DLR, ONERA, NLR, EADS-IW), universities (University of Southampton, Technical University of Braunschweig) and SMEs (ATECA, Free Field Technologies).

9:20

**4aNSa5. Passive and active designs for noise and vibrations reduction in aircraft cabins.** Vincent Marant (Acusttel - Acústica y Telecomunicaciones SL, Pol. Ind. Benieto, C/ Transport n° 12, 46702 Gandía (Valencia), Spain, vmarant@acusttel.com), Antonio Reig Fabado (Universidad Politécnica de Valencia, Camino de Vera s/n, 46022 Valencia, Spain, areig@fis.upv.es), Juan Luís Aguilera De Maya (Acusttel - Acústica y Telecomunicaciones SL, Pol. Ind. Benieto, C/ Transport n° 12, 46702 Gandía (Valencia), Spain, jlaguilera@acusttel.com), José Christian Donayre Ramírez (Acusttel - Acústica y Telecomunicaciones SL, Pol. Ind. Benieto, C/ Transport n° 12, 46702 Gandía (Valencia), Spain, cdonayre@acusttel.com)

Within the sixth European Commission framework programme, the main objective of the SEAT project initiated in September, 2006, consists of the development of a radically new concept, where the aircraft passenger comfort is considered at the highest level. Smart reactive seats and an interior environment able to detect on real time physiological and psychological changes in the passenger conditions will be developed. These data will be analysed and the appropriate parameters, like noise and vibration levels, temperature or air ventilation, will be adapted. Moreover, each passenger will be able to create his own configuration, with his personal entertainment and work characteristics. The project is focussed on the questions previous to the integration of the system, that is above all the creation of a more healthy and comfortable travel environment by means of noise and vibration reduction, as well as specific climatic controls. In this paper, the first passive and active designs under development are presented.

### Invited Paper

9:40

**4aNSa6. Reducing aircraft noise during approach and departure by efficient operations.** Robert J. De Muynck (National Aerospace Laboratory NLR, Anthony Fokkerweg 2, 1059CM Amsterdam, Netherlands, demuynck@nlr.nl)

Reducing the noise from aircraft around airports is a serious challenge. Apart from making the noise source, the aircraft, more silent, by advanced engines and aerodynamics, additional noise reduction can be achieved by moving the noise source away, as well as by a more silent operation of the aircraft. One of these innovative approach procedures is the Continuous Descent Approach. The principle is that aircraft approaching an airport follow a continuous descent profile at low thrust setting instead of making gradual altitude steps. The higher altitude and the lower thrust of the aircraft allow decreasing significantly noise exposure around the airport. First results show considerable noise reduction from 3dB to 8dB compared to conventional practices. Additionally, this procedure allows reducing fuel consumption and emissions. To make CDA procedures operationally feasible however, efforts are needed to develop \*CDA operating procedures so that they can be flown in busy traffic \*improved onboard systems to fly the CDA \*accurate planning and sequencing tools for air traffic controllers \*better interaction and interoperability between aircraft and air traffic control systems The presentation will discuss progress made in EU projects such as Sourdine II, OPTIMAL and ERAT.

**Session 4aNSb****Noise, ASA Committee on Standards, and EURONOISE: Measurement of Occupational Noise Exposure I**

William J. Murphy, Cochair

*National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Mailstop C-27, Cincinnati, OH 45226-1998, USA*

Beat W. Hohmann, Cochair

*Suva, Physics Section, Roesslimattstrasse 39, P.O. Box 4358, Luzern, CH-6002, Switzerland***Invited Paper****8:00****4aNSb1. Measurement strategies according to the new ISO/DIS 9612.** Jürgen Maue (BGIA - Institut für Occupational Safety and Health, Alte Heerstraße 111, D-53757 Sankt Augustin, Germany, juergen.maue@dguv.de)

The European Directive 2003/10/EC on noise protection at the workplace has given the impact for the revision of the ISO 9612 describing the measurement and assessment of occupational noise. The ISO/TC 43 "Acoustics" decided that the revision should only contain requirements for the determination of the noise exposure for the purpose of assessing potential hearing damage. Meanwhile a new Draft ISO/DIS 9612 has been presented, which gives a detailed description of three alternative measurement strategies for the determination of the noise exposure level. Moreover the standard provides a method for estimating the uncertainty of the results. The measuring strategies, called "task based measurements", "job based measurements" and "full day measurements" are illustrated and compared to each other. The choice of the appropriate strategy is explained, depending on the complexity of the work situation and the movements of the worker doing his job. Based on practical experience some comments are given, which may help to reduce time and effort of measurements. The evaluation of measurement uncertainties is explained in a few words.

**Contributed Paper****8:20****4aNSb2. Effective protection of the sense of hearing is prevented by ISO1999 noise standard.** Gerald Fleischer (Liebig University School of Medicine, Aulweg 123, 35392 Giessen, Germany, gerald.fleischer@gmx.net)

To determine auditory performance in a simple way, pure-tone audiometry is being used. While health-related diagnostics is generally becoming more and more refined, ISO1999 is demanding rather rudimentary procedures. Both, frequency range and number of test frequencies are re-

duced, making it impossible to use modern and effective analytical tools such as pattern recognition. There is a clear relation between pressure-time-history of impulses and the details of the resulting auditory damage. But this relation can only be recognized if the restrictions of this standard are being ignored. Typical examples of audiograms from a large data base will be presented, showing simultaneously the data according to ISO1999 and in more modern ways. It will also be demonstrated how useful the tool of pattern recognition can be, for analysis of damages, as well as for preventive measures.

**Invited Paper****8:40****4aNSb3. Comparison of impulse noise damage risk criteria using the Albuquerque blast overpressure walkup study data.** William J. Murphy (National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Mailstop C-27, Cincinnati, OH 45226-1998, USA, wjm4@cdc.gov), Amir Khan (National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Mailstop C-27, Cincinnati, OH 45226-1998, USA, amk1@cdc.gov), Peter B. Shaw (National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Mailstop C-27, Cincinnati, OH 45226-1998, USA, pbs3@cdc.gov)

The 1968 CHABA recommendations to limit impulsive noise exposure to levels below 140 dB SPL form the basis of current United States occupational and military standards. The U.S. military standard, MIL-STD-1474D, estimates the number of allowable shots to which a person may be exposed using peak level, B-duration, for varying levels of hearing protection usage. The European Union upper limit peak exposure action level is 137 dB C-weighted for the unprotected ear and 140 dB C-weighted for the protected ear. The U.S. Army blast overpressure studies in the early 1990's investigated the effects on the hearing of soldier subjects of simulated weapon blasts with varied levels and A-durations. The hearing thresholds of the subjects were tested before and after exposure to blasts. Exposures ranged from 6 shots to 100 shots per day and levels from 173 to 195 dB peak SPL. As judged by information criteria (AIC, BIC), the  $L_{Aeq8}$  index with unprotected data yielded the best fit to logistic models; all the indices produced a better fit with unprotected data than with protected data. Other metrics including the MIL-STD-1474D and the Auditory Hazard Assessment Algorithm for Human were evaluated and will be presented in this paper.

9:00

**4aNSb4. Reconstruction of acoustic field horizontal layer in diffusive ambience.** Arunas Pozera (Kaunas University of Technology/Technological Systems Diagnostics Institute, Kestucio st. 27, 3004-LT Kaunas, Lithuania, arunas.pozera@stud.ktu.lt), Vitalijus Volkovas (Kaunas University of Technology, K.Donelaicio str. 73, LT-44029 Kaunas, Lithuania, tsdi@ktu.lt)

This study introduces a method how to minimize the number of acoustic measurements in the diffusive ambience in order to reconstruct the horizontal layer of primary acoustic field. This reconstruction means determination of the level of the sound pressure at any point of surface, e.g. worker's occupational place, where there are a lot of sound sources. A mathematical modeling, based on regularity of sound spread, is used in the study. The elements of active experiment are applied in the research, the adequacy of mathematical model is evaluated by Fisher criterion, and the mathematical model of horizontal layer of the acoustic field is based on the polynomial of the regressive equation. The results are compared to the measurements received in the real experiment. This study also pays great attention to local and the EU legal regulations about the noise standards at the worker's occupational place and the measures to be taken to lower the noise level. The device of sound measurement INVESTIGATOR type 2260 and MAPLE 11 mathematics software tool have been used in the research.

9:20

**4aNSb5. An experimental investigation of the measurement uncertainties in the assessment of exposure to noise in a working**

**environment according to ISO 9612.** Marco Nabuco(Inmetro / CNPq, Av. N. S. das Graças, 50, Xerém, Duque de Caxias, 25250-020 RJ, Brazil, nabuco@inmetro.gov.br), Ana Claudia Fiorini (Catholic University of São Paulo - Audiology Dept., Rua Ministro Godoy, 969 - Perdizes, 05014-001 São Paulo, Brazil, acfiorini@pucsp.br), Gilberto Fuchs (GROM Acustica e Automação, Rua Indiana, 343 - Apt° 11 - Broklyn, 04562-000 São Paulo, Brazil, gilberto@grom.com.br)

The Brazilian regulation for workplace noise evaluation requires a comparison between the measured noise levels and the noise limits listed on a table. The table establishes the maximum work hours per day for each measured sound level. As in some other countries, the Brazilian table shows limit values with a resolution of 1 dB(A) and considers a 5 dB(A) exchange rate. The measured uncertainty is unknown. The use of type 2 equipment for instance can mean, theoretically, a uncertainty of 3 dB(A). In Brazil it is not possible to estimate the impact of the measurement uncertainty on the costs of compensations paid due to noise exposure. It is also very difficult to assess the number of workers that are receiving unfair compensations or are not receiving any due to inaccurate noise exposition measurements. However it is possible to assure that the amount of money can reach incredible numbers in our country. This paper presents an experimental study of the uncertainty calculated on the ISO 9612 basis for different sound signals, in different workplaces, emitted by different noise sources. The main purpose is to show how important it can be to estimate the uncertainty for each measurement.

## Invited Paper

9:40

**4aNSb6. Simple evaluation of occupational noise exposure without measurements.** Beat W. Hohmann (Suva, Physics Section, Roesslimattstrasse 39, P.O. Box 4358, CH-6002 Luzern, Switzerland, beat.hohmann@suva.ch)

In almost any country, employers must assess the noise exposure of the employees. But for small and medium-sized enterprises (SMEs), individual noise measurements are hardly feasible. Therefore, since almost 30 years, Suva publishes noise level tables, which list typical noise levels of many tools and machines. These 66 noise level tables cover almost any branch of the industry. Moreover, they include shooting noise, music, and so on. However, most SMEs were not able to calculate their workers' long-term noise exposure. Therefore, based on Suva's extensive database, typical noise exposure levels for occupations and activities were calculated. The new type of noise level tables includes now the typical noise exposures for the different occupations. Moreover, since 2007, the noise level tables also indicate the measures to be taken. Therefore, SMEs do not have to put their efforts into the determination of noise exposure but can start immediately to protect the workers against noise-induced hearing loss. The list of the noise level tables available (in French, German and Italian language) and the noise level tables themselves may be found at: [www.suva.ch/waswo/86005](http://www.suva.ch/waswo/86005)

## Contributed Papers

10:00

**4aNSb7. A comparison of two active-speaker-detection methods suitable for usage in noise dosimeter measurements.** Fredric Lindstrom (Dept. of Environ. Medicine, The Sahlgrenska Acad. of Gothenburg Univ., Box 414, 405 30 Gothenburg, Sweden, fredric.lindstrom@amm.gu.se), Keni Ren (Umea University, Dept. of Applied Physics and Electronics, Teknikhuset, Box 414, 901 87 Umea, Sweden, renkeni@gmail.com), Kerstin Persson Waye (Dept. of Environ. Medicine, The Sahlgrenska Acad. of Gothenburg Univ., Box 414, 405 30 Gothenburg, Sweden, kerstin.persson-waye@amm.gu.se), Haibo Li (Umea University, Dept. of Applied Physics and Electronics, Teknikhuset, Box 414, 901 87 Umea, Sweden, haibo.li@tfe.umu.se)

Measuring noise exposure in a working environment is often done by using standard noise dosimeters. This method is suitable for the evaluation of many working environments. However, in some situations the worker uses his/her voice a large amount during the day, e.g. teachers in a preschool environment. Thus, in these situations regular dosimeter measurements will not correspond to the actual noise exposure. In order to provide correct measurements, methods that can detect when the workers own voice is active are required. This paper presents a study of two such methods originating from voice research; the binaural and the throat microphone methods. The meth-

ods are compared using a receiver operating characteristics based method, where the performance is assessed by the Probability-of-failure measure, i.e. the percentage of own voice that the method failures to detect correctly. The evaluation is performed in a lab environment as well as in real field conditions in a preschool. The results of the study show that both methods can be successful in a controlled low noise (<45dBA) environment (Probability-of-failure <0.1% for both methods), while in the preschool environment, the throat microphone method (Probability-of-failure <0.1%) is more suitable than the binaural method (Probability-of-failure >6%).

10:20-10:40 Break

10:40

**4aNSb8. Recovery of distortion product otoacoustic emissions (DPOAE) after impulse vs. continuous equal-energy exposures.** Miguel Angel Aranda De Toro (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark, maat@es.aau.dk), Rodrigo Ordoñez (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark, rop@es.aau.dk), Karen Reuter (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark,



The correct assessment of impulse noise from occupational environments for hearing-conservation purposes is still a controversial issue. Currently, no universally accepted standard defines impulse noise accurately nor does a standard method exist to measure impulses. Moreover, current impulse-damage risk-criteria suffer from lack of empirical data needed to quantify impulse noise exposures and assess potential damage. In this experiment human subjects are exposed to binaural recordings of noises from industrial environments. Stimuli consist of impulse noise, continuous noise,

and combinations of impulse and continuous noise. Noise exposures are normalized to have the same energy ( $L_{Aeq,8h} = 80\text{dB}$ ). The effects in the hearing of the subjects are monitored by measuring the recovery of the distortion product otoacoustic emissions (DPOAE) with high-time resolution. The results can be used to investigate the validity of current assessment methods and descriptors of the temporal characteristics of sound exposures and their relation to the temporal effects produced on the human hearing as well as investigating selected issues that may lead to possible improvements or alternative measuring methods. [Work supported by the Danish Research Council for Technology and Production.]

### *Invited Paper*

11:00

**4aNSb9. "Real world" noise exposure beneath hearing protectors : a scattered international practise.** Pierre Canetto (INRS - Institut National de Recherche et de Sécurité, Avenue de Bourgogne BP No 27, 54501 Vandoeuvre, France, pierre.canetto@inrs.fr), Nicolas Trompette (INRS - Institut National de Recherche et de Sécurité, Avenue de Bourgogne BP No 27, 54501 Vandoeuvre, France, nicolas.trompette@inrs.fr)

Assessing occupational noise exposure "beneath" Hearing Protection Devices (HPD) is a topical subject. Standardized methods allow to calculate the exposure by using the HPD attenuation. The HPD attenuations declared by manufacturers are much higher from the "real-world" ones. To reduce this difference, some "compensation" methods are proposed. The methods and their rules vary a lot from one country to another. The "derating" method reduces the declared values from a certain amount, according to the HPD kind. "Double-labelling" method uses attenuation values measured in laboratory with untrained subjects. "Statistical range enlargement" widens the statistical confidence of the laboratory-test results. All these methods propose a global solution: the reasons due to human behaviour (mainly bad HPD wearing), products quality, and the difference between laboratory and industrial situations are mixed. This "blind" approach could be considered as endorsing the lack of workers' training, and could impede the progress in HPD developments. But a "short-term" answer to the problem is needed. It should allow to build a more relevant medium-term answer, which could be worked out possibly on an international scale.

### *Contributed Papers*

11:20

**4aNSb10. A comparison of earmuff protection measured in real-world and laboratory conditions.** Emil Kozłowski (Centr. Inst. for Labour Prot. - Natl. Res. Inst., Czerniakowska 16, 00-701 Warsaw, Poland, emkoz@ciop.pl), Ewa Kotarbinska (Warsaw Univ. of Technology Inst. of Radioelectronics, Nowowiejska 15/19, 00-665 Warsaw, Poland, Ewa.Kotarbinska@ire.pw.edu.pl)

It has been well known and that noise protection provided by earmuffs in real-world conditions is lower than measured by a laboratory standardized REAT test. In this study, earmuff protection was tested by simultaneous measurement of the  $L_{Aeq}$  under and outside the earmuffs in 91 industrial workplaces and in the laboratory. In all cases, the  $L_{Aeq}$  measured under the earmuffs was compared with the level predicted according to the EN 458 standard, by an octave-band method for calculating the A-weighted SPL under the hearing protector. The  $L_{Aeq}$  levels measured under the earmuffs in real-world conditions were by more than 3 dB(A) and 15 dB(A) higher than predicted by the octave-band method, respectively in 65% and 17% of cases. The main causes of worse protection at workplaces were worn-out earmuffs due to prolonged usage (33%), improper way of wearing earmuffs (15%), or the use of eye-glasses (8%). The data show that attenuation values measured by the REAT method overestimate by 1-5 dB the earmuff protection obtained in real-world conditions. [Work supported by the Polish Ministry of Labour and Social Policy, grants 4.S.03 and 3.S.02].

11:40

**4aNSb11. Empirical evaluation using impulse noise of the level-dependency of various passive earplug designs.** Elliott H. Berger (E-A-R / Aeero Technologies, 7911 Zionsville Rd., Indianapolis, IN 46268-1657, USA, eberger@compuserve.com), Pascal Hamery (French German Institut of Saint Louis (ISL), 5 rue du Général Cassagnou, 68301 Saint-Louis, France, hamery@isl.tm.fr)

An objective in the development of hearing protection devices (HPDs) has been the design of a passive earplug that provides modest or no attenuation at low sound levels, with greater protection at high sound levels. This raises the issue of not only how to construct such a device, but also how to evaluate it. There is the related question of whether conventional HPDs are actually level independent. Passive level dependency is typically accomplished via an orifice that causes sound transmission to decrease as input level increases. We utilized an impulsive noise source (explosives) with peak levels from 110 to 190 dB SPL to measure the insertion loss of a variety of commercially available and developmental earplugs. The tests were conducted at frontal incidence in a reflection-free outdoor environment using the Institute of Saint-Louis acoustical test fixture specifically constructed for HPD attenuation measurements. Conventional foam and premolded earplugs exhibited attenuation that was essentially constant with level, whereas the best of the level-dependent designs provided attenuation that increased by about 25 dB over the 80-dB range of test impulse levels. This latter design has been successfully utilized since 2000 in the Combat Arms™ Plug widely fielded in the U. S. Military.

12:00

**4aNSb12. Impulse and continuous noise reduction of tactical hearing protection systems.** Richard McKinley (AFRL, Wright-Patterson Air Force Base, Dayton, OH 45433-7901, USA, richard.mckinley@wpafb.af.mil), Brian Hobbs (AFRL, Wright-Patterson Air Force Base, Dayton, OH 45433-7901, USA, brian.hobbs@wpafb.af.mil), Karl Buck (French German Institut of Saint Louis (ISL), 5 rue du Général Cassagnou, 68301 Saint-Louis, France, karl.buck@isl.eu), Dean Hudson (AFRL, Wright-Patterson Air Force Base, Dayton, OH 45433-7901, USA, harvey.hudson@wpafb.af.mil)

Tactical hearing protectors are devices designed to protect the wearer from high levels of impulse noise while providing some ambient listening and communication capability. Many of these devices also provide some attenuation of continuous noise. The efficacy of these types of devices depends on the amount of impulse noise protection and continuous noise protection, the quality of the ambient listening capability, and the intelligibility of the speech communication capability. This paper will describe the peak noise reduction of approximately 1 ms duration impulses with peak levels at 165 dB and 195 dB, for several earplugs and earmuffs. The frequency dependent reduction of impulse noise measured using an acoustic test fixture (the French German Research Institute at Saint Louis Head). Generally the data show higher levels of impulse noise reduction with earplugs than with earmuffs. Additionally, continuous noise attenuation was measured using human subjects performing the ANSI S12.6 Real Ear At Threshold (REAT) test. High speed (20k frames/sec) video will show the dynamic motion of typical tactical hearing protection earplugs and earmuffs when stimulated with a 195 dB impulse.

### Contributed Papers

12:20

**4aNSb13. Sound path(s) to the ear protected by double hearing protection.** Karl Buck (French German Institut of Saint Louis (ISL), 5 rue du Général Cassagnou, 68301 Saint-Louis, France, karl.buck@isl.eu), Pascal Hamery (French German Institut of Saint Louis (ISL), 5 rue du Général Cassagnou, 68301 Saint-Louis, France, hamery@isl.tm.fr), Richard McKinley (AFRL, Wright-Patterson Air Force Base, Dayton, OH 45433-7901, USA, richard.mckinley@wpafb.af.mil)

Studies have shown that the attenuation performance of double hearing protection is limited by bone or tissue conduction. We present the first results of a program to determine the different paths of the sound to the ear when a person is protected by double hearing protection. In order to determine the contribution of the noise signal passing by secondary sound paths (bone or tissue conduction) into the volume of the ear canal behind the earplug, identical tests have been performed on humans and artificial heads. Time delay and attenuation through the hearing protection (NR) were measured on the ipsi- and contra-lateral side of the head. The measurements show differences in time delay and attenuation between artificial heads and humans which indicate that the main transmission mechanism is different. They also give a strong indication that the path of the acoustic signal depends on the angle of arrival when using double HP. This may partly explain the poor sound localization performance when using double HP that was reported by Brungart (2002). The single HP, and the double HP were then used

in a sound localization experiment. The localization performance of these conditions will be presented along with the related parameters.

12:40

**4aNSb14. Prediction of impulsive noise waveform under an earmuff worn by a real user.** Rafal Mlynski (Centr. Inst. for Labour Prot. - Natl. Res. Inst., Czerniakowska 16, 00-701 Warsaw, Poland, rmlynski@ciop.pl), Jan Zera (Centr. Inst. for Labour Prot. - Natl. Res. Inst., Czerniakowska 16, 00-701 Warsaw, Poland, jazer@ciop.pl)

For the assessment of hearing damage risk caused by impulsive noise, it is important to know the impulse waveform a user is exposed to under the hearing protector. In this study, a complex transmittance of an earmuff was used to predict the waveform under a hearing protector. Earmuff's transmittance was calculated from impulses recorded outside and under the hearing protector, for a real user, as transmittance of the equivalent FIR filter. The transmittance determined in that way was then used to predict the impulse waveforms under the earmuffs produced in response to various outside impulses. Accuracy of predictions was assessed by a comparison of peak SPL,  $L_{Aeq}$ , A, C, or D duration of the impulse waveforms calculated and measured under the earmuffs. Results obtained for a real user were compared with the measurements made with the use of an artificial test fixture (ATF, ISO 4869-3). [Work supported by Polish grants R18004304 (MNiSzW) and 3.S.03 (MPiPS)].

## Session 4aNSc

## Noise, Computational Acoustics, and EURONOISE: Time-Domain Modeling Methods in Acoustics II

Paul Calamia, Cochair

*Rensselaer Polytechnic Institute, Greene Bldg., 110 8th St., Troy, NY 12180, USA*

Maarten Hornikx, Cochair

*Applied Acoustics, Chalmers University of Technology, Sven Hultins Gata 8a, Gothenburg, SE-41296, Sweden**Invited Papers*

8:00

**4aNSc1. Time domain modeling of acoustic propagation with acoustic wave propagator and absorbing boundary conditions.**

Jan H. Ehrlich (FWG Underwater Acoustics and Marine Geophysics Research Institute, Klausdorfer Weg 2-24, 24148 Kiel, Germany, janehrlich@bwb.org)

The acoustic wave propagator (AWP) is the application of the time evolution operator on the acoustic wave equation for stationary systems in a polynomial expansion of Chebychev polynomials. It allows to increase the time step by more than one order of magnitude compared to finite difference time domain (FDTD) codes. In contrast to other implementations of the AWP the spatial differentiation is carried out with finite difference techniques because this allows the use of the perfectly matched layer formulation as absorbing boundary conditions. The formulation includes the direct implementation of acoustic sources with sinusoidal time evolution. Other sources can be synthesized by their Fourier components. For the calculation of large areas the explicit formulation of a large system matrix can be avoided by calculating the propagation equations for each time step at row and column level repeatedly which reduces memory requirements notably. This procedure and the suitability of the finite difference approach for parallelization makes the extension to fully three dimensional calculations possible. Examples for benchmark problems with sound propagation in air and water are given.

8:20

**4aNSc2. A time domain boundary element method for compliant surfaces.** Jonathan A. Hargreaves (University of Salford, M5 4WT Manchester, UK, j.a.hargreaves@salford.ac.uk), Trevor J. Cox (University of Salford, Acoustics Research Centre, Newton Building, M5 4WT Salford, UK, t.j.cox@salford.ac.uk)

The best way of representing compliant surfaces in time domain prediction models, such as the transient Boundary Element Method (BEM), is currently unresolved. This is not true of frequency-domain, time-invariant models, where the common practice is to represent the characteristics of a material by its surface impedance. A BEM may be used to predict the scattering of sound, and reduces the problem of modelling a volume of air to one involving surfaces conformal to the obstacles. Surface impedance is a convenient concept for inclusion in the frequency domain BEM as it abstracts the obstacle's characteristics into a property of the conformal surface. The time domain BEM predicts transient scattering of sound, and is usually solved in an iterative manner by marching on in time from known initial conditions. For surface impedance data to be utilised it must be Fourier transformed from a frequency dependent multiplication into a temporal convolution. This approach typically yields convolution kernels which involve future sound, hence is not compatible with time-marching solvers. In this paper an alternative time domain representation of compliant locally reacting materials is proposed to overcome this problem, and its implementation and limitations discussed.

8:40

**4aNSc3. Stability behaviour and results of a transient boundary element method for exterior radiation problems.** Michael Stütz (Technische Fachhochschule Berlin, Univ. of Applied Sciences, Luxemburger Str. 10, 13353 Berlin, Germany, stuetz@tfh-berlin.de), Martin Ochmann (Technische Fachhochschule Berlin, Univ. of Applied Sciences, Luxemburger Str. 10, 13353 Berlin, Germany, ochmann@tfh-berlin.de)

Based on the Helmholtz integral equation, a boundary element method in time domain (TD-BEM) can be formulated. Because of instability issues, this method is rarely used in numerical acoustics. The stability and accuracy of the method for exterior radiation problems is investigated using some simple examples. A connection between the internal resonances of the closed structure and the instable behaviour of the method is assumed, but it is mathematically unproven. Numerical evidence of this connection is presented. Because of the sparse structure of the resulting system matrix, the use of iterative solvers is advantageous. The performances of different solvers are compared with respect to stability and numerical costs. For testing purposes, the acoustic radiation from an open turbulent flame is calculated and compared with results of a frequency domain BEM calculation.

9:00

**4aNSc4. Analytical validation of time-step interpolation in transient insular nodal analysis.** Tom De Rybel (The University of British Columbia, Department of Electrical and Computer Engineering, 2332 Main Mall, Vancouver, BC V6T 1Z4, Canada, tomr@ece.ubc.ca), José Martí (The University of British Columbia, Department of Electrical and Computer Engineering, 2332 Main Mall, Vancouver, BC V6T 1Z4, Canada, jrms@ece.ubc.ca), 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)

The transient insular nodal analysis method (TINA) combines elements from the finite differences (FD) and transmission-matrix methods (TLM) in one unified approach. In contrast to existing TLM-FD methods, TINA uses time-decoupled cells, avoiding the need for solving large system matrices. The time-decoupled cells allow for easy parallelisation, and the solution of large systems in detail. Due to the use of an exact transmission-line model in the cells, wave propagation can be computed without the need for discretisation of the equations, nor the use of prediction, yielding an unconditionally stable method. Boundary conditions are implicit, and are solely defined by the wave speed and characteristic impedance of the medium. One key difference with the TLM method is how cells whose transmission time is not an integer multiple of the simulation time step are integrated in the simulation. These mismatches occur due to the varying wave speeds of the different media in the cells. In TINA, the match is obtained through interpolation, as opposed to the stub-matching methodology employed in TLM. In this paper, we will demonstrate the validity of the interpolation approach analytically, as well as compare the TINA method to a theoretical case.

9:20

**4aNSc5. An eigenfunction expansion method to efficiently evaluate spatial derivatives for media with discontinuous properties.** Maarten Hornikx (Applied Acoustics, Chalmers University of Technology, Sven Hultins Gata 8a, SE-41296 Gothenburg, Sweden, maarten.hornikx@chalmers.se), Roger Waxler (University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677, USA, rwax@olemiss.edu)

Pseudo-Spectral methods are often used as an alternative to the Finite Difference Time Domain (FDTD) method to model wave propagation in heterogeneous moving media. The FDTD method is robust and accurate but is numerically expensive. Pseudo-Spectral methods make use of the wavelike nature of the solution to obtain more efficient time-domain algorithms. The most straightforward of the Pseudo-Spectral methods is the Fourier method in which a spatial Fourier transform is used to evaluate the spatial derivatives in the wave equation. Whereas this method is accurate for a weakly heterogeneous moving medium, it degenerates for media with discontinuous properties. The eigenfunction expansion method presented here is a way to accurately and efficiently evaluate spatial derivatives in media with interfaces. As in the Fourier method, transforms may be calculated using FFT's and spatial sampling is limited only by the Nyquist condition. The performance of the method is shown in a time-domain implementation for media with discontinuous density and sound speed.

9:40

**4aNSc6. Optimal tree canopy shape for improving downwind noise barrier efficiency.** Timothy Van Renterghem (University Ghent - Department Information Technology, Sint-Pietersnieuwstraat 41, 9000 Gent, Belgium, Timothy.Van.Renterghem@intec.Ugent.be), Dick Botteldooren (University Ghent - Department Information Technology, Sint-Pietersnieuwstraat 41, 9000 Gent, Belgium, dick.botteldooren@intec.ugent.be)

The presence of a row of trees behind a highway noise barrier significantly reduces the screen-induced refraction of sound by wind. In this paper, the influence of quantitative tree properties, such as the pressure resistance coefficient of the canopy and the distribution of biomass over height, was studied numerically. Three computational models were involved. First, computational fluid dynamics (CFD) software is used to accurately predict the wind fields. The finite-difference time-domain (FDTD) method is then used to simulate sound propagation in the direct vicinity of the noise barrier in combination with trees. In a last step, the Parabolic Equation (PE) method is used to predict sound fields at larger distances. As a general conclusion, it was found that coniferous trees are superior to deciduous trees to improve downwind noise barrier efficiency.

10:00

**4aNSc7. Green roofs to enhance quiet sides.** Timothy Van Renterghem (University Ghent - Department Information Technology, Sint-Pietersnieuwstraat 41, 9000 Gent, Belgium, Timothy.Van.Renterghem@intec.Ugent.be), Dick Botteldooren (University Ghent - Department Information Technology, Sint-Pietersnieuwstraat 41, 9000 Gent, Belgium, dick.botteldooren@intec.ugent.be)

In this paper, the finite-difference time-domain method is used to study sound propagation over a green roof in an urban situation. Sound propagation between adjacent city canyons is considered, and the focus is on the reduction of the sound pressure level in the non-exposed canyon due to the presence of a vegetated (green) roof. Numerical calculations have been conducted for both intensive and extensive green roofs, showing that an important reduction of the sound pressure level in the shielded canyon can be achieved, compared to a rigid roof. In case of an extensive green roof, there is a strong dependence on the substrate layer thickness; a maximum reduction of 10 dB at the octave band of 1000 Hz was found. For intensive green roofs, the influence of the substrate layer thickness is limited.

4a THU. AM



## Session 4aNSd

## Noise and EURONOISE: Car Acoustics II

Luc Mongeau, Cochair

*McGill University, 817 Sherbrooke St. West, Montreal, QC H3A 2K6, Canada*

Virginie Maillard, Cochair

*RENAULT, Groupe Acoustique, Technocentre, 1 avenue du Golf, 78288 Guyancourt Cedex, France*

## Invited Paper

8:00

**4aNSd1. Trim FEM simulation of a dash and floor insulator cut out modules with structureborne and airborne excitations.**

Arnaud Duval (Faurecia AST, Center of Acoustic Technology, Dämmstoffwerk 100, 38524 Sassenburg, Germany, arnaud.duval@faurecia.com), Julien Baratier (Faurecia AST, Center of Acoustic Technology, Dämmstoffwerk 100, 38524 Sassenburg, Germany, julien.baratier@faurecia.com), Christian Morgenstern (Faurecia AST, Center of Acoustic Technology, Dämmstoffwerk 100, 38524 Sassenburg, Germany, christian.morgenstern@faurecia.com), Ludovic Dejaeger (Faurecia AST, Center of Acoustic Technology, Dämmstoffwerk 100, 38524 Sassenburg, Germany, ludovic.dejaeger@faurecia.com), Norimasa Kobayashi (Toyota Motor Corporation, 1, Toyota-cho, 471-8572 Toyota, Aichi, Japan, koba@nori.tec.toyota.co.jp), Hiroo Yamaoka (Toyota Motor Corporation, 1, Toyota-cho, 471-8572 Toyota, Aichi, Japan, yamaoka@giga.tec.toyota.co.jp)

During a vehicle development, measurements on cut out modules in large coupled reverberant rooms are often carried out in the middle and high frequency range in order to optimize the insulation performance of the trims (Transmission Loss). Using optimal controlled mounting conditions, we have been able to extend the frequency range to the low frequencies in order to validate trim FEM models of a dash and floor insulator modules with structureborne and airborne excitations. Both coupled response with movable concrete cavities (structureborne excitation) and Transmission Loss with the coupled reverberant rooms (airborne excitation) have been measured and simulated for various types of insulators on the same setup, without any change on the mounting conditions. An additional movable absorbing environment in the large reception room has been deployed in order to carry out laser vibrometer (skeleton velocity) and p-u probes (particle velocity and intensity) measurements on the surface of the trims. By incorporating the maximal treatment mock-ups of the cut out modules as additional trims in the models, we have obtained good correlation results between measurements and simulations for both bare and trimmed configurations for a dash and floor insulators with structureborne and airborne excitations.

## Contributed Papers

8:20

**4aNSd2. Practical aspects of implementing car interior active noise control systems.** Rolf Schirmacher (Müller-BBM, Robert-Koch-Straße 11, 82110 Planegg, Germany, Rolf.Schirmacher@MuellerBBM.de), Roland Lippold (Müller-BBM, Robert-Koch-Straße 11, 82110 Planegg, Germany, Roland.Lippold@MuellerBBM.de), Frank Steinbach (Müller-BBM, Robert-Koch-Straße 11, 82110 Planegg, Germany, Frank.Steinbach@MuellerBBM.de), Florian Walter (Müller-BBM, Robert-Koch-Straße 11, 82110 Planegg, Germany, Florian.Walter@MuellerBBM.de)

When implementing real-world, close-to-production active noise control (ANC) systems for car interiors, many aspects far beyond textbook theory have to be taken into consideration -- many of which might also be left out for first demonstrator systems. Due to the predominant role of robustness and reliability, any kind of uncertainty in the system has to be considered carefully. Among others, the uncertainties and changes of the acoustical environment, e.g. due to temperature changes, number of passengers, open windows and also of system components, e.g. loudspeakers and microphones have to be measured and/or modeled. This also gives some new insight into the acoustical environment inside cars at low frequencies. In addition, the quality of input data (e.g., the update rate for rpm information) is of major importance for the acoustical performance in terms of noise reduction. Finally, stability analyses and robustness calculations must be extended to incorporate uncertainties as well as time domain effects even for more or less frequency domain problems like engine noise. This requires to re-formulate feed-forward systems as feed-back systems and calculate system responses. Measurement results on the interior acoustics (and its uncertainty) as well as additional developments on ANC system robustness will be presented.

8:40

**4aNSd3. Fluctuations under turbulent flows: Enhanced methods for separation of propagative wavenumbers from wall pressure dataset.**

Sébastien Debert (PSA-Peugeot-Citroën, Route de Gizy, CC : VV013 - Bât. 14, F-78943 Vélizy-Villacoublay Cedex, France, sebastien.debert@mpsa.com), Marc Pachebat (PSA-Peugeot-Citroën, Route de Gizy, CC : VV013 - Bât. 14, F-78943 Vélizy-Villacoublay Cedex, France, marc.pachebat@mpsa.com), 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), Yves Gervais (Laboratoire d'Etudes Aérodynamiques (LEA), Université de Poitiers - ENSMA - CNRS, Bâtiment K, 40 Avenue du Recteur Pineau, 86022 Poitiers, France, yves.gervais@lea.univ-poitiers.fr)

This work is a part of a more general study on automobile interior noise due to acoustic and aerodynamic wall-pressure fluctuations. Using experimental data of wall-pressure fluctuations measured with a microphone array beneath several kinds of flows, a wave-number analysis based on recently developed signal processing methods -- the spatial Empirical Mode Decomposition (sEMD) and the Ensemble Empirical Mode Decomposition (E-EMD) -- is carried out, in order to separate acoustic and aerodynamic pressure fluctuations. In opposition with an existing classical method previously used, based on the spatial correlogram, these methods do not require a stationary uniform flow. A turbulent boundary layer on a flat plate and a detached/reattached flow downstream three different forward-facing steps are tested, with flow velocities from 0 to 40 m.s<sup>-1</sup>, with or without a well-controlled artificial acoustic source. The sEMD method is first used as a wavenumber filter, and is shown to improve the detection of acoustic fluctuations.

tuations of about 5 dB on classical wavenumber ( $k, f$ ) representations. The E-EMD method is developed in order to decompose the  $(x, t)$  representation to make possible the separation of the acoustic and aerodynamic components, regardless of the stationarity or the uniformity of the flow.

9:00

**4aNSd4. Stochastic modeling of the vibro-acoustic behavior of production cars.** Laurent Gagliardini (PSA Peugeot Citroën, Route de Gisy, 78943 Vélizy-Villacoublay Cedex, France, laurent.gagliardini@mps.com), Jean-Francois Durand (Université Paris-Est, Laboratoire de Mécanique, 5 bd Descartes, 77454 Marne-la-Vallée, France, jean-francois.durand@astrium.eads.net), Christian Soize (Université de Marne la Vallée, 5, Boulevard Descartes, 77454 Marne la Vallée, France, soize@univ-mlv.fr)

Production cars -as any industrial product- are subject to various causes of variability including process uncertainties or product diversity. Many authors have shown that vibroacoustic problems sensitivity to small uncertainties increases dramatically with frequency until only statistical approaches remain relevant in the high frequency range. Moreover, modeling uncertainties due to numerous model simplifications induce similar dispersion effects on the computed responses. Both kind of uncertainties may be addressed when using a non-parametric stochastic modeling, based on the random matrices theory. Such a modeling, appears to be very practicable for industrial vibroacoustic problems while relying on a strong mathematical background. In a first part, the application of the non-parametric modeling of uncertainties to vibro-acoustics problems will be addressed. Stochastic aspects are controlled by only 7 dispersion parameters that provide most of the dynamic behaviors that can be observed experimentally. A Monte-Carlo simulation is performed to provide converged statistics of the stochastic problem solution. In a second part, the dispersion parameters are identified so that the stochastic model fits experimental data. Frequency Response Functions of 25 production vehicles were measured for this purpose and compared to the computed results in the low frequency range ( $<200$  Hz).

9:20

**4aNSd5. Improved spectrofilter applied to diesel engine noise : combustion and mechanical noise separation.** Laurent Pruvost (Laboratoire Vibrations Acoustique, Insa Lyon, 25 bis, av. J. Capelle, 69621 Villeurbanne Cedex, France, laurent.pruvost@insa-lyon.fr), Quentin Leclerc (Laboratoire Vibrations Acoustique - INSA Lyon, 25 bis avenue Jean Capelle, Bâtiment Saint-Exupéry, F-69621 Villeurbanne cedex, France, quentin.leclerc@insa-lyon.fr), Etienne Parizet (Laboratoire Vibrations Acoustique, Insa Lyon, 25 bis, av. J. Capelle, 69621 Villeurbanne Cedex, France, etienne.parizet@insa-lyon.fr)

For engineers working on alternative fuels and diesel engine sound quality, being able to separate combustion noise and mechanical noise would be of prime interest. This separation can be attempted using a spectrofilter (also called Wiener filter) when in-cylinder pressure signals can be recorded. The major drawback of the spectrofilter is its inability to separate correlated sources like combustion and piston slap. An upgraded version of the spectrofilter can be computed. Its computation just requires to consider only the random parts of the engine signals. Actually, considering these random parts artificially uncorrelates the noise sources. Highly correlated signals have been synthesised and successfully separated by the upgraded spectrofilter. When dealing with real-life engine noise, the quality of the separation cannot be judged directly since the signals to separate are unknown. The spectrofilter's causality and stability have been used as criteria to judge its quality. These two criteria both confirmed the superiority of the spectrofilter computed using the random parts of the signals. A synchronous averaging step is required to estimate the signals random parts. This estimation has been found to depend on the phase-locking strategy.

9:40

**4aNSd6. Prediction of the Excitation Force Based on the Dynamic Analysis for Flexible Model of a Powertrain.** Yoon Sug Kim (Inha University, Mechanical Engineering, 253 Yonghyun Dong, 402-751 Incheon, Republic of Korea, black\_boxter@hanmail.net), Sung Jong Kim (Inha University, Mechanical Engineering, 253 Yonghyun Dong, 402-751 Incheon,

Republic of Korea, kimsungjong@gmail.com), Min Geun Song (Inha University, Mechanical Engineering, 253 Yonghyun Dong, 402-751 Incheon, Republic of Korea, somiku@hanmail.net), Sang Kwon Lee (Inha University, Mechanical Engineering, 253 Yonghyun Dong, 402-751 Incheon, Republic of Korea, sangkwon@inha.ac.kr)

The powertrain is one of the import sources for the interior noise. In order to predict the interior noise due to a powertrain, the experimental method is has been used based on the TPA (transfer path analysis). Although this experimental method is a useful tool for the identification of the noise source and the transfer path due to the powertrain, it is difficult to modify the structure of a powertrain by using the experimental method for the reduction of vibration and noise. In order to solve this problem, the paper presents the noble approach for the prediction of interior noise caused by the vibration of the powertrain based on the hybrid TPA (transfer path analysis) technology. Therefore, the vibration of the powertrain in a vehicle is numerically analyzed by using FEM (finite element method). The vibration of the other part in a vehicle is investigated by using the experimental method based on VATF (vibrato-acoustic transfer function) analysis. These two methods are combined for the prediction of interior noise caused by powertrain. This paper present the prediction of the excitation force of the powertrain to the vehicle body based on numerical simulation.

10:00-10:20 Break

10:20

**4aNSd7. Sound quality evaluation of air-conditioning sounds in a vehicle using psychoacoustic parameters.** Ryota Nakasaki (Utsunomiya Univ., Calsonickansei Corp., 7-3 Sakae-cho, Tochigi-ken, 327-0816 Sano-shi, Japan, ryota\_nakasaki@ck-mail.com), Takaharu Ogata (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, mt076508@ced.is.utsunomiya-u.ac.jp), Hiroshi Hasegawa (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, hasegawa@is.utsunomiya-u.ac.jp), Yukio Ozeki (Utsunomiya Univ., Calsonickansei Corp., 7-3 Sakae-cho, Tochigi-ken, 327-0816 Sano-shi, Japan, ryota\_nakasaki@ck-mail.com), Masaharu Onda (Utsunomiya Univ., Calsonickansei Corp., 7-3 Sakae-cho, Tochigi-ken, 327-0816 Sano-shi, Japan, ryota\_nakasaki@ck-mail.com), Masao Kasuga (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, kasuga@is.utsunomiya-u.ac.jp)

With recent developments of noise reduction technologies from mechanical components of a vehicle, air-conditioning systems become a major noise source in the compartment of a vehicle. To improve quietness in the compartment further, it is important not only to reduce the sound pressure level (SPL), but also to improve the sound quality. In this study, we tried to evaluate the air-conditioning sounds from a viewpoint of sound quality. First, we carried out a subjective evaluation experiment using some evaluation words that represent characteristics of the air-conditioning sounds. As a result of a factor analysis, noting that the air-conditioning sounds can be explained by three factors of "rough," "space" and "refreshing." "Rough factor" has strong correlation with the SPL, however, "space factor" and "refreshing factor" have little correlation with SPL. Next, to investigate parameters that correlate with these factors, we carried out an experiment to evaluate the air-conditioning sounds using the psychoacoustic parameters of loudness and sharpness. As a result, noting that "rough" factor strongly correlates with loudness, "space" and "refreshing" factors correlate with sharpness. This result shows that it is possible to evaluate the air-conditioning sounds a viewpoint of sound quality by using the sharpness that is different from SPL.

10:40

**4aNSd8. Low noise truck study for distribution in urban areas.** Nicolas Blairon (Volvo 3P, 99 route de Lyon, 69802 Saint - Priest, France, nicolas.blairon@volvo.com), Bruno Carton (Volvo 3P, 99 route de Lyon, 69802 Saint - Priest, France, bruno.carton@volvo.com)

Today, vehicle noise is identified in opinion pools as one of the main annoyance source in urban areas. To face this issue, truck manufacturers are today developing low noise vehicles in order to allow their customer to deliver goods in urban areas during night. An acoustic research program has

been launched by Volvo 3P (Volvo Trucks, Renault Trucks and Mack Trucks): the LUT project as Low Noise Urban Truck. The partners of the project are Michelin, the Institut National des Sciences Appliquées de Lyon (INSA) - a French university - and Marmonier - a noise shield manufacturer. This project has been founded by ADEME. The target in terms of noise reduction is ambitious because the low noise truck will emit an acoustic power four times lower than today's vehicles in specific rolling conditions! Methods and tools have been developed to analyze the vehicle noise sources in urban conditions and define acoustic solutions to reduce the main vehicle noise sources.

11:00

**4aNSd9. Operational transfer path analysis: Comparison with conventional methods.** Martin Lohrmann (Müller-BBM Vibroakustik Systeme, Robert Koch Strasse 13, 82152 Planegg / München, Germany, MLohrmann@MuellerBBM-vas.de)

Transfer Path Analysis describes how sound and vibration propagates through complex structures. The correct determination of transfer coefficients between sources and receivers is essential for a high quality analysis. Conventional methods use artificial excitations (forces or volume velocities) to evaluate transfer functions. Therefore, they do not consider the influence of different load conditions on the transfer function behaviour of complex structures. In a second step, operational data (vibrations or sound pressures) are applied to the transfer functions to calculate the different path contributions. In order to overcome these restrictions, transfer coefficients can be evaluated directly from operational data. Thus, the actual load conditions are taken into account and the quality of transfer characteristics is improved. The relation between simultaneously measured data of sources and receivers can be derived by statistical methods. Principal Component Analysis is used to separate the total signal into individual path contributions, while operating on airborne and structure-borne contributions simultaneously. Operating in the time-domain gives direct access to auralisation and in-depth analysis of dominant path contributions. This new single-step approach has been proven to generate more precise analysis results within a much shorter testing time.

11:20

**4aNSd10. A transversal substructuring modal method for the acoustic analysis of dissipative mufflers with mean flow.** José Albelda (Dept. Ingeniería Mecánica y de Materiales. Univ. Politécnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, jalbelda@mcm.upv.es), Francisco D. Denia (Dept. Ingeniería Mecánica y de Materiales. Univ. Politécnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, fradegu1@mcm.upv.es), F. Javier Fuenmayor (Dept. Ingeniería Mecánica y de Materiales. Univ. Politécnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, ffuenmay@mcm.upv.es), Manuel J. Martínez (Dept. Ingeniería Mecánica y de Materiales. Univ. Politécnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, mamarbor@doctor.upv.es)

This work presents a modal approach to evaluate the transversal modes and wavenumbers for dissipative mufflers with mean flow. The method is based on the division of the transversal section of the muffler into subdomains, for which two simple sets of modes are considered. The first set of modes satisfies the condition of zero pressure at the common boundary between subdomains, while the second fulfils the condition of zero derivative in the direction normal to this boundary. From these sets, a substructuring procedure is applied that provides the final modes of the complete cross section, considering the presence of absorbent material, a perforate and mean flow. The technique avoids iterative schemes associated with the nonlinear

characteristic equation found, for instance, in the analytical modelling of perforated dissipative circular mufflers. Once the final transversal modes have been calculated, the mode matching technique is applied at the geometrical discontinuities to completely define the acoustic field inside the muffler. The acoustic attenuation is then predicted by means of the transmission loss. Comparison with finite element calculations and results available in the literature show good agreement. The attenuation of some selected mufflers is analysed, including the effect of the perforate and the mean flow.

11:40

**4aNSd11. Prediction of sound transmission through automotive door seal systems.** Bertrand Andro (Renault, Technocentre, 1 avenue du Golf, 78288 Guyancourt, France, bertrand.andro@renault.com), Sébastien Chaigne (Renault, Technocentre, 1 avenue du Golf, 78288 Guyancourt, France, sebastien.chaigne@renault.com), Alpha Diallo (Renault, Technocentre, 1 avenue du Golf, 78288 Guyancourt, France, alpha.diallo@renault.com), Matthieu Mermet (CEMEF, 1, rue Claude Daudenne, 06904 Sophia Antipolis, France, matthieu.mermet-renexter@renault.com)

In automotive industry the door seal systems is an important contributor to vehicle interior noise in the middle and high frequency range. The aim of the study was to develop a numerical model in order to predict the sound transmission loss through elastomeric seals. At the early stage of the development process, this type of numerical tool is very interesting to investigate the influence of the design parameters of the seal. Two steps were necessary: a static analysis to calculate the seal shape after compression (door closure event), an acoustic analysis based on dynamic parameters to determine the sound transmission. Finite element methods were used for both steps (commercial softwares). These two steps were validated experimentally for two types of geometry, different compression ratios and loading cases. One original contribution of the paper concerns the determination of a valid nonlinear model for the static part and of a good approximation of the dynamic behavior of the elastomer's Young modulus. Finally, a sensibility analysis was performed in order to evaluate the influence of the design parameters of the door seal system such as the compression ratio or the dynamic parameters.

12:00

**4aNSd12. Quantification of structural damping of a multi-layered windshield at low and medium frequencies.** Manuel Etchessahar (PSA Peugeot Citroën, Route de Gisy, 78943 Vélizy-Villacoublay Cedex, France, manuel.etchessahar@mpsa.com), Laurent Gagliardini (PSA Peugeot Citroën, Route de Gisy, 78943 Vélizy-Villacoublay Cedex, France, laurent.gagliardini@mpsa.com)

Structural damping is known as one of the most efficient design variable in order to reduce structure-borne noise. At low and medium frequencies (100-500 Hz), the vibro-acoustic behavior of the car is essentially governed by large panels as windshield, roof or front panel for example. In order to improve the car body design regarding cost, weight, performances and reliability, one must have a good understanding of the damping performances of these large panels submitted to various operating conditions. In a previous paper, the authors have proposed an improved method for structural damping numerical assessment of structural elements with non uniform damping using a stochastic distribution of input forces. In the present paper, this method is applied to quantify the structural damping of a multi-layered windshield at low and medium frequencies. Effects of the geometry, of the viscoelastic properties of the PVB layer and of the windshield glue bead on the total damping properties will be studied.

## Session 4aNSE

## Noise and EURONOISE: Noise from Wind Power Projects I

Eddie Duncan, Cochair

*Resource Systems Group (RSG), 55 Railroad Row, White River Junction, VT 05001, USA*

Kerstin Persson Waye, Cochair

*Dept. of Environ. Medicine, The Sahlgrenska Acad. of Gothenburg Univ., Box 414, Gothenburg, 405 30, Sweden**Invited Paper*

10:20

**4aNSE1. Improving sound propagation modeling for wind power projects.** Eddie Duncan (Resource Systems Group (RSG), 55 Railroad Row, White River Junction, VT 05001, USA, eduncan@rsginc.com), Kenneth Kaliski (Resource Systems Group (RSG), 55 Railroad Row, White River Junction, VT 05001, USA, kkaliski@rsginc.com)

Sound propagation from wind power projects can be modeled in the same manner as other more common outdoor noise sources, but are these models suited to wind turbines' uniquely high source heights, operating under high wind conditions, and various degrees of terrain ruggedness. In "Propagation Modeling Parameters for Wind Turbines" (K. Kaliski and E. Duncan, Proceedings of Institute of Noise Control Engineers NOISECON 2007), the effects of ground attenuation and various adjustments for wind conditions on sound propagation modeling were discussed. This paper continues the discussion and explores the accuracy of existing sound propagation modeling methods for wind power projects including ISO 9613 and other standards. Model data for wind power projects and the implications of various terrain and ground coverage will be discussed.

*Contributed Papers*

10:40

**4aNSE2. A review of the use of different noise prediction models for windfarms and the effects of meteorology.** Graham Parry (ACCON UK Limited, Unit B, Fronds Park, Frouds Lane, Aldermaston, RG7 4LH Reading, UK, graham.parry@accon-uk.com)

As a result of involvement in a specific wind farm development at Guestwick, Norfolk and a requirement to determine the efficacy of competing noise prediction models a review of the potential impacts of a six-turbine wind farm was carried out. The paper considers the results of comparing three specific noise prediction methods and algorithms and determines the extent to which adherence to either one of the methodologies could result in relatively large differences in predicted noise levels under varying wind conditions and accordingly the potential for differing conclusions being reached as to the acceptability of the wind farm with respect to the ETSU-R-97 assessment methodology. The paper also examines other noise modelling research carried out on behalf of ETSU.

11:00

**4aNSE3. Noise Impact of Wind Farms: Uncertainties due to wind data reference at 10m.** David Slaviero (Acouphen Environnement, Campus de la DOUA, 66, BD Niels Bohr, BP 52132, 69603 Villeurbanne, France, david.slaviero@acouphen-environnement.com), Alexis Bigot (Acouphen Environnement, Campus de la DOUA, 66, BD Niels Bohr, BP 52132, 69603 Villeurbanne, France, alexis.bigot@acouphen-environnement.com)

Noise impact prediction or measurement of Wind Farms requires wind reference data at 10 meters. From data collected on more than 200 Farms in France, over the last 5 years, this paper shows that, in the sampling of noise and meteorological data for noise prediction and assessment, differences in the wind profile from one site to another, in the period of the day (day, night) can lead to different estimations of emission data selection and impact assessment. Different parameter studies are presented both with regard to the effect on impact studies and farms impact monitoring and control. The work presented is in accordance with the project of acoustic standard for the assessment of noise impact of Wind Farms under study in France.

*Invited Papers*

11:20

**4aNSE4. Models of natural background noise and masking of wind turbine noise.** Karl Bolin (KTH/MWL, Teknikringen 8, SE100 44 Stockholm, Sweden, kbolin@kth.se)

Wind turbine (WT) noise limits adjusted to background noise levels are used in several countries among others Britain and France. To determine the background noise level extensive measurements at locations near the proposed WT site are performed. This paper present methods to avoid these measurements in woodland and coastal areas, it also include a pre-study concerning the audibility of WT noise when mixed with background noise. A prediction model for noise from vegetation is described. This has been coupled to wind field simulations and fluctuations of vegetation noise can therefore be predicted. Measurements and a model for sea wave noise are also presented. Furthermore the paper present results from psycho acoustic tests with 8 subjects. These involve hearing thresholds and partial loudness when WT noise is mixed with background noise. These are compared to two loudness models. Two different WT sounds have been used as stimuli. The first sound is from a single WT and the second sound is from a WT park. Results show how natural background noises influence the audibility of WT noise and could be used as a tool to optimize the power generated from WTs without causing disturbance among nearby residents.



11:40

**4aNSe5. Criteria for wind farm noise: Lmax and Lden.** Frits Van Den Berg (University of Groningen - Science & Society Group, Nijenborgh 4, 9747AG Groningen, Netherlands, fvdberg@ggd.amsterdam.nl)

Wind turbine noise limits are based on either the highest sound immission level (Lmax) or several sound immission levels for a series of wind speed classes (Lmax,v). As yet no procedure has been proposed to determine the day-evening-night sound level (Lden) that is now commonly used in the European Union for all noise sources. Wind speed dependent rating wind turbine noise levels  $L_{r,v}$  can be predicted based on climatological data. This has been verified by measurements over a nine month period for a wind farm at a coastal location in the Netherlands. From these measurements also the long term average sound level Lden can be determined. Lden can also be determined from previously published wind speed measurements at an inland location over one year. The procedure shows that for a wind turbine or wind farm the Lden can be derived from Lmax by taking into account the regional climatology.

12:00

**4aNSe6. Response to wind turbine noise in the Netherlands.** Eja Pedersen (Occupational and Environmental Medicine, Göteborg University, PO Box 100, SE-405 30 Göteborg, Sweden, eja.pedersen@set.hh.se), Jelte Bouma (Northern Centre for Healthcare Research, University Medical Centre Groningen, PO Box 30001, 9700 RB Groningen, Netherlands, j.bouma@med.umcg.nl), Roel Bakker (Northern Centre for Healthcare Research, University Medical Centre Groningen, PO Box 30001, 9700 RB Groningen, Netherlands, R.H.Bakker@med.umcg.nl), Frits Van Den Berg (University of Groningen - Science & Society Group, Nijenborgh 4, 9747AG Groningen, Netherlands, fvdberg@ggd.amsterdam.nl)

A cross-sectional study with the objective to explore the impact of wind turbine noise on people living in the vicinity of wind farms was carried out in the Netherlands in 2006. A postal questionnaire assessing response to environmental exposures in the living area, including wind turbine noise, was answered by 725 respondents (response rate: 37%). Immission levels of wind turbine noise outside the dwelling of each respondent were calculated in accordance with ISO-9613. The risk for being annoyed by wind turbine noise outdoors increased with increasing sound levels ( $r_s = 0.501$ ,  $n = 708$ ,  $p < 0.001$ ). The risk for annoyance was decreased for respondents who could not see wind turbines from their dwelling and for respondents who benefited economically from the turbines. No statistically significant correlations between immission levels of wind turbine noise and health or well-being were found. However, noise annoyance due to wind turbine noise was associated with stress symptoms, psychological distress and lowered sleep quality.

12:20

**4aNSe7. Laboratory assessment of noise annoyance from large wind turbines.** Steffen Pedersen (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark, stp@acoustics.aau.dk), Henrik Møller (Acoustics, Aalborg University, Fredrik Bajers Vej 7 B5, 9220 Aalborg Ø, Denmark, hm@acoustics.aau.dk)

An investigation of the annoyance from the wind turbine noise, to which neighbors may be exposed, is carried out. The aim is to obtain dose-response relationships and to uncover if specific noise components (e.g. low-frequencies) are primary contributors to the annoyance. In the experiments, sounds recorded close to large wind turbines are filtered (and levels adjusted accordingly) to represent indoor and outdoor positions at the neighbors' dwellings and played back in the laboratory. Challenges relating to the recording and transformation of sounds are discussed. The exposure technique is a combination of an advanced low-frequency chamber that can reproduce the frequency range 2-250 Hz (with uniform distribution in the room) and additional loudspeakers for the higher frequencies. The listening test is a randomized design. The stimuli, of 10 minute duration, are presented at three levels and in combinations of filtered versions (low- and mid-frequency) such that the influence of low-frequency tonal components and level fluctuations is investigated. 25 subjects are exposed to the stimuli while reading a novel and afterwards they rate annoyance on a visual analog scale.

## Session 4aNSf

## Noise and EURONOISE: Fan Noise and Low-Mach Number Rotating Blade Noise I

Scott C. Morris, Cochair

*Notre Dame, 109 Hessert Laboratory, University of Notre Dame, Notre Dame, IN 46556, USA*

Michel Roger, Cochair

*Ecole Centrale de Lyon, 36 Avenue Guy de Collongue, Centre Acoustique, Ecully, 69134, France*

## Contributed Papers

10:40

**4aNSf1. Overview of turbofans noise prediction methods based on CFD computations.** Cédric Morel (Snecma, Site de Villaroche, Rond-Point René Ravaud, 77550 Moissy-Cramayel, France, cedric.morel@snecma.fr), Benoît Farvacque (Snecma, Site de Villaroche, Rond-Point René Ravaud, 77550 Moissy-Cramayel, France, benoit.farvacque@snecma.fr)

Overview of turbofans noise prediction methods based on CFD computations. The constant trend to increase bypass ratio in turbofan aircraft engines has led to an increase of the relative contribution of rotors to the overall engine noise. There is therefore a need for efficient and accurate prediction tools to achieve a silent design of the rotor components. Here is given an overview of the different aeroacoustic methods investigated by Snecma and used for prediction of tone interaction noise, broadband interaction noise and broadband fan self noise. Different levels of methods refinement, from various analytical models to steady and unsteady CFD, will be compared in terms of accuracy of prediction and computational cost. Examples of implementation of these methods on conventional turbofan and counter-rotating fan configurations will be presented.

11:00

**4aNSf2. Benchmark of fan noise propagation tools.** Jacky Mardjono (SNECMA Villaroche, 77550 Moissy Cramayel, France, jacky.mardjono@snecma.fr)

Various propagating codes based on different formulations have been developed over the last years to simulate the fan noise propagation into the nacelle of a turbofan engine and its radiation in the Far field. The abilities of the ACTRAN softwares (Potential formulation / FEM), SPACE (Linearised EULER equations / DGM) and ICARE (Rays-tracing method) to simulate typical industrial configurations in the frequential domain have been evaluated based on measurements carried out at the RACE acoustic test facility on a fan model at 1/2 scale. The results of the computations carried out with each of these codes on intake and exhaust configurations (axisymmetric assumption) in hard wall and lined duct will be compared to the measured far field directivities. These comparisons will be discussed depending on the establishment of patterns (sources types, mean flows types, boundaries conditions) considered in the simulations of each of these codes. The impacts of the considered convected mean flow types (Uniform, Euler, NS,...) and the capacities of these codes to predict noise attenuation of the treatments will be investigated in particular.

11:20

**4aNSf3. Comparative helicopter noise analysis in static and in-flight conditions.** Doris Novak (University of Zagreb, Vukeliceva 4, 10000 Zagreb, Croatia, doris.novak@fpz.hr)

Rotary wing aircraft, i.e. helicopter, is a source of intense noise, external and internal alike, in conclusion becoming serious environmental and health issue. The generated noise is in some aspects similar to propeller noise in

fixed wing aircraft (airplane), while differing in main noise source alignment in respect to the relative airflow: in helicopters, both rotors, i.e. main and tail, that produce forces necessary for flight, are inline with the direction of flight, while in airplanes rotor(s) are perpendicular to it. Another distinctive noise in helicopters, well known as "slapping", comes from the rotor cutting its own wake/vortex air inflow, especially while descending. In this article main helicopter noise sources will be discussed and most significant results of various static and in-flight noise measurements on two different types of helicopters will be presented and analyzed.

11:40

**4aNSf4. Validation of Thickness and Loading Noise Codes by Isom's Formula Applied to Subsonic Axial and Centrifugal Fans.** Sofiane Khelladi (Lab. d'Energétique et de Mécanique des Fluides Interne, Arts et Métiers ParisTech, 151 boulevard de l'Hôpital, 75013 Paris, France, sofiane.khelladi@paris.ensam.fr), Djafer Fedala (Lab. d'Energétique et de Mécanique des Fluides Interne, Arts et Métiers ParisTech, 151 boulevard de l'Hôpital, 75013 Paris, France, djafer.fedala@paris.ensam.fr), Farid Bakir (Lab. d'Energétique et de Mécanique des Fluides Interne, Arts et Métiers ParisTech, 151 Boulevard de l'Hôpital, 75013 Paris, France, farid.bakir@paris.ensam.fr), Robert Rey (Lab. d'Energétique et de Mécanique des Fluides Interne, Arts et Métiers ParisTech, 151 Boulevard de l'Hôpital, 75013 Paris, France, robert.rey@paris.ensam.fr)

The thickness noise predicted by the Ffowcs Williams and Hawkings (FW&H) equation depends on the normal velocity  $v_n$  which is very sensitive to the meshing size. Isom showed that in far field a monopolar source is equivalent to a dipolar source induced by a uniform distribution of the load on the entire moving surface. Consequently, the calculation of the thickness noise becomes completely independent of the normal velocity  $v_n$ . Its expression, as suggested by Farassat, is for any moving surfaces. The main objective of this paper is to determine a specific expression of Isom's thickness noise in time and frequency domains for axial and centrifugal subsonic fans. The scope of the proposed expression of Isom's thickness noise is threefold: (1) highlight the effect of each geometrical parameter of the fan on the overall thickness noise, (2) a fast computational mean and low memory storage capability since the acoustic pressure in frequency domain is calculated for only one blade, and (3) a benchmark test of consistency for thickness and loading noise codes in both time and frequency domains when using the free field solution of FW&H's equation.

12:00

**4aNSf5. Prediction of the tonal noise radiation of an axial flow fan located downstream of a Bluff Body.** Richard Gault (Queen's University Belfast, School of Mechanical and Aerospace Engineering, Ashby Building, Stranmillis Road, BT9 5AH Belfast, UK, r.gault@qub.ac.uk), Richard K. Cooper (Queen's University Belfast, School of Mechanical and Aerospace Engineering, Ashby Building, Stranmillis Road, BT9 5AH Belfast, UK,

R.Cooper@qub.ac.uk), Jian Wang (Queen's University Belfast, School of Mechanical and Aerospace Engineering, Ashby Building, Stranmillis Road, BT9 5AH Belfast, UK, j.wang@qub.ac.uk), Christophe Schram (LMS International, Researchpark Z1, Interleuvenlaan 68, 3001 Leuven, Belgium, Christophe.Schram@lmsintl.com)

This paper investigates the tonal noise radiated by a subsonic axial flow fan when installed downstream of a bluff body. Typically industrial axial flow fans operate in cluttered environments, and are usually driven through direct coupling from an engine. For this investigation, the fan is located in a knife-edged shroud and placed downstream of a bluff body having representative dimensions of a typical engine. Axial flow fans in general radiate

broadband and discrete frequency noise, the latter of which modelling efforts are maturing. The numerical simulation is based on the aero-acoustic analogy where the unsteady flow is first computed using CFD and then passed to a BEM solver to compute the acoustic radiation. URANS and DES methods are examined for the turbulence modelling. The time-varying force on a single blade in the CFD solver over one complete rotation is used to construct the equivalent fan source in the BEM model. Experimental measurements of Sound Pressure Level are performed in a hemi-anechoic chamber, and a comparison between numerical predictions made. This numerical procedure can be used to further aid designers in understanding the effects of fan tonal noise in cluttered environments.

## Invited Papers

12:20

**4aNSf6. Near-field scattering of ducted fan noise using a boundary element method.** Christophe Schram (LMS International, Researchpark Z1, Interleuvenlaan 68, 3001 Leuven, Belgium, Christophe.Schram@lmsintl.com)

The noise emitted by rotating machinery is a concern in many applications, such as aeroengines, wind turbines, and cooling devices for IC engines or electronic appliances. A specific derivation of the Ffowcs Williams and Hawkins analogy for the tonal noise emitted by a fan was presented by Goldstein (1976), under the assumption that the listener is placed in the acoustical and geometrical far field. That formulation accounts for the modulation of the Doppler frequency shift during the fan revolution, but neglects near field effects. This work presents the application of an alternative derivation, introduced by Roger (2007), which preserves the near-field features of the sound field. This analytical method is compared with a second method in which the fan is modelled by a fixed azimuthal distribution of dipoles. A validation is performed for the case of a generic fan located in an infinite circular duct. The sound field within the duct is obtained by two means: i) calculating the sound field emitted by the fan, modelled by the above mentioned approaches, and scattered by the duct through the application of a Boundary Element Method, ii) computing the sound field by projecting the source in the duct modes.

12:40

**4aNSf7. LES prediction of wall-pressure fluctuations and noise of a low-speed airfoil.** Meng Wang (University of Notre Dame, Department of Aerospace and Mechanical Engineering, Notre Dame, MD 46556, USA, m.wang@nd.edu), Stéphane Moreau (Valeo Thermal Systems, rue Louis Normand, 8, 78321 La Verrière, France, stephane.moreau@valeo.com), Gianluca Iaccarino (Stanford University, Dept. of Mechanical Engineering, Stanford, CA 94305, USA, jops@ctr.stanford.edu), Michel Roger (Ecole Centrale de Lyon, 36 Avenue Guy de Collongue, Centre Acoustique, 69134 Ecully, France, michel.roger@ec-lyon.fr)

The wall-pressure fluctuations and noise of a low-speed airfoil are computed using large-eddy simulation (LES). The results are compared with experimental measurements made in an open-jet anechoic wind-tunnel at Ecole Centrale de Lyon. To account for the effect of the jet on airfoil loading, a RANS calculation is conducted in the full wind-tunnel configuration, which provides velocity boundary conditions for the LES in a smaller domain within the potential core of the jet. The flow field is characterized by an attached laminar boundary layer on the pressure side and a transitional and turbulent boundary layer on the suction side. The predicted unsteady surface pressure field shows reasonable agreement with the experimental data in terms of frequency spectra and coherence in the trailing-edge region. In the nose region, characterized by unsteady separation and transition to turbulence, the wall-pressure fluctuations are highly sensitive to small perturbations and difficult to predict or measure with certainty. The LES, in combination with the Ffowcs Williams and Hall solution to the Lighthill equation, also predicts well the radiated trailing-edge noise. A finite-chord correction is derived and applied to the noise prediction, which is shown to improve the overall agreement with the experimental sound spectra.

1:00-2:00 Lunch Break

## Contributed Papers

2:00

**4aNSf8. The influence of the design parameters of centrifugal fans on the difference between outlet and inlet noise levels.** Mikhail Y. Liberman (MOVEN Co., 17 Plekhanova str., 24/30 Zemlyanoi val str., apt. 27, 111141 Moscow, Russian Federation, mikhail.liberman@yahoo.com)

The operation of the industrial centrifugal fans is accompanied by the air-borne noise generation thanks to such phenomenon as: vortex formation within blade channels of impeller (because of flow separation in channel) and interactions between vortex wakes (at outlet of impeller) and cut-off of fan casing. Sound waves, which are formed at outlet of blade channels and within fan casing, propagate as through outlet side of casing, as in opposite direction: through rotating blade channels of impeller and inlet cone of casing. As it follows from theoretical analysis and experimental research the noise levels at outlet side of centrifugal fan are higher than noise levels at inlet of fan (as for broadband as for tonal noise), because of noise reduction, which is caused by sound wave propagation through inhomogeneous channel, consisted of impeller and inlet cone of casing. The efficiency of the

noise reduction depends on the design parameters of impeller and casing. In particular, according to results of research the difference between levels of noise, radiated from outlet and inlet sides of fan, depend on: shape of blade channels, relation between width of casing and impeller.

2:20

**4aNSf9. Prediction of flow induced noise in rotating devices using nonmatching grids.** Jens Grabinger (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, jens.grabinger@lse.eei.uni-erlangen.de), Branimir Karic (University Erlangen-Nuremberg, Institute of Fluid Mechanics, Cauerstr. 4, 91058 Erlangen, Germany, branimir.karic@lstm.uni-erlangen.de), Simon Triebenbacher (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, simon.triebenbacher@lse.eei.uni-erlangen.de), Manfred Kaltenbacher (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, manfred.kaltenbacher@lse.eei.uni-erlangen.de), Stefan

Becker (University Erlangen-Nuremberg, Institute of Fluid Mechanics, Cauerst. 4, 91058 Erlangen, Germany, stefan.becker@lstm.uni-erlangen.de), Reinhard Lerch (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, reinhard.lerch@lstm.uni-erlangen.de)

With increasing number of electrical devices, e.g. air conditioning systems, used in homes and offices, noise pollution is becoming a more and more relevant topic. A large amount of this noise is generated by turbulent flows and laminar flows at leading and trailing edges, where mainly tonal noise is generated. The objective of our contribution is to investigate shape optimizations of rotating devices in order to reduce their noise levels. For this purpose, we conduct a simulation of the turbulent flow in a ventilator. The acoustic source terms are obtained from the fluid dynamics solution by using Lighthill's acoustic analogy. The acoustic domain is decomposed into a rotating part and a fixed part. The coupling between these two parts is enforced at their interface by a mortar finite element method, which uses Lagrange multipliers in order to "glue" the geometrically independent parts together. The mortar method takes into account the movement of the rotating part through a moving nonmatching grid, that is recomputed at each time step.

2:40

**4aNSf10. Trailing edge noise computation of a fan blade profile.** Julien Christophe (von Karman Institute, Chaussee de Waterloo, 72, 1640 Rhode-Saint-Genese, Belgium, christju@vki.ac.be), Jerome Anthoine (von Karman Institute, Chaussee de Waterloo, 72, 1640 Rhode-Saint-Genese, Belgium, anthoine@vki.ac.be), Stéphane Moreau (Valeo Thermal Systems, rue Louis Normand, 8, 78321 La Verrière, France, stephane.moreau@valeo.com)

In problem involving noise generated by fans or high-lift devices in uniform stationary flow, trailing edge noise has a primary interest. This paper proposes to study the trailing edge noise produced by a Controlled-Diffusion (CD) airfoil specially developed for automotive engine cooling by Valeo. A LES flow computation is realised through the Fluent solver 6.3 for a Reynolds number based on the chord of  $1.5 \times 10^5$  and an angle of attack of 8 degrees. This computation is compared to pressure and velocity measurements performed at ECL (France) and obtained by measurement techniques involving pressure probes and hot-wire anemometry. Amiet's theory using surface pressure spectra around the airfoil trailing-edge region is used to obtain far field acoustic predictions that are compared to microphone measurements.

3:00

**4aNSf11. Optimization problem for the automatic positioning of flow obstructions to control tonal fan noise.** Anthony Gerard (Univ. de Sherbrooke, Mechanical Engineering Depart., 2500 Boulevard de l'Université, Sherbrooke, QC J1K 2R1, Canada, Anthony.Gerard@USherbrooke.ca), Alain Berry (Univ. de Sherbrooke, Mechanical Engineering Depart., 2500 Boulevard de l'Université, Sherbrooke, QC J1K 2R1, Canada, alain.berry@usherbrooke.ca), Patrice Masson (Univ. de Sherbrooke, Mechanical Engineering Depart., 2500 Boulevard de l'Université, Sherbrooke, QC J1K 2R1, Canada, Patrice.Masson@USherbrooke.ca)

Tonal noise from subsonic axial fans can be controlled by adding obstructions in the downstream or upstream flow field of the rotor. These obstructions must be located with care to create secondary circumferential modes of the blade unsteady lift of equal magnitude but opposite in phase with the primary unsteady lift modes responsible for the tonal noise. The general optimization problem of controlling N circumferential modes using N obstructions with 2 degrees of freedom or 2N obstructions with 1 degree of freedom is first posed. This optimization problem can be greatly simplified using nB-periodic obstructions (where B is the number of blade) designed to control the most radiating mode of the blade unsteady lift at frequencies  $n \times \text{BPF}$ . In this case, the control of each frequency is uncoupled and a single error microphone in the axis of the fan can be used to globally control the selected frequencies.

3:20

**4aNSf12. Axial fan noise: Towards sound prediction based on numerical unsteady flow data - a case study.** Hauke Reese (Ansys, Birkenweg 14a, 64295 Darmstadt, Germany, hauke.reese@ansys.com), Thomas Carolus (Institute of Fluid- and Thermodynamic, University of Siegen, Paul-Bonatz-Str. 9-11, 57068 Siegen, Germany, thomas.carolus@uni-siegen.de)

Objective of this work is to evaluate modern numerical methods for predicting flow induced fan noise. A generic fan assembly is investigated consisting of a low pressure axial impeller (diameter 0.3 m, hub to tip ratio of 0.45) including an optional turbulence generator. The flow field is simulated with different state of the art unsteady computational fluid dynamic methods. All results are compared with each other and with hot wire flow velocity and surface pressure measurements. From the numerical data, the relevant dipole sound sources, i.e. the unsteady forces on the fan blades are derived. Eventually both, a free field formulation in the time domain (acoustical analogy by FFWCS WILLIAMS and HAWKINGS), and a boundary element formulation in the frequency domain (SYSNOISE®) are employed to predict the radiated sound field based on the numerical source data. The acoustical results are compared and contrasted with measurements.

### 3:40-5:20 Posters

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

### Contributed Papers

5:20

**4aNSf13. Aeroacoustic Computation of Ducted-Fan Broadband Noise Using LES Data.** Gabriel Reboul (ONERA, 29 avenue Division Leclerc, 92320 Châtillon, France, gabriel.reboul@onera.fr), Cyril Polacsek (ONERA, 29 avenue Division Leclerc, 92320 Châtillon, France, cyril.polacsek@onera.fr), Serge Lewy (ONERA, 29 avenue Division Leclerc, 92320 Châtillon, France, serge.lewy@onera.fr), Sébastien Heib (ONERA, 29 avenue Division Leclerc, 92320 Châtillon, France, sebastien.heib@onera.fr)

Following large efforts to reduce tone noise during the last decades in modern high-bypass ratio turbofans, fan broadband noise reduction has become now an industrial priority. A hybrid computational method providing source-to-far-field predictions of broadband noise due to rotor-stator interaction is presented. The acoustic model is based on the loading term of the

FWH (Ffowcs Williams and Hawkins) equation with a modal Green's function valid for an infinite annular duct, and a Kirchhoff approximation for the free-field radiation. The aerodynamic sources on the airfoils required by the model are expected to be directly issued from a LES (Large Eddy Simulation) computation. The method is applied to a simplified configuration tested in a laboratory rig. The first part of the study is concerned with the assessment of in-duct acoustic field. Usual assumptions about coherence and energy distribution between the acoustic modes are analyzed. PSD (Power Spectrum Density) are calculated through several ways. The second part is focused on the ability to generate an equivalent PSD by means of equivalent source distributions. The purpose is to validate a practical way for coupling LES with Computational Aero-Acoustics Euler solver, in order to include realistic geometry and mean flow effects.



5:40

**4aNSf14. Improvements of a parametric model for fan broadband and tonal noise.** Antoine Moreau (DLR - German Aerospace Center, Mueller-Breslau-Str. 8, 10623 Berlin, Germany, antoine.moreau@dlr.de), Lars Enghardt (DLR - German Aerospace Center, Mueller-Breslau-Str. 8, 10623 Berlin, Germany, lars.enghardt@dlr.de)

The fan of an aero-engine is one of the most significant noise sources of civil aircraft. The purpose of the present work is to provide a tool for fan noise prediction, which can be integrated into the design process of innovative fans. Fan noise is predicted by means of an analytic model. A single noise source is considered: the rotor-stator interaction noise due to rotor blade wakes impinging onto the stator vanes. This source is known to be the major source of fan broadband and tonal noise. The formula for the sound power is based on the single airfoil theory in far field developed by Amiet. It accounts for the effect of subsonic compressible flow and source non-compactness. The model accounts for fan parameters such as geometry (blade and vane count, chord length, rotor-stator gap) and flow parameters (mean velocities, wake and turbulence characteristics). Future work will investigate a more detailed description of the sound field based on an acoustic modal approach, in which each frequency component of the sound spectrum is decomposed into a distribution of duct acoustic modes.

6:00

**4aNSf15. Turbulence ingestion fan noise predictions using an advanced turbulence correlation model.** Scott C. Morris (Notre Dame, 109 Hessert Laboratory, University of Notre Dame, Notre Dame, IN 46556, USA, s.morris@nd.edu)

The generation of sound by an isolated rotating blade row results from the interaction of the blades with turbulence in the approach flow. The relationship between the unsteady velocity field and the sound produced can be expressed directly in terms of the two point velocity correlation function of the approach flow field. It is often assumed that the turbulent flow can be approximated by a number of simplifying assumptions, such as homogeneous, isotropic flow. However, in the frequency range of interest for a number of applications these assumptions are not valid, and their use can lead to significant errors in the prediction of generated sound. The present work will describe a method for using advanced models of two-point velocity correlations and outline new experimental results that validate this approach.

6:20

**4aNSf16. Investigation into the effect of altering incoming gust shape to far field noise radiation in Amiet's theory.** Michael Bilka (von Karman Institute, Chaussee de Waterloo, 72, 1640 Rhode-Saint-Genese, Belgium, bilka@vki.ac.be), Jerome Anthoine (von Karman Institute, Chaussee de Waterloo, 72, 1640 Rhode-Saint-Genese, Belgium, anthoine@vki.ac.be)

Noise prediction for turbulent interactions with fan-blades or high-lift devices has been continuously developed for the last thirty years. A benchmark solution to this problem is Amiet's theory for acoustic radiation from an airfoil in a turbulent stream. In most aerodynamic theories the incoming gust is assumed to be sinusoidal, which is also the case for the theory of Amiet. This assumption affects the derivation of the unsteady lift function which is used to propagate the noise to the far-field. This paper proposes to show the effect of altering the shape of the incoming gust by assuming a Gaussian function. This analysis will then be compared to the case of the sinusoidal gust as well as to acoustic measurements of an airfoil placed in a jet taken in the VKI semi-anechoic chamber.

6:40

**4aNSf17. Aeroacoustics of a low Mach number tip-gap flow.** Julien Grilliat (Ecole Centrale de Lyon, 36, av. Guy de Collongue, 69134 Lyon, France, julien.grilliat@ec-lyon.fr), Marc C. Jacob (Ecole Centrale de Lyon, 36, av. Guy de Collongue, 69134 Lyon, France, marc.jacob@ec-lyon.fr)

A thorough experimental study was performed in several campaigns in the anechoic wind tunnel of the Ecole Centrale de Lyon on a single airfoil at a low Mach number to investigate the tip leakage flow and the associated broadband noise. The influence of the inflow velocity, the airfoil angle of attack and the gap size were characterized and hence an extensive data set was obtained. Both near field aerodynamic and far field acoustic features of this configuration were recorded. Statistical post-processing of the data highlighted some of the governing parameters of this jet-like flow configuration and two different noise generation mechanisms. Some scaling laws were derived from the experimental data which gave also inputs for semi-analytical far field noise prediction models relying on the theory for linearised unsteady aerodynamics around a slender airfoil. The underlying sound generation mechanism is the scattering of the tip clearance flow perturbations by the airfoil trailing edge. The aerodynamic perturbation is described as a gust with a spanwise distribution that is concentrated near the gap region. The models are presented and compared to experimental results and CFD calculations. This work has been funded by the European Community as part of the 6th Framework Project PROBAND n° AST4-CT-2005-012222.

## Session 4aPAa

## Physical Acoustics: Thermoacoustics I

Steven Garrett, Cochair

*Penn State, Applied Research Laboratory, PO Box 30, State College, PA 16804, USA*

Philippe Blanc-Benon, Cochair

*Ecole Centrale de Lyon, LMFA, UMR CNRS 5509, Ecully, Lyon, 69134, France*

## Invited Papers

8:00

**4aPAa1. Recent progress on thermoacoustic heat engines and refrigerators.** Ercang Luo (Technical Institute of Physics and Chemistry, CAS, Beiyitiao Rd., Zhongguancun St., P.O.Box 2711, 100080 Beijing, China, Ecluo@cl.cryo.ac.cn)

In this talk, recent progress on thermoacoustic heat engines and refrigerators developed at the Technical Institute of Physics and Chemistry (TIPC), Chinese Academy of Sciences (CAS), is covered, which mainly includes three aspects: (i) Energy-focused thermoacoustic-Stirling heat engines (EF-TASHE) by using tapered resonators are highlighted for both low-frequency ( $\sim 50\text{Hz}$ ) and medium-frequency ( $\sim 300\text{Hz}$ ) operation; (ii) A thermoacoustically driven two-stage pulse tube cryocoolers capable of achieving liquid hydrogen temperature ( $\sim 20\text{K}$ ) is described. In addition, a 300 Hz compact thermoacoustically driven pulse tube cryocooler operating below 80K is also covered. In this part, an acoustical pressure amplifier is highlighted; (iii) A heat-driven thermoacoustic refrigeration system with double thermoacoustic-Stirling configuration for room temperature cooling is reported. This system is able to provide a cooling power of more than 300 W at  $-20^\circ\text{C}$ , showing good prospect as an alternative of CFC refrigeration. Finally, consideration and prospect for future development are forecasted.

8:20

**4aPAa2. Study of a thermoacoustic-Stirling engine.** Hassan Tijani (Energy research Centre of the Netherlands (ECN), Westerduinweg 3, 1755 LE Petten, Netherlands, tijani@ecn.nl), Simon Spoelstra (Energy research Centre of the Netherlands (ECN), Westerduinweg 3, 1755 LE Petten, Netherlands, spoelstra@ecn.nl), Gaelle Poignand (Energy research Centre of the Netherlands (ECN), Westerduinweg 3, 1755 LE Petten, Netherlands, poignand@ecn.nl)

During the last decade most efforts in thermoacoustics have been focused on the development and understanding of the traveling-wave thermoacoustic systems. These systems get much attention because they employ the inherently efficient Stirling cycle. This makes them much more efficient than the standing-wave counter parts which are intrinsically irreversible. A 1 kW thermal power thermoacoustic-Stirling engine is designed and performance measurements are performed. The engine incorporates a compact acoustic network to create the traveling-wave phasing necessary to operate in a Stirling cycle. The acoustic network consists of a regenerator unit, an acoustic compliance and a feedback inertance. The design, construction and performance measurements of the traveling-wave thermoacoustic engine will be presented and discussed.

8:40

**4aPAa3. Low operating temperature integral thermo acoustic devices for solar cooling and waste heat recovery.** Kees De Blok (Aster Thermoakoestische Systemen, Smeestraat 11, NL 8194 LG Veessen, Netherlands, c.m.deblok@aster-thermoacoustics.com)

Utilizing low temperature differences from solar vacuum tube collectors or waste heat in the range  $70\text{--}200^\circ\text{C}$  seems to be the most promising field of applications for thermoacoustic systems. At these reduced temperatures overall system performance is increasingly affected by the ratio between amplified (useful) power and acoustic power stored in the resonance circuitry. Well known is that this ratio can be improved by deploying multiple regenerator units (hex-reg-hex). However, in commonly used torus or coaxial bypass configurations the correct timing (real and high acoustic impedance) is hardly realized inside more than two regenerator units (soft spot). Acoustic losses in the standing wave resonator account for another fundamental limitation because of the relatively low transferred power at a given pressure amplitude. Therefore a novel acoustic geometry will be presented in which a high and near real impedance can be maintained in even more than two regenerator units and in which acoustic feedback is performed by a true traveling wave. This approach improves the overall performance of integral thermoacoustic systems. Details and experimental results of a solar driven thermoacoustic cooler and ongoing work on thermoacoustic tri-generation utilizing exhaust gas of a standard gas-engine will be presented.

9:00

**4aPAa4. Commercial thermoacoustic products and next-step developments in acoustic cooling.** John Corey (CFIC-Qdrive, 302 Tenth St., Troy, NY 12180, USA, jcorey@cficinc.com), Philip Spoor (CFIC-Qdrive, 302 Tenth St., Troy, NY 12180, USA, pspoor@cficinc.com)

CFIC-Qdrive committed to development and commercialization of thermoacoustic energy conversion devices in 1999, by combining the newly developed acoustic perspective and tools developed at Los Alamos with a deep well of Stirling-cycle experience to achieve a total physics model with both inertial and viscous behavior of the working fluid. Such acoustic-Stirling devices combine the mechanical simplicity, robustness, and efficiency. Focusing first on refrigeration for cryogenics; this work has led to a family of standard products that have found uses worldwide, from air-quality sampling and oil refinery support to military aviation oxygen liquefaction. This paper details the basic operation of these acoustic-Stirling products and the key technological elements that make them viable and attractive in cryogenics; then examines the implications for less-cold uses like food-storage and air-conditioning; with a discussion of achievable performance in accessible applications. We review the work now underway to develop devices to meet those opportunities with environmentally benign cooling of superior performance.

9:20

**4aPAa5. Generating electricity from burning wood using Thermo-acoustics for use in developing countries.** Paul H. Riley (University of Nottingham, Room 1211 Tower Block, School of Electrical and Electronic Engineering, NG7 2RD Nottingham, UK, paul.riley@nottingham.ac.uk), Mark Johnson (University of Nottingham, Room 1211 Tower Block, School of Electrical and Electronic Engineering, NG7 2RD Nottingham, UK, mark.johnson@nottingham.ac.uk)

SCORE is a wood burning stove that will cook food, generate electricity and cooling for use in developing countries by means of thermoacoustics. The consortium of Nottingham (Lead), Manchester, Queen Mary, Imperial and City universities with the charity Practical Action, believe that the very demanding cost targets can be achieved by using thermo-acoustic technology due to the no-moving-part design. Standing wave and travelling wave designs are being evaluated with support from Los Alamos Laboratories in the US. The presentation will concentrate on the stove requirements and needs of the developing world and how this affects the thermo-acoustic and mechanical design. Results from mathematical modelling and measurements from an early demonstrator will be presented. More information can be found at [Http://www.score.uk.com](http://www.score.uk.com). The £2M Score research project is funded by EPSRC, a UK government agency.

9:40

**4aPAa6. Recent developments on heat to electricity thermoacoustic conversion.** Guadalupe Huelsz (UNAM, Privada Xochicalco Temixco Centro, Temixco Morelos, 62580 Temixco, Mexico, ghl@cie.unam.mx), Miguel Piñeirua (UNAM, Privada Xochicalco Temixco Centro, Temixco Morelos, 62580 Temixco, Mexico, mipim@cie.unam.mx), Alfonso A. Castrejon-Pita (UNAM, Privada Xochicalco Temixco Centro, Temixco Morelos, 62580 Temixco, Mexico, aacp@atm.ox.ac.uk), Fabrisio Gomez (UNAM, Privada Xochicalco Temixco Centro, Temixco Morelos, 62580 Temixco, Mexico, flgog@cie.unam.mx)

In this work we present recent developments for the conversion of heat into electricity based on the combined effects of a thermoacoustic prime mover coupled to a magnetohydrodynamic generator where different working fluids can be optimally chosen for each process. We consider the acoustically produced oscillatory motion of a liquid drop confined into a horizontal squared cross section capillary tube as a possible flow configuration for the system. We investigated the energy losses of the system and concluded that this system would be a convenient configuration for small systems. [Work supported by CONACYT 25116 project]

10:00-10:20 Break

### *Contributed Paper*

10:20

**4aPAa7. Fundamental study of a loop-tube-type thermoacoustic cooling system using heat energy from condensed sunlight.** Shin-Ichi Sakamoto (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, ssakamot@mail.doshisha.ac.jp), Shintaro Komiya (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, bte1031@mail4.doshisha.ac.jp), Naoki Miya (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, dtg0144@mail4.doshisha.ac.jp), Jiro Senda (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, jsenda@mail.doshisha.ac.jp), Yoshiaki Watanabe (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, kwatanab@mail.doshisha.ac.jp)

The feasibility of implementation of a sunlight-based loop-tube-type thermoacoustic cooling system is investigated. Sunlight is condensed using a 0.60 m diameter Fresnel lens and irradiated to the high-temperature heat exchanger. Then, the temperature of heat exchanger is risen. This heat energy from condensed sunlight is applied to the driving heat energy of the loop-tube-type thermoacoustic cooling system. The total length of the loop tube is 3300 mm, and a gaseous mixture of He and Ar (50% / 50%) is used as the working fluid. A 50-mm-long ceramic honeycomb is used for the stack. The channel radius of the prime mover stack is 0.45 mm; that of the heat pump is 0.35 mm. The cooling point temperature falls from 29°C to -4.3°C before sunlight is irradiated to the high-temperature heat exchanger. Consequently, a temperature drop of 33.3°C is achieved using sunlight. Result obtained in this experiment underscores the feasibility of implementation of a sunlight-based loop-tube-type thermoacoustic cooling system.

10:40

**4aPAa8. Geometry effects and scaling in thermoacoustics.** Jos Zeegers (Eindhoven University of Technology, Den Dolech 2, 5612 AZ Eindhoven, Netherlands, j.c.h.zeegers@tue.nl)

Current work at TU Eindhoven on thermoacoustics will be discussed. The end effects of the geometry of a stack on the performance of thermoacoustic machines will be shown. End effects and the formation of vortices is an issue that contributes to enhanced convective losses at the stack ends. Influence of Reynolds and Strouhal numbers on the oscillatory flow field in the stack are studied. It is possible to plot  $Sr$  and  $Re$  number diagrams in which various zones can be identified that display characteristic flow patterns. Furthermore the influence of the type of regenerator material is studied in traveling wave engines. Performance of honeycomb material of high-density pores is compared with wire screen regenerators in a thermoacoustic motor. As a last point the effects of downscaling to miniature systems is discussed. Limits of how far the size of a thermoacoustic cooler can be downscaled before conduction effects limit the performance are relevant.

### Contributed Papers

11:00

**4aPAa9. Suppression of harmonics in a high frequency standing-wave thermoacoustic engine.** Wei Dai (Technical Institute of Physics and Chemistry, CAS, Beiyitiao Rd., Zhongguancun St., P.O.Box 2711, 100080 Beijing, China, dwpeng@yahoo.com), Bo Yu (Technical Institute of Physics and Chemistry, Chinese Academy of Sciences, Zhongguancun BeiYiTiao 2, Hai Dian, 100080 Beijing, China, yubo@mail.ipc.ac.cn), Guoyao Yu (Technical Institute of Physics and Chemistry, Chinese Academy of Sciences, Zhongguancun BeiYiTiao 2, Hai Dian, 100080 Beijing, China, gyyao@cl.cryo.ac.cn), Ercang Luo (Technical Institute of Physics and Chemistry, CAS, Beiyitiao Rd., Zhongguancun St., P.O.Box 2711, 100080 Beijing, China, Ecluo@cl.cryo.ac.cn)

A thermoacoustic engine converts heat into acoustic power and could be used to driven a cooler for refrigeration purposes or to drive a piston for electric power generation. Due to non-linear effects inside the system, higher order harmonics could be generated which may deteriorate the thermal performance of the whole system. In this report, a 500Hz standing wave thermoacoustic engine has been built. The occurrence of higher order acoustic oscillations has been closely observed. A series of experiments has been done to investigate the influence of resonance tube configuration on this phenomenon. The influence on the related thermal performance is also reported.

11:20

**4aPAa10. Energy conversion efficiency improvement of a thermoacoustic cooling system - The influence of a lamination mesh on cooling effect-.** Atsushi Sakaguchi (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, dth0165@mail4.doshisha.ac.jp), Shin-Ichi Sakamoto (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, ssakamot@mail.doshisha.ac.jp), Yoshiyuki Tsuji (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, dtg0925@mail4.doshisha.ac.jp), Yoshiaki Watanabe (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, kwatanab@mail.doshisha.ac.jp)

It was generally considered that energy conversion efficiency of a thermoacoustic cooling system was improved by increasing proportion of a thermal boundary layer in the stack. Thinning down a channel radius of the stack is required to increase the thermal boundary layer per unit area. A ceramic stack is difficult to satisfy this requirement. To satisfy this requirement, we propose to use a lamination mesh which is formed by piled up a stainless mesh. Since the lamination mesh has complex channel, the effect of changing the mesh number on the cooling effect is obscure. The experiments were carried out by changing the mesh number and the insertion position of the stack using the straight acoustic tube to measure the cooling effect. From the experimental results, changing mesh number causes the distribution shift of the phase difference between pressure and particle velocity. The insertion position of the stack for which the maximum temperature decrease come close to center of the acoustic tube. This is corresponded to a

result using the ceramic stack. It suggests that a same design method, which is used in case of applying the ceramic stack, can be adapted to the lamination mesh.

11:40

**4aPAa11. Scalability of Thermoacoustic Refrigerator.** Ehab Abdel-Rahman (The American University, 211 Science Building, 113 Kasr El Aini St., P.O. Box 2511, 11511 Cairo, Egypt, ehab\_ab@aucegypt.edu)

The uses of thermoacoustic effect are the conversion of heat onto acoustic wave by thermoacoustic prime mover and pumping heat by acoustic wave using a thermoacoustic refrigerator. Thermoacoustic refrigerator is a good alternative to conventional vapor compression device. It can be very compact, using a minimum of uncomplicated, economical components. It can also provide variable cooling capacity and scalability to different sizes in ways that conventional vapor compression technology is not capable of. We are investigating the minimum size of a thermoacoustic refrigerator that can meet the cooling requirement for different applications. The coefficient of performance (COP) of small thermoacoustic refrigerators is in the range of 20% of the ideal (Carnot cycle) COP, which is actually better than similarly scaled vapor compression coolers which is at about 10%. The efficiency of thermoacoustic refrigerator can be improved by better designing of its components. In this paper the effect of components design on the performance of thermoacoustic refrigerator is discussed. The scalability of such devices is also investigated.

12:00

**4aPAa12. Advanced flow measurements in thermoacoustic systems.** Arganthaël Berson (Ecole Centrale de Lyon, LMFA, UMR CNRS 5509, Ecully, 69134 Lyon, France, arganthaël.berson@ec-lyon.fr), Philippe Blanc-Benon (Ecole Centrale de Lyon, LMFA, UMR CNRS 5509, Ecully, 69134 Lyon, France, Philippe.Blanc-Benon@ec-lyon.fr)

One of the bottlenecks for the conception of highly efficient thermoacoustic systems is the poor performances of heat exchangers. A better understanding of both aerodynamical and thermal phenomena that occur at the interface between the stack and the heat exchangers is necessary for the improvement of heat transport between these components, especially at high acoustic amplitudes that are required for industrial applications. To this end, a specific Particle Image Velocimetry (PIV) method has been developed and measurements are performed within a standing-wave thermoacoustic refrigerator model driven at high drive ratios (up to 5%). Vortex streets are observed behind the plates of a single stack at high acoustic level. The flow is characterized using advanced vortex analysis tools and dimensionless numbers. Vortices also appear within the gap between the stack and the heat exchangers. They will influence heat transport as was previously shown in numerical simulations from the literature. Moreover, as a first step toward enthalpy flux measurements, cold and hot wire anemometry methods are developed specifically for temperature and velocity measurements in oscillating flows. First results with these techniques will be presented and compared to available numerical and analytical models. This work is supported by ANR (project MicroThermAc NT051\_42101).



## Invited Paper

12:20

**4aPAa13. Powerful, efficient, robust, electro-acoustic transducers.** John Corey (CFIC-Qdrive, 302 Tenth St., Troy, NY 12180, USA, jcorey@cficinc.com)

The STAR™ resonant, reciprocating transducer began as a lightweight linear alternator design for a space-power free-piston Stirling engine in the early 1990's. It has since been developed into a range of commercially available motors and alternators with rated powers from 100 to over 10,000 watts (acoustic). As motors, these are acoustic pressure drivers with unlimited operating life and typical transduction efficiencies of 80-90 percent. This paper explains the electrodynamics and operation of these moving-magnet Lorentz-force devices and the unique geometric configuration that has allowed scaling over such a wide range. We discuss the design and function of the unique single-degree-of-freedom flexure suspension that enables both the compact geometry and unlimited service life without wear. Data is presented from a large sample of units placed in service during the last decade, demonstrating the durability and performance of these remarkable devices.

12:40-1:40 Lunch Break

## Invited Paper

1:40

**4aPAa14. Recent developments in miniaturization of thermoacoustic devices.** Pierrick Lotton (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, Pierrick.Lotton@univ-lemans.fr), Guillaume Penelet (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, guillaume.penelet@univ-lemans.fr), Etienne Gaviot (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, etienne.gaviot@univ-lemans.fr), Stephane Durand (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, stephane.durand@univ-lemans.fr), Lionel Camberlein (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, lionel.camberlein@univ-lemans.fr), Philippe Blanc-Benon (Ecole Centrale de Lyon, LMFA, UMR CNRS 5509, Ecully, 69134 Lyon, France, Philippe.Blanc-Benon@ec-lyon.fr), Michel Bruneau (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, michel.bruneau@univ-lemans.fr)

This talk will present a review of recent works on thermoacoustic devices miniaturization conducted at Laboratoire d'Acoustique de l'Université du Maine (LAUM), in collaboration with Laboratoire de Mécanique de Fluides et d'Acoustique (LMFA). A part of these works deals with new designs allowing higher compactness of devices. As an example, a compact non-resonant thermoacoustic refrigerator will be described, and experimental results obtained on a prototype will be presented. Another part of these works deals with the miniaturization of each element of a thermoacoustic device, especially the stack and the acoustic source. The acoustical and thermal sensors used to control these small devices have also to be miniaturized. Some specific actuators and sensors designed for miniaturized thermoacoustic refrigerators will be presented. Finally, potential applications of these miniaturized devices will be discussed.

## Contributed Papers

2:00

**4aPAa15. Miniaturization of thermoacoustic refrigerators.** Yan Li (Eindhoven Univ. of Tech, Applied Physics, Low Temperature Physics, Cascade 3.13, PO Box 513, 5600MB Eindhoven, Netherlands, yan.li@tue.nl)

The possibility to miniaturize thermoacoustic refrigerators is theoretically investigated. Both standing-wave and traveling-wave systems are considered. In the consideration of standing-wave refrigerators, a system consisting of a resonator tube (50 cm) with a closed end and a PVC stack (length 5 cm) is taken as a reference. Helium is used at a mean pressure of 10 bars and an amplitude of 1 bar. The operating frequency is 1 kHz. The variation of the performance of the refrigerator when scaled down in size is computed under the prerequisites that the temperature difference over the stack or the energy flux or energy flux density are fixed. The analytical results show, as expected, that there is a limitation for scaling-down of a standing-wave thermoacoustic refrigerator due to heat conduction. Similar scaling trends are also shown in traveling-wave refrigerators. The traveling-wave reference system consists of a feedback inertance tube of 0.567 m long, inside diameter 78 mm, a compliance volume of 2830 cm<sup>3</sup> and a 24 cm thermal buffer tube. The regenerator is sandwiched between two heat exchangers. The system is operated at 125 Hz and filled with 30 bar helium

gas. Again, the thermal conductance forms a practical limitation in down-scaling.

2:20

**4aPAa16. Nonlinear effects in standing-wave types of thermoacoustic devices.** Paul Aben (Eindhoven University of Technology, Den Dolech 2, 5612 AZ Eindhoven, Netherlands, p.c.h.aben@tue.nl), Jos Zeegers (Eindhoven University of Technology, Den Dolech 2, 5612 AZ Eindhoven, Netherlands, j.c.h.zeegers@tue.nl)

In order to create high amplitudes ( $p'/p_0 > 10\%$ ) for a relatively low frequency range (between 5 and 125 Hz), a large subwoofer is connected to a resonator tube by an exponential horn. A parallel-plate stack, with various plate thicknesses and separations, can be placed at different positions in the resonator tube. The position of the subwoofer membrane, the voltage and current of the subwoofer, as well as the pressure at six different positions in the resonator is measured. The measurements are in good agreement with simulations. Using a multi-microphone method the transfer matrix of a stack is determined experimentally. Using a PIV method a 2-D velocity field between and around the stack plates is measured. The vortex shedding at the end of stack plates is studied in particular. The amplitude, frequency, plate thickness, plate separation and plate-ending shape are varied. Also the streaming velocity field is studied. Small asymmetries in the geometry have a huge influence on the streaming velocity.

2:40

**4aPaa17. Intensity measurement of a periodic acoustic shock wave in a resonator.** Tetsushi Biwa (Dept. of Mechanical Systems and Design, Tohoku Univ., Aramaki 6-6-01, Aoba-ku, 980-8579 Sendai, Japan, biwa@amsd.mech.tohoku.ac.jp), Taichi Yazaki (Aichi Univ. Education, Igaya, 448-8542 Kariya, Japan, tyazaki@aeu.ac.jp)

A periodic shock wave of a gas column is formed in a duct, when the gas column is sinusoidally driven near the resonance frequency. This phenomenon has been one of the fundamental problems in nonlinear acoustics and has been studied extensively both theoretically and experimentally. In this work, we study the nonlinear effect leading to the shock formation through measurements of the acoustic intensity. A gas column of atmospheric air is filled in a resonator with a length of 1.15 m and internal radius of 10.5mm, and driven by an oscillating piston at 144.4 Hz near the fundamental resonance frequency. Pressure and axial acoustic particle velocity of the gas column are measured as a function of the resonator axis using small pressure transducers and a laser Doppler velocimeter. We show the spatial distribution of the acoustic intensity associated with the fundamental and the second modes, from which we show the nonlinear interaction between these oscillating modes.

3:00

**4aPaa18. Transition to turbulence and acoustic Rayleigh streaming in thermoacoustic devices.** Helene Baillet (Laboratoire d'Etudes Aérodynamiques (LEA), Université de Poitiers, ENSMA, CNRS, Bat K, 40 avenue du recteur Pineau, 86022 Poitiers, France, helene.baillet@lea.univ-poitiers.fr), Jean-Christophe Valière (Laboratoire d'Etudes Aérodynamiques (LEA), Université de Poitiers, ENSMA, CNRS, Bat K, 40 avenue du recteur Pineau, 86022 Poitiers, France, jean-christophe.valiere@lea.univ-poitiers.fr), Solenn Moreau (Laboratoire d'Etudes Aérodynamiques (LEA), Université de Poitiers, ENSMA, CNRS, Bat K, 40 avenue du recteur Pineau, 86022 Poitiers, France, solenn.moreau@lea.univ-poitiers.fr), David Marx (Laboratoire d'Etudes Aérodynamiques - CNRS, Bat K, 40 avenue du recteur Pineau, 86022 Poitiers, France, david.marx@lea.univ-poitiers.fr)

Thermoacoustic engines and refrigerators with practical levels of heating or pumping power must generally operate at high pressure amplitudes. When used to describe the behavior of such high-amplitude thermoacoustic devices, the well-established foundations of thermoacoustics, based on the acoustic approximation, reach their limits. It is necessary to gain a deeper understanding of the high-amplitude phenomena in order to improve the performances of thermoacoustic devices, and efforts of several research groups have been directed towards this goal over the last decade. In this presentation, we will consider recent advances in the understanding of some of the gas-dynamics phenomena leading to limitation of devices performances, namely transition to turbulence and acoustic Rayleigh streaming. The common point for these phenomena is that they owe their origin in the dynamic of oscillating flows in very near wall regions, so that their quantification implies measurements of acoustic particle velocity in adverse conditions. Recent progresses in Laser techniques used to perform such measurements will therefore also be reviewed.

3:20

**4aPaa19. PIV contribution for measuring acoustic and streaming flow in thermoacoustic systems, using phase average dynamics.** Diana Baltean Carlès (LIMS-CNRS, BP 133, F-91403 Orsay, France, baltean@limsi.fr), Philippe Debesse (LIMS-CNRS, BP 133, F-91403 Orsay, France, debesse@limsi.fr), François Lusseyran (LIMS-CNRS, BP 133, F-91403 Orsay, France, francois.lusseyran@limsi.fr), Maurice-Xavier François (LIMS-CNRS, BP 133, F-91403 Orsay, France, mxf@limsi.fr)

The present study deals with the experimental challenge of the measurement of the velocity field generated by a thermoacoustic wave. The system consists in a cylindrical standing-wave resonator, filled with gas confined at high mean pressure, driven by a thermoacoustic prime-mover. The axial and radial components of fluid velocity are measured using Particle Image Velocimetry (PIV) with an optical flow technique. The average cycle of acoustic oscillation of the velocity field is reconstructed from a temporally under-sampled set of PIV snapshots, using an embedding method for building out a suitable phase space based on Singular Value Decomposition (SVD). This reconstruction allows us to extract both oscillation component of the velocity field (with the harmonic content) and time-averaged component of velocity (streaming flow). The measurements are confirmed using a second experimental procedure, based on a classical phase-averaged method: velocity measurements are synchronized with the pressure signal, the fundamental time period being decomposed in 16 phases. The measurements are repeated for different values of the drive ratio (acoustic pressure/mean pressure). The results are compared with available theory. Different experimental methods used in measuring the velocity field in thermoacoustic systems are analysed and compared with the present method.

3:40-5:00 Posters

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

## Invited Papers

5:00

**4aPAa20. Interactive analysis, design, and teaching for thermoacoustics using DeltaEC.** William C. Ward (Los Alamos National Laboratory, MS C914, Los Alamos, NM 87545, USA, ww@lanl.gov), Greg W. Swift (Los Alamos National Laboratory, MS C914, Los Alamos, NM 87545, USA, swift@lanl.gov), John P. Clark (Los Alamos National Laboratory, MS C914, Los Alamos, NM 87545, USA, jpclark@stanford.edu)

The 2008 release of the Los Alamos thermoacoustics code, DeltaEC, is distinctly different from the text-based program that was first made available in 1993. The physics captured by "Design Environment for Low-Amplitude Thermoacoustic Energy Conversion" has been steadily extended over the years. Toroids and other acoustic network topologies are now possible, along with superimposed steady flow, time-averaged pressure gradients, gas diodes, Gedeon streaming, thermoacoustic mixture separation, and resonator vibration solutions. Eight different stack pore geometries are supported, and a powerful algebraic user language allows complex, custom results to be derived without source code revisions at Los Alamos. Over the last year, the numerical methods of DeltaEC were condensed into a FORTRAN computational core and wrapped with a Python-based graphical user interface to provide modern interactive features: a multi-model tabbed interface, colorizing editors, scaled schematics, and 2D plotter windows. An intricate model can now be divided into interlinked sub-models that can be solved independently (and consistently). In addition to providing usability and new capabilities, the Python front end makes the legacy code more maintainable, extensible, and verifiable. The latest download of DeltaEC is available to all researchers at [www.lanl.gov/thermoacoustics/](http://www.lanl.gov/thermoacoustics/).

5:20

**4aPAa21. Design road-map for thermoacoustic refrigerators.** Kaveh Ghorbanian (Sharif Univ. of Technology, Dept. of Aerospace Engineering, Azadi Street, 14588-89694 Tehran, Iran, ghorbanian@sharif.edu), Hemed Hosseini (Sharif Univ. of Technology, Dept. of Aerospace Engineering, Azadi Street, 14588-89694 Tehran, Iran, hamed\_hosseini20042000@yahoo.com), Mahmoud Jafargholi (Sharif Univ. of Technology, Dept. of Aerospace Engineering, Azadi Street, 14588-89694 Tehran, Iran, mahmoud.jafargholi@gmail.com)

The main purpose of this work is to develop a road-map to enhance the design procedure and performance analysis of thermoacoustic refrigerators (TAR). The basic mechanism of TAR is very simple and is based on the wave interaction processes of gas particles with their surrounding environment. As known, the performance of a TAR system is highly dependent to the number of gas particles involved in the process. The essential components of TAR include a sound generating device, a resonance tube, a stack of plates, and heat exchangers. In this paper, the parameters influencing the performance of a TAR system are grouped into four blocks: (I) Operation (drive ratio, operating frequency), (II) Fluid (Prandtl number), (III) Geometry (blockage ratio, tube diameter, stack length, stack positioning), and (IV) Material. A parametric study is executed to determine the optimized design of a TAR system for fixed Block II and IV scenario. First, an analysis is made on obtaining the optimum number of gas particles to be involved in the process. Then, based on this, an optimization approach is carried out to identify the best drive ratio, blockage ratio, stack positioning, stack length, and resonance tube diameter. Finally, results are compared with experimental data.

## Contributed Paper

5:40

**4aPAa22. Thermoacoustics in random fibrous materials.** Carl R. Jensen (The University of Mississippi - NCPA, 1 Coliseum Drive, University, MS 38677, USA, crjensen@olemiss.edu), Richard Raspet (National Center for Physical Acoustics, University of Mississippi, University, MS 38677, USA, raspet@olemiss.edu), Henry E. Bass (The University of Mississippi - NCPA, 1 Coliseum Drive, University, MS 38677, USA, pbass@olemiss.edu)

Current approaches to acoustics in fibrous and porous materials use fitting parameters to match theoretical models to measured values for the material's complex compressibility and wavenumber. In effect, these models

treat the material as though it were composed of an array of rigid capillary tubes; they have proven accurate in fitting the model to data for various different porous materials such as wools and foams. However, these models do not address thermoacoustic heat transfer when the material is put under a static temperature gradient. A direct simulation has been performed using a three-dimensional thermal fluid solver to calculate both the acoustic properties and the thermoacoustic properties of a random fibrous material. The results of the simulation will be compared to experimental results for complex compressibility and wavenumber [Tarnow, H., J. Acoust. Soc. Am., 97(4), 2272-81] as well as a proposed extension to porous theory that incorporates thermoacoustics [Roh et al., J. Acoust. Soc. Am., 121(3), 1413-22]. [Work supported by U.S. Army Space & Missile Defense Command.]

## Invited Paper

6:00

**4aPAa23. Numerical study of the performance of thermally isolated thermoacoustic-stacks in the linear regime.** Antonio Piccolo (Dept. of Civil Engineering - Univ. of Messina, Contrada di Dio, Villaggio S.Agata, 98100 Messina, Italy, apiccolo@unime.it), Giuseppe Pistone (Dept. of Matter Physics and Advanced Physical Technologies - Univ. of Messina, Contrada di Dio, Villaggio S.Agata, 98100 Messina, Italy, pistone@unime.it)

A simplified calculus model to investigate on the transverse heat transport near the edges of a thermally isolated thermoacoustic stack in the low acoustic Mach number regime is presented. The proposed methodology relies on the well known results of the classical linear thermoacoustic theory which are implemented into an energy balance calculus-scheme through a finite difference technique. Details of the time-averaged temperature and heat flux density distributions along a pore cross-section of the stack are given. It is shown that a net heat exchange between the fluid and the solid walls takes place only near the edges of the stack plates, at distances from the ends not exceeding the peak-to-peak particle displacement amplitude. The structure of the mean temperature field within a stack plate is also investigated; this last results not uniform near its terminations giving rise to a smaller temperature difference between the plate

extremities than that predicted by the standard linear theory. This result, when compared with experimental measurements available in literature, suggests that thermal effects localized at the stack edges may play an important role as sources of the deviations found between linear theory predictions and experiments at low and moderate Mach numbers.

### Contributed Paper

6:20

**4aPAa24. Time-domain modelling of thermoacoustic devices: Reflections from the stack.** Stig Kleiven (Chalmers University of Technology, Division of Applied Acoustics, SE-41296 Gothenburg, Sweden, stig.kleiven@chalmers.se), Krister Larsson (Chalmers University of Technology, Division of Applied Acoustics, SE-41296 Gothenburg, Sweden, kl@ta.chalmers.se), Wolfgang Kropp (Chalmers University of Technology, Division of Applied Acoustics, SE-41296 Gothenburg, Sweden, wolfgang.kropp@chalmers.se)

Thermoacoustic devices are today mainly simulated using frequency-domain models. Contrary to frequency-domain models, time-domain models may include time-varying boundary conditions and time varying effects. Furthermore, effects like reflections from the stack can be visualised which,

in turn, can improve the understanding. The primary aims were: (1) to study the initial behaviour of the acoustic field in a thermoacoustic device using a time-domain method; and (2) to compare the results from the simulations with experiments. Since time-domain techniques are computational expensive, the detailed Finite-Difference Time-Domain (FDTD) method are combined with the quicker Equivalent Source Method (ESM). The acoustic field in the stack, only including viscous effects, was modelled using the FDTD method, and the ESM was used outside the stack. The experimental setup consisted of a loudspeaker connected to a circular tube containing a ceramic stack with rectangular pores. From the results of both the simulations and the experiments, clear reflections from the stack were seen. Since these reflections influence the total acoustic field in thermoacoustics devices, time-domain methods provide useful tools for further development of thermoacoustic devices.

### Invited Paper

6:40

**4aPAa25. Cooling load and coefficient of performance of thermoacoustic refrigerators: the role of the working fluid.** Cila Herman (Johns Hopkins University, 3400 N. Charles Street, Baltimore, MD 21218, USA, cherman.jhu@gmail.com)

Thermoacoustic refrigeration is a technology that uses mechanical energy in the form of sound waves to drive a heat pumping process that offers an environmentally safe, relatively low maintenance alternative to vapor compression refrigeration. Improving the design of the thermoacoustic core, composed of the stack plates and heat exchangers, may have the potential to bring thermoacoustic technology closer to commercial use. The stack plates have been analyzed to increase efficiency (expressed in terms of the coefficient of performance, COP) but in some applications a very high efficiency can lead to a smaller cooling load. The thermoacoustic stack was analytically optimized for maximum heat transfer (cooling load) and coefficient of performance (COP) for a range of working fluids of interest. Different noble gas mixtures were analyzed as the working fluid and helium was found to produce the highest cooling load because of its low molecular weight. The thermoacoustic stack plate center location, length, thickness and spacing were analyzed and optimum values to maximize cooling load and COP were found to exist for the specific input parameters considered. These optimization techniques may be used to design devices where maximum cooling load is more desirable than high efficiency.

### Contributed Papers

7:00

**4aPAa26. van der Waal gaz and direct simulation for thermoacoustics.** Alain Fontaine (Université Paul Sabatier, PHASE, 118, route de Narbonne, 31062 Toulouse cedex 9, France, fontaine@cict.fr), Marie-Catherine Mojtabi (Univ. Paul Sabatier, PHASE, 118 route de Narbonne, 31062 Toulouse Cedex 9, France, cmojtabi@cict.fr), Abdelkader Mojtabi (IMFT, Al-lée du Professeur Camille Soula, 31400 Toulouse, France, mojtabi@cict.fr), Jean-Louis Breton (Univ. Paul Sabatier, PHASE, 118 route de Narbonne, 31062 Toulouse Cedex 9, France, breton@cict.fr), Vincent Gibiat (Université Paul Sabatier, PHASE, 118, route de Narbonne, 31062 Toulouse cedex 9, France, gibiat@cict.fr)

Thermoacoustic refrigerators work with high amplitude acoustic waves and lead to high thermal local gradient near the stack. In order to understand nonlinear thermoacoustic effects, an acoustic plane wave is propagated in a model with a specific geometry. It is a two-dimensional channel with adiabatic walls, including two conductive plates whose thickness is not regarded as null. The fluid is supposed to be a real gas with thermodynamic properties described by a van der Waals law. A two-dimensional direct numerical model for compressible flow is used to investigate unsteady dynamic and energetic behaviours in the channel. This model relies on a finite volume formulation of the mass, momentum and energy equations for compressible flow. Thermal equilibrium between gas and plates is assured by a Dirichlet boundary condition for the interface temperature and a thermal conduction equation in the plate. Adiabatic walls are described by a Neumann boundary condition for the temperature on the wall. The acoustic wave is generated at the input. Perfectly Matched Layer (PML) is used on input and output as a

boundary condition to deal with wave reflections. Numerical investigation will be presented and compared with linear classical model.

7:20

**4aPAa27. Modeling and simulation of Taconis oscillations in a framework of the boundary-layer theory.** Nobumasa Sugimoto (Osaka University, Department of Mechanical Science, Graduate School of Engineering Science, Toyonaka, 560-8531 Osaka, Japan, sugimoto@me.es.osaka-u.ac.jp), Dai Shimizu (Osaka University, Department of Mechanical Science, Graduate School of Engineering Science, Toyonaka, 560-8531 Osaka, Japan, shimizu@mars.me.es.osaka-u.ac.jp)

This paper develops a simplified one-dimensional model to simulate numerically the onset of Taconis oscillation in a helium-filled, quarter-wavelength tube in cryogenics. Introducing a boundary layer on the tube wall, nonlinear fluid-dynamical equations are averaged over the cross-section of the acoustic main-flow region outside of the boundary layer. The boundary layer gives rise to memory effects, which are taken into account in the form of half-order derivatives. For a smooth temperature distribution of the tube wall, an initial and boundary-value problem of the equations derived is solved numerically for evolution of a small disturbance. The boundary condition at the open end neglects radiation and requires the excess pressure to vanish, while the condition at the closed end takes account of the boundary layer. When the ratio of the temperature at the closed end to the one at the open end exceeds a critical value, the initial helium column becomes unstable to grow in amplitude and stationary self-excited Taconis os-



cillations of finite amplitude emerge. It is shown that even the first-order boundary-layer theory can describe such an evolution of a small unstable disturbance into self-excited oscillations.

7:40

**4aPa28. Numerical model of boundary layer effects in Rayleigh streaming.** Brian Tuttle (Penn State, 412 W Foster Ave, State College, PA 16801, USA, btuttle@psu.edu), Victor W. Sparrow (Penn State, Graduate Program in Acoustics, 201 Applied Science Bldg., University Park, PA 16802, USA, vws1@psu.edu)

In thermoacoustic devices such as the pulse-tube refrigerator, efficiency is diminished by the formation of a second-order mean velocity known as

Rayleigh streaming. This flow emerges from the interaction of the working gas with the wall of the tube in a thin boundary layer. This research develops a numerical model to investigate Rayleigh streaming in straight and tapered tubes. Since the accuracy of the model depends on the correct representation of boundary layer effects, special consideration is given to the computation of thermal and viscous boundary layers including finite difference methods, computational grid refinement, and exaggeration of physical parameters for testing of boundary layers at low grid resolution. The model also allows for the inclusion or exclusion of temperature-dependent viscosity and thermal conductivity terms, the effects of which will be examined. [Work supported in part by the Office of Naval Research.]

THURSDAY MORNING, 3 JULY 2008

ROOM 351, 8:00 A.M. TO 7:40 P.M.

## Session 4aPAb

### Physical Acoustics: Ultrafast Acoustics I

David H. Hurley, Cochair

*Materials Characterization Department, Idaho National Laboratory, P.O.Box 1625, Idaho Falls, ID 83415-2209, USA*

Clément Rossignol, Cochair

*LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, Talence, 33405, France*

#### Invited Paper

8:00

**4aPAb1. Coherent phonons in semiconductor superlattice under DC electrical bias.** Anthony Kent (University of Nottingham, School of Physics and Astronomy, University Park, NG9 3JE Nottingham, UK, anthony.kent@nottingham.ac.uk), Ryan Beardsley (University of Nottingham, School of Physics and Astronomy, University Park, NG9 3JE Nottingham, UK, ppxrb1@nottingham.ac.uk), Andrey Akimov (University of Nottingham, School of Physics and Astronomy, University Park, NG9 3JE Nottingham, UK, andrey.akimov@nottingham.ac.uk), Mohamed Henini (University of Nottingham, School of Physics and Astronomy, University Park, NG9 3JE Nottingham, UK, mohamed.henini@nottingham.ac.uk)

Resonant pumping of superlattices SLs by femtosecond laser pulses results in the generation of coherent phonons with frequency centred approximately on  $s/d_{\text{SL}}$ , where  $s$  is the speed of longitudinal sound and  $d_{\text{SL}}$  is the SL period. The phonons can be detected by measuring the changes in reflectance of time-delayed probe pulses. To date, measurements have been made on nominally undoped SLs, and phonon frequencies are typically in the range 100 GHz - 1 THz. The motivation for these studies is that such hypersound could be used for acoustic probing of nanostructures. Here we describe the generation and detection of coherent phonons in a doped and electrically-biased SL. The studied SL consisted of  $50 \times (6 \text{ nm GaAs and } 4 \text{ nm AlAs})$ , uniformly doped with Silicon to density  $2 \times 10^{22} \text{ m}^{-3}$ . Pump-probe measurements were made at  $\lambda \sim 770 \text{ nm}$  on an optical MESA at  $T = 12 \text{ K}$ . At zero bias, we observed a similar phonon spectrum as previously observed in comparable but undoped SLs, with a mode at  $\sim 450 \text{ GHz}$ . Under applied bias this mode increased in amplitude, and the decay time was also increased. We discuss these observations in terms of the effects of the bias on the coherent phonon generation and detection process and also the possibility of coherent phonon amplification occurring in the structure.

#### Contributed Papers

8:20

**4aPAb2. Acoustic Bloch oscillations, Wannier-Stark ladders and negative refraction in ultra- and hypersonic superlattices.** Yuriy A. Kosevich (Semenov Institute of Chemical Physics Russian Academy of Sciences, ul. Kosygina 4, 119991 Moscow, Russian Federation, yukosevich@yahoo.com)

A new and very efficient ultrasonic superlattice is realized for the study of such fundamental effects of quantum transport in a perturbed periodic potential as acoustic Bloch oscillations, Wannier-Stark ladders and resonant Landau-Zener tunneling. The acoustic equivalent of the Wannier-Stark ladders is employed in a set of water cavities, with a gradient of the thick-

nesses, in a simple water-solid multilayer system. Bloch oscillations in different acoustic minibands are observed as time-resolved oscillations of the transmission of ultrasonic pulses with corresponding spectral positions and widths. Acoustic Bloch oscillations with different temporal periods for the pulses centered in two neighboring acoustic minibands are observed. Experimental observations are in very good agreement with the transfer-matrix simulations [1]. The propagation of acoustic pulse in an even acoustic miniband in orthogonal or oblique with respect to the superlattice axis direction will result in negative refraction and Bloch oscillations of acoustic pulse with "negative effective mass", which can be visualized by mapping out of acoustic pressure field. Hypersonic phononic structures, which can be fabricated with the use of interference lithography, render possible the obser-

vation of all the abovementioned acoustic phenomena in the hypersound frequency range. I. H. Sanchis-Alepuz, Yu. A. Kosevich, and J. Sanchez-Dehesa, 2007, Phys. Rev. Lett. v. 98, 134301.

8:40

**4aPAb3. Strain wave induced electron transport in superlattices.**

Anthony Kent (University of Nottingham, School of Physics and Astronomy, University Park, NG9 3JE Nottingham, UK, anthony.kent@nottingham.ac.uk), Daivid Fowler (University of Nottingham, School of Physics and Astronomy, University Park, NG9 3JE Nottingham, UK, daivid.fowler@nottingham.ac.uk), Mohamed Henini (University of Nottingham, School of Physics and Astronomy, University Park, NG9 3JE Nottingham, UK, mohamed.henini@nottingham.ac.uk), Mark Greenaway (University of Nottingham, School of Physics and Astronomy, University Park, NG9 3JE Nottingham, UK, ppxmg@nottingham.ac.uk), Alexander Belanov (University of Nottingham, School of Physics and Astronomy, University Park, NG9 3JE Nottingham, UK, alexander.belanov@nottingham.ac.uk), Mark Fromhold (University of Nottingham, School of Physics and Astronomy, University Park, NG9 3JE Nottingham, UK, mark.fromhold@nottingham.ac.uk)

We show that propagating high-amplitude coherent strain pulses, generated by ultrafast optical excitation of a metal film can induce a charge current in a GaAs/AlAs superlattice (SL). The studied SL had a period of 12.5 nm and a miniband width of 12 meV. It was grown by MBE on a 0.35 mm-thick semi-insulating GaAs substrate, and a 100 micron device MESA fabricated. On the other side of the substrate a 100 nm-thick Al film was deposited. A coherent picosecond strain pulse was generated opposite the device by exciting the Al film with 40 fs, 800 nm pulses from a 5 KHz, 2.5 mJ Ti:Sapphire amplifier. A strong current pulse from the device was observed about 80 ns after the laser pulse was incident on the Al film, this time delay being equal to the time of flight of longitudinal polarized strain pulses across the GaAs substrate. We attribute the current pulse to electrons that are confined and dragged along by the potential generated by the strain wave. Theoretical calculations show that this "wave dragging" effect in the presence of the SL potential can give rise to the generation of ultra-high (THz) frequency electron dynamics.

9:00

**4aPAb4. Thickness dependence of the acoustical response of ultra-thin metallic films studied by Colored Picosecond Ultrasonics.**

Arnaud Le Louarn (IEMN-CNRS, Cité Scientifique - Avenue Poincaré, BP 60069, 59652 Villeneuve d'Ascq Cedex, France, arnaud.lelouarn@isen.fr), Arnaud Devos (IEMN-CNRS, Cité Scientifique - Avenue Poincaré, BP 60069, 59652 Villeneuve d'Ascq Cedex, France, arnaud.devos@isen.fr), Clément Rossignol (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, c.rossignol@lmp.u-bordeaux1.fr)

We have previously demonstrated [1,2] that a connection exists between Colored Picosecond Ultrasonic (CPU) experiments and electronic structure of metallic thin films. Indeed, a strong change of the detected acoustic echoes is observed when the laser is tuned around an interband transition wavelength. This connection suggests that CPU can be a useful tool for measuring interband transitions in thin metallic films. Surprisingly, by doing such a measurement on a series of ultra-thin Aluminium films, we observed a significant shift of the transition (from 880 to 970 nm) as the film thickness is reduced (from 400 to 120 Å). We will discuss the origin of the phenomenon and propose some applications to the characterization of ultra-thin metallic films. [1] A. Devos and C. Lerouge, Phys. Rev. Lett. 86, 2669 (2001) [2] A. Devos and A. Le Louarn, Phys. Rev. B 68, 045405 (2003)

9:20

**4aPAb5. Monochromatic high frequency coherent phonons propagation with superlattice transducers.** Agnès Huynh (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, agnes.huynh@insp.jussieu.fr), Maria Florencia Pascual Winter (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, Florencia.PascualWinter@insp.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), Bernard Jusserand (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, Bernard.Jusserand@insp.jussieu.fr), Aristide Lemaitre (LPN, CNRS route de Nozay, 91460 Marcoussis, France, Aristide.Lemaitre@lpn.cnrs.fr), Alejandro Fainstein (Centro Atómico Bariloche & Instituto Balseiro, C. N. E. A., 8400 S. C. de Bariloche, Argentina, afains@cab.cnea.gov.ar)

The availability of efficient and compact phonons transducers in the THz range would be very interesting for phonons spectroscopy, acoustic microscopy and study of vibrational and electronic properties of nanostructures. Thanks to epitaxial growth of semiconductors multilayers, high quality phononic nanostructures with standard semiconductors, such as superlattices (SL) and nanocavities can be obtained for the GHz and THz transduction. Picosecond ultrasonics experiments have been performed in transmission geometry with pump and probe incident on opposite sides of the substrate, allowing decoupling acoustic generation and detection processes. By these means, we have shown independently that SL are very efficient high frequency monochromatic phonon generators and detectors. We report on experiments where two superlattices have been grown on the opposite sides of a substrate: a first SL with uniform layer thickness over the whole surface sample is used as a generator; the other one, used as the detector, presents a thickness gradient and the location of the detection is chosen in order to have the best matching with the emitted frequency. This setup is used to study the propagation of monochromatic high frequency coherent phonons as a function of temperature.

9:40

**4aPAb6. Progress on coherent generation of terahertz acoustic phonons by resonant absorption of nanosecond-pulsed far-infrared laser radiation in silicon doping superlattices.**

Thomas E. Wilson (Marshall University, Department of Physics, One John Marshall Drive, Huntington, WV 25755-2570, USA, wilson@marshall.edu)

We report progress on experiments to generate and detect coherent terahertz acoustic phonons. Coherent phonons are generated<sup>1</sup> in a silicon-doping superlattice by the absorption of grating-coupled high peak-power cavity-dumped far-infrared laser radiation<sup>2</sup>. The superlattice period is chosen to match the phonon wavelength at the excitation frequency of the laser radiation. The phonons propagate across the Si:B substrate and are detected by a novel superconducting granular aluminium/palladium bilayer microbolometer<sup>3</sup> with sub-nanosecond resolution. The phonon spectrum is obtained by piezo-phonon spectroscopy<sup>4</sup> via the boron impurities in the silicon substrate. (1) P. Ruden and G.H. Dohler, "Anisotropy Effects and Optical Excitation of Acoustic Phonons in n-i-p-i Doping Superlattices", Solid State Commun. 45 (1), 23 (1983). (2) T.E. Wilson, "A High-Power NH<sub>3</sub> Laser Pumped in a Three-Mirror CO<sub>2</sub> Laser Cavity with Optically-Switched Cavity-Dumping", International Journal of Infrared and Millimeter Waves 14 (2), 303 (1993). (3) T E Wilson, "Fabrication and characterization of granular aluminum/palladium bilayer microbolometer", Meas. Sci. Technol. 18 N53-N59 (2007). (4) S. Roshko and W. Dietsche, "Phonon Spectroscopy in High Magnetic Fields: The B<sup>+</sup> Center in Si", Solid State Comm. 98(5), 453 (1996).

10:00-10:20 Break

10:20

**4aPAb7. Escape time of an acoustic nanocavity mode.**

Maria Florencia Pascual Winter (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, Florencia.PascualWinter@insp.jussieu.fr), Guillermo Rozas (Centro Atómico Bariloche & Instituto Balseiro, C. N. E. A., 8400 S. C. de Bariloche, Argentina, rozasg@ib.cnea.gov.ar), Alejandro Fainstein (Centro Atómico Bariloche & Instituto Balseiro, C. N. E. A., 8400 S. C. de Bariloche, Argentina, afains@cab.cnea.gov.ar), Bernard Jusserand (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, Bernard.Jusserand@insp.jussieu.fr), Bernard Perrin (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015

We present a study of the escape time of a 1 THz nanocavity phonon mode as a function of the Q-factor of the cavity. We compare results from picosecond acoustics and high resolution Raman scattering experiments, the latter obtained by means of a Fabry-Perot/triple-additive spectrometer tandem. A nanocavity consists of a GaAs spacer enclosed by two acoustic GaAs/AlAs Bragg mirrors. The number of periods of the inner mirror varies from sample to sample, spanning a range  $64 < Q < 2470$ . This means that the

cavity mode tunnels through the inner mirror to the substrate at theoretical time intervals that vary from 64 to 2470 ps for the different samples. An optic AlGaAs/AlAs Bragg mirror was grown between the substrate and the inner acoustic mirror in order to allow for the observation of the cavity mode in a backscattering configuration of the Raman experiments, otherwise forbidden by symmetry. At room temperature we observe escape times that vary from 65 to 278 ps. The theoretical values match the experimental results if a 3.0-GHz-wide lorentzian convolution is included to account for broadening effects. Possible explanations for this broadening will be discussed, as well as low temperature results.

### Invited Papers

10:40

**4aPAb8. Measurement of the velocity dispersion and attenuation in a liquid metal at GHz frequencies.** Oliver B. Wright (Division of Applied Physics, Graduate School of Engineering, Hokkaido University, 060 8628 Sapporo, Japan, olly@eng.hokudai.ac.jp), Bernard Perrin (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, bernard.perrin@insp.jussieu.fr), Osamu Matsuda (Division of Applied Physics, Graduate School of Engineering, Hokkaido University, 060 8628 Sapporo, Japan, omatsuda@eng.hokudai.ac.jp), Vitali Gusev (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, vitali.goussev@univ-lemans.fr)

Ultrashort optical pulses are used to excite and interferometrically detect picosecond longitudinal acoustic pulses in thin films of liquid mercury sandwiched between sapphire plates. By analysing consecutive acoustic echoes we derive the dispersion of the ultrasonic attenuation and sound velocity for this liquid at frequencies up to 10 GHz. Two types of optical detection, from the same side of the film as the excitation light and from the opposite side to the excitation light, are presented. Significant effects of structural relaxation are observed and are compared to a simple model that indicates the presence of picosecond relaxation times in mercury.

11:00

**4aPAb9. Pulse laser induced wave propagation in graded media and focusing devices.** Jacqueline Vollmann (ETH Zurich, Institute of Mechanical Systems, Dept. of Mechanical and Process Engineering, CH 8092 Zurich, Switzerland, vollmann@imes.mavt.ethz.ch), Juerg Bryner (ETH Zurich, Institute of Mechanical Systems, Dept. of Mechanical and Process Engineering, CH 8092 Zurich, Switzerland, bryner@imes.mavt.ethz.ch), Laurent Aebi (ETH Zurich, Institute of Mechanical Systems, Dept. of Mechanical and Process Engineering, CH 8092 Zurich, Switzerland, aebi@imes.mavt.ethz.ch), Jurg Dual (ETH Zurich, Institute of Mechanical Systems, Dept. of Mechanical and Process Engineering, CH 8092 Zurich, Switzerland, dual@imes.mavt.ethz.ch)

Near-infrared-laser pulses having durations of 100 fs are used to excite elastic waves thermoelastically propagating in a sub-THz frequency range. The elastic waves interact with inhomogeneities and carry information to the surface. The arrivals of the elastic pulses at the surface lead to transient changes of the optical reflectance which are monitored with short laser pulses which have a defined and controlled time delay relative to the initial pulses. Two activities of the research group are presented: The reflection and transmission behavior of acoustic waves propagating in graded media shows a frequency dependent nature and can therefore be used for filtering purpose. Time-boundary value problems are solved for various gradients with a finite-difference method. Results of the numerical simulation are presented and compared with laser-acoustic measurements. A 'classical' photoacoustic set-up provides an in-depth resolution of about 5 nm whereas the lateral resolution is in the order of 5 to 10 microns. To enhance the lateral resolution of the pump-probe technique, the elastic wave propagation along structures with arbitrary tip-like geometries consisting of orthotropic material is analyzed. With such structures representing ultrasonic lenses, the elastic energy is focused to a spot size given by the sharpness of the tip thereby leading to a higher lateral resolution.

### Contributed Papers

11:20

**4aPAb10. Acoustic dynamics in glasses in the mesoscopic range.** Giulio Monaco (ESRF, 6, rue Horowitz, 38043 Grenoble, France, gmonaco@esrf.fr)

The investigation of the high-frequency acoustic excitations in glasses and of their connection to the universal anomalies in the thermal properties remains a largely debated topic. For instance, one interpretation is based on the observation that the high-frequency acoustic dynamics in simulated harmonic glasses shares the same main features as those found in experiments on real glasses [1]. Another interpretation is based on the observation that both acoustic dispersion and attenuation measured in glasses and in the corresponding poly-crystals are indistinguishable [2]. A further interpretation is based on the observation that in some glasses the high-frequency acoustic attenuation increases as a power of  $q$  with an exponent of four or larger up to frequencies corresponding to the Boson peak [3]. Here, I will discuss the above approaches on the basis of recent inelastic x-ray scattering results on the high-frequency acoustic dynamics of glasses. [1] G. Ruocco et al., Phys. Rev. Lett. 84, 5788 (2000). [2] A. Matic et al., Phys. Rev. Lett. 93, 145502 (2004). [3] B. Ruffl  et al., Phys. Rev. Lett. 96, 045502 (2006).

11:40

**4aPAb11. Perspectives on spatial dispersion in cubic crystals provided by neutron scattering, phonon imaging, picosecond laser ultrasound and lattice dynamics models.** Arthur G. Every (School of Physics, University of the Witwatersrand, PO Wits 2050 Johannesburg, South Africa, arthur.every@wits.ac.za), Kudakwashe Jakata (School of Physics, University of the Witwatersrand, PO Wits 2050 Johannesburg, South Africa, Kudakwashe.Jakata@students.wits.ac.za)

Spatial dispersion is the variation of acoustic wave speed with wavelength, and sets in when the wavelength approaches the natural scale of length of a medium, or lattice spacing in the case of a crystal. The first onset of dispersion can be treated within the context of continuum mechanics by the incorporation of third and fourth order spatial derivatives of the displacement field in the elastic wave equation. These additional terms yield corrections to the phase velocity which in general are quadratic in the spatial frequency  $k$ . This paper will survey the experimental techniques that give one access to the coefficients in these expansions, in particular inelastic neutron scattering, ballistic phonon imaging and picosecond laser ultrasound. The main emphasis of the paper will be on deriving the numerical values of the

dispersion coefficients for four cubic crystals, Si, Ge, GaAs and InSb, from published neutron scattering data and demonstrating how modified continuum elastodynamics with these values of the coefficients is able to account well for the available dispersive phonon images of these crystals. Comparison will be made with values for the dispersion coefficients that have been obtained from laser ultrasound measurements and from lattice dynamics models.

#### 12:00

**4aPAb12. High frequency ultrasonic waves in metals and dielectrics.** Maria Eleftheriou (Department of Music Technology and Acoustics, Technological Educational Institute of Crete, 1 E. Daskalaki Str., 74100 Rethymnon, Greece, marel@physics.uoi.gr), Makis Bakarezos (Department of Music Technology and Acoustics, Technological Educational Institute of Crete, 1 E. Daskalaki Str., 74100 Rethymnon, Greece, bakarezos@stef.teicrete.gr), Andreas Lyras (Department of Physics, University of Ioannina, 45110 Ioannina, Greece, alyras@uoi.gr), Costas Kosmidis (Department of Physics, University of Ioannina, 45110 Ioannina, Greece, kkosmidis@uoi.gr), Michael Tatarakis (Department of Electronics, Technological Educational Institute of Crete, Romanou 3, 73133 Chania, Greece, m.tatarakis@chania.teicrete.gr), Nektarios Papadogiannis (Department of Music Technology and Acoustics, Technological Educational Institute of Crete, 1 E. Daskalaki Str., 74100 Rethymnon, Greece, npapadogiannis@stef.teicrete.gr)

We theoretically study the generation of high frequency ultrasonic waves by short laser pulses, as well as their propagation, in metals and dielectrics. For this purpose, we employ a theoretical model that applies to both cases of materials. In the case of the dielectric the theoretical model is reduced properly. We compute key physical quantities of the lattice deformation such as the temperature, the strain and the displacement of the bulk while we compare the obtained results for the two different abovementioned types of materials. The dependence of these quantities on the generating laser intensity and pulse duration is investigated, revealing interesting differences in their behavior.

#### 12:40-1:40 Lunch Break

#### Contributed Paper

#### 1:40

**4aPAb14. Surface acoustic waves propagating on microstructured phononic crystals.** Dieter M. Profunser (Division of Applied Physics, Graduate School of Engineering, Hokkaido University, 060 8628 Sapporo, Japan, dieter@profunser.net), Oliver B. Wright (Division of Applied Physics, Graduate School of Engineering, Hokkaido University, 060 8628 Sapporo, Japan, olly@eng.hokudai.ac.jp), Osamu Matsuda (Division of Applied Physics, Graduate School of Engineering, Hokkaido University, 060 8628 Sapporo, Japan, omatsuda@eng.hokudai.ac.jp), Yukihiro Tanaka (Division of Applied Physics, Graduate School of Engineering, Hokkaido University, 060 8628 Sapporo, Japan, yuki@eng.hokudai.ac.jp), Abdelkrim Khelif (Institut FEMTO-ST/CNRS, 32 avenue de l'Observatoire, 25044 Besançon cedex, France, abdelkrim.khelif@femto-st.fr), Vincent Laude (Institut FEMTO-ST/CNRS, 32 avenue de l'Observatoire, 25044 Besançon cedex, France, vincent.laude@femto-st.fr), Sarah Benchabane (Institut FEMTO-ST/CNRS, 32

#### 12:20

**4aPAb13. Imaging of 3D acoustic wave-fronts by means of picosecond laser ultrasonics in GaAs substrate.** Emmanuel Péronne (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, emmanuel.peronne@insp.jussieu.fr), Laurent Belliard (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, lbelliar@ccr.jussieu.fr), Shuo Zhang (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, Shuo.Zhang@insp.jussieu.fr), Clément Rossignol (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, c.rossignol@lmp.u-bordeaux1.fr), Bernard Perrin (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, bernard.perrin@insp.jussieu.fr)

Picosecond ultrasonic experiments are widely used as pump-probe technique using longitudinal acoustic pulses and measuring time dependence of the signal. We present an experiment which combines an interferometer with the control of the pump-probe spatial overlap. Such set-up can perform a new kind of bulk phonon imaging based on ultrafast optical generation and detection scheme. When ultrafast acoustic longitudinal wave propagate in crystals they generate coherent transverse acoustic wave by diffraction. Transverse waves are known to give rise to phonon focusing patterns. The imaging of such patterns in time and space allowed us to partially reconstruct the wave front surface of bulk acoustic wave in crystals. The data may be viewed as 2D slices of the 3D acoustic wavefront revealing the basic anisotropy of the solid with unprecedented micron lateral and in-depth resolution. Measurements are carried out in a 356  $\mu\text{m}$ -thick GaAs sample as an example, but many different experimental conditions are compatible with the present set-up. This technique provides a new way to perform phonon energy and phonon polarisation dependent experiment with high frequency phonons.

avenue de l'Observatoire, 25044 Besançon cedex, France, sarah.benchabane@femto-st.fr)

We investigate the interaction between high frequency surface acoustic waves and periodic microstructured patterns that form phononic crystals. The experimental method combines an optical pump-probe setup with interferometric detection and provides picosecond temporal and micron spatial resolutions. Surface acoustic waves with frequency components up to 1.3 GHz are imaged in real-time propagating over the periodic metamaterial. We used a DRIE (deep reactive ion etching) process to fabricate 2D air-silicon phononic crystals in the form of a square lattice. We present real-time animations of surface acoustic waves scattered by the phononic crystals. In particular we describe the frequency and angular dependence of the surface acoustic wave reflection from a 2D phononic crystal boundary. Fourier analysis allows us to reveal details of the acoustic band structure including gaps. The presence of such phononic band gaps enables us to visualize surface acoustic waves in waveguides, cavities and other phononic circuits at GHz frequencies.



2:00

**4aPab15. Nanoacoustic waves in nanomaterials.** Jaap I. Dijkhuis (Utrecht University, Ornsteinlaboratorium, Princetonplein 1, 3508TA Utrecht, Netherlands, j.i.dijkhuis@phys.uu.nl), Peter J. Van Capel (Utrecht University, Ornsteinlaboratorium, Princetonplein 1, 3508TA Utrecht, Netherlands, p.j.s.vancapel@phys.uu.nl)

We discuss the formation and detection of acoustic soliton trains in crystal slabs. High-amplitude strain pulses are generated by impact of intense femtosecond optical pulses on a metallic film and injected into the crystal slab. Nonlinear acoustic propagation leads to the formation of shock fronts (N-waves), that, in absence of viscous damping and by virtue of dispersion, may develop into soliton trains at the leading edge and high frequency tails at the trailing edge. Interferometric pump-probe optical experiments are discussed to directly detect the ultrafast surface displacements when a soliton is reflected at the surface of the crystal slab. Finally, we experimentally prove that these acoustic solitons are capable of impulsively exciting THz transitions in electronic centers in solids. The coherent terahertz acoustic pulse trains are applied to manipulate the optical response of two-dimensional excitons in a III-V quantum well on the ultrafast timescale. By virtue of the deformation potential, the coherent exciton emission becomes strongly "chirped" when the acoustic pulse train passes the quantum well. This yields the prospect to perform pump-probe terahertz acoustic experiments with nanoacoustic waves in semiconductor nanomaterials.

2:20

**4aPab16. Phonons rectification in picosecond laser ultrasonics.** Bernard Perrin (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, bernard.perrin@insp.jussieu.fr), Shuo Zhang (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, Shuo.Zhang@insp.jussieu.fr), Laurent Belliard (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, lbelliar@ccr.jussieu.fr), Emmanuel Péronne (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, emmanuel.peronne@insp.jussieu.fr)

In standard ultrafast acoustic experiments very short coherent acoustic pulses are generated by the absorption of a femtosecond laser pulse in a thin metallic transducer deposited on the sample. Subsequently the acoustic pulses and the heat generated in the transducer cross the sample and are partially transmitted in the underlying substrate. At low temperature, heating of the metallic transducer gives rise to the emission of incoherent phonons wave packets which propagate ballistically over large distances in the substrate. We report on a series of experiments which demonstrate the acoustic rectification [1-3] of these wave packets as they propagate through large GaAs or Si substrates. [1] B. Perrin, J. de Phys. **C8** (1979) 216. [2] S. M. Avanesyan, V. E. Gusev, Solid State. Commun. **54** (1985) 1065. [3] B. Perrin, E. Péronne, L. Belliard, Ultrasonics, **44** (2006) 1277.

2:40

**4aPab17. Optical characterization of the acoustic response in a nanostructure using the transient reflection matrix formalism.** Denis Mounier (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, denis.mounier@univ-lemans.fr), Pascal Picart (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, pascal.picart@univ-lemans.fr), Pascal Ruello (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, pascal.ruello@univ-lemans.fr), Jean-Marc Breteau (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, jean-marc.breteau@univ-lemans.fr), Vitali Gusev (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, vitali.goussev@univ-lemans.fr)

The transient reflectometry and transient interferometry are the most commonly used techniques of picosecond acoustics for the study of isotropic planar stratified nanostructures. Nevertheless when anisotropy is present in the sample, the standard techniques have to be completed by transient polarimetry. The reflection properties of an anisotropic sample at oblique incidence are completely determined by the 2x2 reflection matrix (RM):  $R = [r_{pp}, r_{ps}; r_{sp}, r_{ss}]$ . Considering that the transient acoustic phenomena induce a perturbation  $\Delta R$  of the reflection matrix, we demonstrate that the transient reflection matrix (TRM):  $\Delta R \cdot R^{-1}$ , where  $R^{-1}$  is the inverse of the reflection matrix, can be completely determined experimentally using the three techniques: transient reflectometry, interferometry and polarimetry (TRIP). In particular, the off-diagonal components of the TRM can be determined by transient polarimetry measurements only. Moreover, theoretical calculations of the TRM point up the close relation between the off-diagonal components of the TRM and the presence of a shear strain wave propagating perpendicularly to the free surface of the sample. Experimental results using the transient polarimetry technique will be presented to support the theoretical prediction of the TRM formalism.

3:00

**4aPab18. Picosecond ultrasonics in a single biological cell.** Bertrand Audoin (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, b.audoin@lmp.u-bordeaux1.fr), Clément Rossignol (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, c.rossignol@lmp.u-bordeaux1.fr), Nikolay Chigarev (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, n.chigarev@lmp.u-bordeaux1.fr), Marie-Christine Durrieu (INSERM U577, Université Victor Segalen Bordeaux 2, 146, rue Léo-Saignat, Case 45, 33076 Talence, France, durrieumc@bordeaux.inserm.fr), Fabien Guillemot (INSERM U577, Université Victor Segalen Bordeaux 2, 146, rue Léo-Saignat, Case 45, 33076 Talence, France, fabien.guillemot@bordeaux.inserm.fr), Guillaume Forget (Laboratoire de Biomatériaux et Réparation Tissulaire, INSERM U 577, Université Bordeaux 2, 146, rue Léo-Saignat, 33076 Bordeaux, France, guillaume.forget@bordeaux.inserm.fr)

The picosecond ultrasonic technique has been developed during the last 20 years due to permanent interest in the study of mechanical properties of nano- and micro-layers made of metals and semiconductors used in solid state physics and microelectronics. The technique relies on generation, propagation and detection of ultra-short acoustic waves by femtosecond laser pulses. Important advances

tages of these techniques are the absence of direct contact to the sample and extremely high frequency range. In this presentation, the picosecond ultrasonics technique is applied to generate and detect acoustic waves with a frequency as high as 15 GHz in a single biological cell. Hypersound velocity and attenuation can thus be mapped in the cell with the small lateral resolution provided by optics, ie  $1\mu\text{m}$ . In addition to single cell imaging, the sensitivity of the measurements to cell compressibility suggests promising perspectives in the field of biology. An application to the analysis of mouse cells grafting on biomaterials will illustrate the potentialities for quantitative evaluation of implants bio compatibility.

### Contributed Paper

3:20

**4aPAb19. Parallel detection for picosecond ultrasonics.** Richard Smith (University of Nottingham, School Electrical and Electronic Engineering, University Park, NG7 2RD Nottingham, UK, richard.j.smith@nottingham.ac.uk), Mike Somekh (University of Nottingham, School Electrical and Electronic Engineering, University Park, NG7 2RD Nottingham, UK, mike.somekh@nottingham.ac.uk), Steve Sharples (University of Nottingham, School Electrical and Electronic Engineering, University Park, NG7 2RD Nottingham, UK, steve.sharples@nottingham.ac.uk), Roger Light (University of Nottingham, School Electrical and Electronic Engineering, University Park, NG7 2RD Nottingham, UK, roger.light@nottingham.ac.uk), Nicholas Johnston (University of Nottingham, School Electrical and Electronic Engineering, University Park, NG7 2RD Nottingham, UK, Nicholas.Johnston@nottingham.ac.uk), Mark Pitter (University of Nottingham, School Electrical and Electronic Engineering, University Park, NG7 2RD Nottingham, UK, mark.pitter@nottingham.ac.uk)

Picosecond laser ultrasonics is a powerful technique for measurement and diagnosis of micro- and nano-scale structures. Unfortunately the data acquisition speeds are slow, this is a major drawback for widespread acceptance and usually makes imaging impractical. We are engaged in a program to accelerate the data capture rate by parallel acquisition of the incoming signals. This involves electronic developments as well as optical design. In order to parallelise the electronics we have used two approaches (i) a modified commercial camera and (ii) a custom built CMOS array detector. These approaches use an integrating camera which by application of suitable algorithms can perform the task of parallel lock in detection. Modulation depths below 1 part in 106, over 512 pixels can be readily detected with the commercial detector and smaller modulation depths are possible with the custom detector on account of the large well depth and hence high signal to noise. We also discuss the novel optical detection configurations that allow parallel detection of ultrasonic waves with ultrasonic wavelengths below the optical diffraction limit. Results are presented that show image acquisition rates orders of magnitude faster than normally possible with picosecond ultrasonic systems. Prospects for ultrahigh resolution optical resolution are discussed.

3:40-5:00 Posters

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

### Contributed Paper

5:00

**4aPAb20. Acoustic phonon generation by intrinsically localized vibrational modes in double-helices of DNA macromolecules and transition from inter-strand energy exchange to nonlinear self-trapping.** Yuriy A. Kosevich (Semenov Institute of Chemical Physics Russian Academy of Sciences, ul. Kosygina 4, 119991 Moscow, Russian Federation, yukosevich@yahoo.com), Alexander V. Savin (Semenov Institute of Chemical Physics Russian Academy of Sciences, ul. Kosygina 4, 119991 Moscow, Russian Federation, asavin@center.chph.ras.ru), Voislav L. Golo (Department of Mechanics and Mathematics, Lomonosov Moscow State University, 119992 Moscow, Russian Federation, voislav.golo@gmail.com), Yuriy S. Volkov (Department of Mechanics and Mathematics, Lomonosov Moscow State University, 119992 Moscow, Russian Federation, yu.volkov@gmail.com)

We study ultrafast dynamics of intrinsically localized vibrational modes (breathers) in a double helix of two weakly coupled chains of nonlinear oscillators. With this we model nonlinear dynamics of DNA-type macromol-

ecules, which can be studied by means of femtosecond infrared pump-probe laser spectroscopy similar to the case of protein  $\alpha$ -helices [1]. We show that there are two regimes of coupled breathers: the time-periodic wandering of low-amplitude breather between the chains, and the one-chain-localization (self-trapping) of high-amplitude breather. We also find bound states of two breathers, localized in different chains, with different positions in the chains. The helix symmetry of the system results in a specific chiral mode which accomplishes the interaction between torsional and longitudinal acoustic modes in the constituent chains. In both nonlinear regimes, the inter-strand energy exchange gives rise to acoustic phonon generation in the coupled chains, and the generation is much stronger in the wandering-breather regime. Ultrafast acoustic phonon generation can be detected by means of optoacoustics, which can therefore provide a tool to study in time domain the inter-strand energy exchange and the transition to nonlinear self-trapping in DNA-type macromolecules. 1. J. Edler, R. Pfister, V. Pouthier, C. Falvo, and P. Hamm, 2004, Phys. Rev. Lett. v. 93, 106405.

### Invited Papers

5:20

**4aPAb21. Coherent acoustic excitation of nanostructures probed with asynchronous optical sampling.** Thomas Dekorsy (University Konstanz, Fach M700, Fachbereich Physik, 78457 Konstanz, Germany, thomas.dekorsy@uni-konstanz.de)

We report the high-sensitivity detection of coherent acoustic excitation in semiconductor heterostructures and metallic nanostructures by using high-speed asynchronous optical sampling. Asynchronous optical sampling is based on two tunable femtosecond Ti:sapphire lasers with slightly different repetition rates close to 1 GHz. This new technique provides the performance of an all-optical oscilloscope for coherent excitations in a pump-probe set-up without any mechanically moving part. A time delay of 1 ns is scanned with a frequency of 10 kHz and a time resolution of 100 fs. Investigations on coherent zone-folded phonons in semiconductor superlattices and nanoscale metallic structures are discussed. For the latter the influence of the substrate on the damping of acoustic excitations is investigated in detail.

**4aPab22. Opto-acousto-optic evaluation of the physical properties of nanoporous materials.** Charfeddine Mechri (Université du Maine, avenue O. Messiaen, 72085 Le Mans, France, cmehri@hotmail.com), Pascal Ruello (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, pascal.ruello@univ-lemans.fr), M Nsenkoue (Université du Maine, avenue O. Messiaen, 72085 Le Mans, France, m.nsenkoue@univ-lemans.fr), Denis Mounier (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, denis.mounier@univ-lemans.fr), Jean-Marc Breteau (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, jean-marc.breteau@univ-lemans.fr), Mathieu Edely (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, mathieu.edely@univ-lemans.fr), Y Minhao (Université du Maine, avenue O. Messiaen, 72085 Le Mans, France, y.minhao@univ-lemans.fr), Alain Gibaud (Université du Maine, avenue O. Messiaen, 72085 Le Mans, France, alain.gibaud@univ-lemans.fr), I Povey (Tyndall National Institute, University College Cork, Prospect Row, 115 Cork, Ireland, i.povey@tyndall.ie), M Pemble (Tyndall National Institute, University College Cork, Prospect Row, 115 Cork, Ireland, m.pemble@tyndall.ie), Sergei Romanov (Tyndall National Institute, University College Cork, Prospect Row, 115 Cork, Ireland, sergei.romanov@tyndall.ie), Mikhail Baklanov (IMEC, Kapeldreef 75, B-3001 Leuven, Belgium, baklanov@imec.be), P Verdonck (IMEC, Kapeldreef 75, B-3001 Leuven, Belgium, verdonck@imec.be), Vitali Gusev (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, vitali.goussev@univ-lemans.fr)

The porous materials with the characteristic dimensions of the pores from few nanometers up to a few hundred of nanometers find applications in microelectronic industry (as low-k materials), in photovoltaics and for developing of effective chemical sensors. When the pores are ordered in a spatially periodic structure, these systems present photonic and/or phononic properties which are of a prime interest in applied optics and telecommunication (light and/or phonons spectrum control). Here we report how the methods of picosecond laser ultrasonics based on the generation and detection by lasers of the acoustic waves with frequencies in the band of 10 GHz - 1 THz (with the lengths of hundreds of nanometers down to few nanometers) are applied for the evaluation of the mechanical and optical properties of these materials.

6:00

**4aPab23. Nanoscale objects as promising high frequency acoustic transducers in picosecond acoustics.** Arnaud Devos (IEMN-CNRS, Cité Scientifique - Avenue Poincaré, BP 60069, 59652 Villeneuve d'Ascq Cedex, France, arnaud.devos@isen.fr)

Twenty year ago, H. Maris opened up the field of nanoscale acoustics by demonstrating the opportunity of using ultrashort optical pulses for generating and detecting high frequency acoustic waves. Roughly, an optical pulse is converted in a picosecond acoustic pulse through the optical absorption in a thin metallic layer. Since then, this so-called picosecond ultrasonics has known a larger and larger success all around the world. Up to now, picosecond ultrasonics meets two main limitations. First it is difficult to reach the THz range using the usual way of producing the acoustic pulse. Second, in the common geometry only longitudinal waves are excited by the laser. Here we present some results showing that nanoscale objects could help in overcoming both difficulties. We first show that semiconductor quantum dots can be a very efficient emitter of coherent phonons whose frequency can be higher than those obtained in metallic thin films. Second we show that 2D arrays of nanosize metallic dots offers a way of generating and detecting high frequency surface acoustic waves.

6:20

**4aPab24. Nanoultrasonics based on piezoelectric semiconductor nanolayers.** Chi-Kuang Sun (National Taiwan University, 1, Section 4, Roosevelt Road, 10617 Taipei, Taiwan, sun@cc.ee.ntu.edu.tw)

In this presentation, we will review our recent work on the development of nanoultrasonics based on piezoelectric semiconductor nanolayers. Through epitaxial growth of multiple or single piezoelectric semiconductor layers with a period on the order of 10 nm, nanoacoustic waves with a frequency of 1 terahertz and a wavelength of 10 nm can be excited and measured with femtosecond optical pulses. Using temporal coherent and spatial nonlinear optical controls, we are able to synthesize nanoacoustic waveforms and generate a lateral acoustic spot on the order of 100 nm without the need of the near-field optical techniques. In this presentation, we will also discuss the potential use of this terahertz acoustic source for various nanoacoustic applications, including nanoultrasonic imaging.

6:40

**4aPab25. Photoelastic transduction in photo-phononic nanodevices.** Alejandro Fainstein (Centro Atomico Bariloche & Instituto Balseiro, C. N. E. A., 8400 S. C. de Bariloche, Argentina, afainst@cab.cnea.gov.ar), Bernard Jusserand (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, Bernard.Jusserand@insp.jussieu.fr), Maria Florencia Pascual Winter (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, Florencia.PascualWinter@insp.jussieu.fr), Norberto Daniel Lanzillotti Kimura (Centro Atomico Bariloche & Instituto Balseiro, C. N. E. A., 8400 S. C. de Bariloche, Argentina, lanzilln@ib.cnea.gov.ar), Guillermo Rozas (Centro Atomico Bariloche & Instituto Balseiro, C. N. E. A., 8400 S. C. de Bariloche, Argentina, rozasg@ib.cnea.gov.ar), Bernard Perrin (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, bernard.perrin@insp.jussieu.fr), Agnès Huynh (INSP - UMR 7588 CNRS & Université Pierre et Marie Curie, 140 Rue de Lourmel, 75015 Paris, France, agnes.huynh@insp.jussieu.fr), Aristide Lemaitre (LPN, CNRS route de Nozay, 91460 Marcoussis, France, Aristide.Lemaitre@lpn.cnrs.fr)

We will discuss the new possibilities that semiconductor superlattices and acoustic nanocavities open for the controlled manipulation of quasi-monochromatic acoustic waves in the terahertz range. Playing with the specific electronic properties of quantum wells constituting acoustic nanodevices allows to selectively generate or detect phonons with a specific spatial distribution of the deformation along the acoustic device propagation axis. We could for instance demonstrate the selective generation of cavity phonons at resonance with cavity excitonic transitions or the increased photoelastic coupling of folded acoustic modes in mirrors when the number of nodes of the acoustic mode coincide with the one of the dominantly resonant excitonic transition. We also used the combination of photonic and phononic cavities to ensure phase matching with cavity phonons in the standard detection scheme corresponding to transient reflectivity

in the time domain or Raman backscattering in the frequency domain. Photonic cavities moreover provide a strong increase of the internal optical fields by quality factors up to 100 typically, resulting in high enhancements of the transduction efficiency of monochromatic phonons to the benefit of the envisioned high speed modulation of optoelectronic properties of coupled photo-phononic nanodevices.

## Contributed Papers

7:00

**4aPAb26. Thermomechanical behavior of surface acoustic waves in ordered arrays of nanodisks studied by near-infrared pump-probe diffraction experiments and finite element simulations.** Claudio Giannetti (Università Cattolica del Sacro Cuore, via dei Musei 41, I-25121 Brescia, Italy, c.giannetti@dmf.unicatt.it), Francesco Banfi (Università Cattolica del Sacro Cuore, via dei Musei 41, I-25121 Brescia, Italy, f.banfi@dmf.unicatt.it), Damiano Nardi (Università Cattolica del Sacro Cuore, via dei Musei 41, I-25121 Brescia, Italy, dnardi@dmf.unicatt.it), Bernard Revaz (Département de Physique Théorique, Université de Genève, 24 Quai Ansermet, CH-1211 Genève, Switzerland, bernard.revaz@gmail.com), Gabriele Ferrini (Università Cattolica del Sacro Cuore, via dei Musei 41, I-25121 Brescia, Italy, gabriele@dmf.unicatt.it), Paolo Vavassori (Dipartimento di Fisica - Università di Ferrara, via dell'università, I-44100 Ferrara, Italy, vavassori@fe.infn.it), Vitali Metlushko (Department of Electrical and Computer Engineering, University of Illinois at Chicago, Chicago, IL 60607, USA, vmetlush@ece.uic.edu), Fulvio Parmigiani (Dipartimento di Fisica, Università degli Studi di Trieste and Sincrotrone Trieste, Basovizza, I-34012 Trieste, Italy, fulvio@dmf.unicatt.it)

The ultrafast thermal and mechanical dynamics of a two-dimensional lattice of metallic nanodisks has been studied by near-infrared pump-probe diffraction measurements over a temporal range spanning from 100 fs to several nanoseconds. The experiments demonstrate that in these systems a surface acoustic wave (SAW), with a wave vector given by the reciprocal periodicity of the two-dimensional array, can be excited by  $\sim 120$  fs Ti:sapphire laser pulses. We unambiguously show that the observed SAW velocity shift originates from the mechanical interaction between the SAWs and the nanodisks, while the correlated SAW damping is due to the energy radiation into the substrate. In order to clarify the interaction between the nanodisks and the substrate, numerical calculations of both the elastic eigenmodes and the time-dynamics of the system, following the impulsive heating excitation by the laser, are performed. Simulations based on finite-

elements analysis, together with a wavelet analysis of our experimental data, suggest the opening of a band-gap at the centre of the super-Brillouin zone. The modes at the centre of the super-Brillouin zone are excited following laser excitation, as opposed to thermal population, of the elastic modes.

7:20

**4aPAb27. High frequency acoustics in nanostructures by spontaneous Brillouin light scattering.** Tim Still (Max Planck Institute for Polymer Research, Ackermannweg 10, 55128 Mainz, Germany, still@mpip-mainz.mpg.de), Markus Retsch (Max Planck Institute for Polymer Research, Ackermannweg 10, 55128 Mainz, Germany, retsch@mpip-mainz.mpg.de), Revekka Sainidou (Instituto de Óptica - CSIC, Serrano 121, 28006 Madrid, Spain, rsainid@io.cfm.ac.csic.es), Ulrich Jonas (Max Planck Institute for Polymer Research, Ackermannweg 10, 55128 Mainz, Germany, jonas@mpip-mainz.mpg.de), George Fytas (Department of Materials Science and Technology, University of Crete and Forth, 71110 Heraklion, Greece, fytas@mpip-mainz.mpg.de)

We report on the dispersion of high frequency (GHz) acoustic excitations in three-dimensional colloidal crystalline or amorphous assemblies of sub-micron particles in different matrices by Brillouin light scattering (BLS) in order to explore the phononic and elastic properties of nanostructured materials. In air, we record eigenmodes of the individual particles, which are shown to be independent from the crystallinity and the composition of the sample but sensitively depend on the particle architecture (e.g core/shell silica/PMMA, hollow spheres) and their mechanical properties at nanoscale. In fluid matrices, the dispersion relations are recorded. We demonstrate the occurrence of two hypersonic phononic band gaps of different nature. One is a Bragg gap appearing only in crystalline samples, the other one is a particle resonance-induced hybridization gap, which appears in crystalline samples along with the Bragg gap and alone in amorphous samples. Furthermore, we investigate the influence of filling fraction, crystallinity and monodispersity of size on the hypersonic behaviour of our samples.



**Session 4aPac****Physical Acoustics, Acoustical Oceanography, and Biomedical Ultrasound/Bioresponse to Vibration:  
Acoustically Activated Bubble Dynamics and Applications I**

Erich C. Everbach, Cochair

*Swarthmore College, 500 College Avenue, Engineering Department, Swarthmore, PA 19081, USA*

Joachim Holzfuss, Cochair

*Institute of Applied Physics, TU Darmstadt, Schlossgartenstr. 7, Darmstadt, 64289, Germany****Invited Papers*****8:00**

**4aPac1. Bubbles, surfactants, shape oscillations, optical levitation, and light scattering: a survey.** Philip L. Marston (Washington State University, Physics and Astronomy Department, Pullman, WA 99164-2814, USA, marston@wsu.edu), David B. Thiessen (Washington State University, Physics and Astronomy Department, Pullman, WA 99164-2814, USA, thiessen@wsu.edu)

Research emphasizing relatively slow aspects of bubble dynamics will be summarized. Though the attention is mainly on acoustically levitated bubbles larger than the size for monopole resonance, early experiments on stable optical levitation of gas bubbles in water [J. Acoust. Soc. Am. 83, 970-975 (1988)] will also be noted as well as fundamental aspects of light scattering by bubbles [J. Opt. Soc. Am. and/or Applied Optics (1979-1991)]. Modulated radiation pressure was demonstrated to be an effective way for mode-specific excitation of shape oscillations of acoustically levitated bubbles [J. Acoust. Soc. Am. 93, 706-713 (1993)]. The damping of shape oscillations was demonstrated to be strongly influenced by the presence of insoluble or soluble surfactants [J. Fluid Mech. 300, 149-167 (1995); Phys. Rev. Lett. 75, 2686-2689 (1995); J. Acoust. Soc. Am. 102, 3372-3377 (1997)]. A convenient way of measuring the damping was to use laser beam extinction to monitor freely decaying shape oscillations. This technique was sufficiently sensitive to reveal the strong dependence of the damping on surface elasticity. Even for clean bubbles, an improved analysis of the damping was needed. [Sponsored in part by NASA and by ONR.]

**8:20**

**4aPac2. Measuring the Extreme Conditions Created During Cavitation.** Kenneth S. Suslick (University of Illinois at Urbana-Champaign, 600 S. Mathews Av., Urbana, IL 61801, USA, ksuslick@uiuc.edu)

Extreme temperatures and pressures are produced through acoustic cavitation: the formation, growth and collapse of bubbles in a liquid irradiated with high intensity ultrasound. Single bubbles have generally been assumed to give higher temperature conditions than bubble clouds, but confirmation from the single bubble sonoluminescence (SBSL) emission spectra have been problematic because SBSL typically produces featureless emission spectra that reveal little about the intra-cavity physical conditions or chemical processes. Here we present definitive evidence of the existence of a hot, highly energetic plasma core during SBSL. From a luminescing bubble in sulfuric acid, excited state to excited state emission lines are observed both from noble gas ions (Ar<sup>+</sup>, Kr<sup>+</sup>, and Xe<sup>+</sup>) and from neutral atoms (Ne, Ar, Kr, and Xe). The excited states responsible for these emission lines range from 8.3 eV (for Xe) to 37.1 eV (for Ar<sup>+</sup>) above the ground state. Observation of emission lines allows for identification of intra-cavity species responsible for light emission; the energy levels of the emitters indicate the plasma generated during cavitation is comprised of highly energetic atomic and ionic species.

**8:40**

**4aPac3. Nonlinear dynamics of sonoluminescing bubbles.** Joachim Holzfuss (Institute of Applied Physics, TU Darmstadt, Schlossgartenstr. 7, 64289 Darmstadt, Germany, Joachim.Holzfuss@physik.tu-darmstadt.de)

The current work focuses on the acoustical and nonlinear dynamical aspects of sonoluminescence (SBSL). Several hydrodynamical instabilities in parameter space are analyzed in detail numerically. Their occurrence in experiments is discussed especially in the context of period doubled unisotropic light emission. The acoustical emissions during stable and unstable oscillations show characteristics of shock waves. The emitted sound generates a complex acoustic environment in the driving cell leading to backreactions to the bubble. Characteristic dynamical effects during unstable sonoluminescence are clarified. Chemical processes during high temperature and high pressure, spatial translations, gas diffusion, the highly nonlinear bubble oscillations and acoustic emissions are attributed to oscillations and modulations of bubble dynamics outside the range of stable SBSL. In particular reasons for quasiperiodic oscillations with incommensurate frequencies in different setups are found.

**4aPac4. Bubble-bubble interaction in disperse bubble clouds.** Andrew J. Szeri (University of California, 6119 Etcheverry Hall, Berkeley, CA 94720-1740, USA, andrew.szeri@berkeley.edu), Jean O. Toilliez (University of California, 6119 Etcheverry Hall, Berkeley, CA 94720-1740, USA, toilliez@me.Berkeley.EDU), Chris Heckman (University of California, 6119 Etcheverry Hall, Berkeley, CA 94720-1740, USA, checkman@berkeley.edu), Parastou Eslami (University of California, 6119 Etcheverry Hall, Berkeley, CA 94720-1740, USA, peslami20@berkeley.edu)

Disperse bubble fields driven by pressure waves feature effects of time delays associated with the finite speed of travel of the driving acoustic wave, and the finite travel time of pressure waves between bubbles. The relative spatial arrangement of bubble nuclei in a disperse field, and the direction of the incoming pressure wave, both influence the cavitation behavior of a cloud of nuclei. In this paper, we consider the dynamics of a disperse field of bubble nuclei driven by a strong rarefaction, such as one sees in shock wave lithotripsy. We make comparisons to experimental work published in J. I. Ilroeta, A. J. Szeri, Y. Zhou, G. Sankin and P. Zhong, Assessment of shock wave lithotripters via cavitation potential, *Physics of Fluids* 19, 086103 (2007). Bubble-bubble interactions are the key ingredient to resolving a mystery concerning the extent of cavitation bubble growth.

**4aPac5. The effects of sonication on bacteria.** Timothy J Mason (Sonochemistry Centre, Coventry University, Priory Street, CV1 5FB Coventry, UK, t.mason@coventry.ac.uk), L Paniwnyk (Sonochemistry Centre, Coventry University, Priory Street, CV1 5FB Coventry, UK, apx077@coventry.ac.uk), E Joyce (Sonochemistry Centre, Coventry University, Priory Street, CV1 5FB Coventry, UK, apx077@coventry.ac.uk)

High power ultrasound at frequencies around 20kHz is capable of killing bacteria and for many years has been standard technique in microbiology for the disruption of living cells to release their contents. So successful is this effect that ultrasound has been studied as a possible method for water disinfection. The energies required for using ultrasound alone are high but commercial equipment is available and is often used for disinfection in conjunction with other techniques such as ozonation or uv irradiation. Now evidence is emerging that is possible to induce effects on bacteria other than kill by modifying the acoustic energy entering the suspension (Ultrason. Sonochem. 10:315, 2003). This can be done by altering the duration of exposure, the acoustic power used or the frequency of the ultrasound. In this way sonication can lead to such effects as deagglomeration, enhanced reaction to biocides and gene transfer (Nucleic Acids Res. 35:e129, 2007).

**4aPac6. Cavitation activity in bacterial biofilms exposed to 1 MHz ultrasound.** Erich C. Everbach (Swarthmore College, 500 College Avenue, Engineering Department, Swarthmore, PA 19081, USA, ceverba1@swarthmore.edu), Roby Velez (Swarthmore College, 500 College Avenue, Engineering Department, Swarthmore, PA 19081, USA, rvelez1@swarthmore.edu), Amy C. Vollmer (Swarthmore College, 500 College Avenue, Biology Department, Swarthmore, PA 19081, USA, avollme1@swarthmore.edu)

An exposure system was previously developed to quantify destruction of bacterial biofilms by 1 MHz c.w. ultrasound at 0.8 MPa peak-to-peak acoustic pressure amplitude (JASA 122(5):3052, 2007). Bacterial killing is quantified via confocal microscopy using fluorescent *E. Coli* and image processing. Recently, a passive detector of inertial and stable cavitation was included, relying upon the presence and character of acoustic emissions. The detector, a PVDF array placed on the microscope slide forming the base of the exposure chamber, produces a proxy measure of cavitation activity during ultrasound exposure. Acoustic pressure thresholds for biofilm destruction and cavitation activity suggest that inertial and stable cavitation both play a role in biofilm destruction by ultrasound.

**4aPac7. Bubbles in piezo-acoustic inkjet printing.** Detlef Lohse (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, d.lohse@utwente.nl), Roger Jeurissen (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, r.j.m.jeurissen@tnw.utwente.nl), Jos De Jong (University of Twente, P.O. Box 217, Department of Science and Technology, 7500 AE Enschede, Netherlands, jos.dejong@tnw.utwente.nl), Michel Versluis (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, m.versluis@utwente.nl), Herman Wijshoff (Oce Technology, P.O. Box 101, 5900 MA Venlo, Netherlands, herman.wijshoff@oce.com), Marc Van Den Berg (Oce Technology, P.O. Box 101, 5900 MA Venlo, Netherlands, marc.van.den.berg@oce.com), Hans Reinten (Oce Technology, P.O. Box 101, 5900 MA Venlo, Netherlands, hans.reinten@oce.com)

Ink-jet printing is considered as the hitherto most successful application of microfluidics. A notorious problem in piezo-acoustic ink-jet systems is the formation of air bubbles during operation. They seriously disturb the acoustics and can cause the droplet formation to stop. We could show by a combination of acoustical detection and high-speed visualization that the air-bubbles are entrained at the nozzle and then grow by rectified diffusion. Experimental results on the droplet velocity as a function of the equilibrium radius  $R_0$  of the entrained bubble are presented, too. Surprisingly, the droplet velocity shows a pronounced maximum around  $R_0=17$  micrometer before it sharply drops to zero around  $R_0=19$  micrometer. A simple one-dimensional model is introduced to describe this counterintuitive behavior which turns out to be a resonance effect of the entrained bubble. We show that the bubble counteracts the pressure buildup necessary for the droplet formation. The channel acoustics and the air bubble dynamics are modeled. It is crucial to include the confined geometry into the model: The air bubble acts back on the acoustic field in the channel and thus on its own dynamics. This two-way coupling limits further bubble growth and thus determines the saturation size of the bubble.

10:40

**4aPac8. The acoustic excitation mechanism of bubbles released from a nozzle.** Grant Deane (Scripps Inst. Oceanography, Univ. California, San Diego, La Jolla, CA 92093, USA, gdeane@ucsd.edu), Helen Czerski (Scripps Inst. Oceanography, Univ. California, San Diego, La Jolla, CA 92093, USA, hczerski@ucsd.edu)

At the moment of their formation, bubbles emit a short pulse of sound. Bubble noise is associated with sound from a variety of natural processes, including whitecaps, waterfalls, breaking surf and rain. A number of acoustic excitation mechanisms for bubble noise have been proposed, including the increase in internal pressure of the bubble associated with the Laplace pressure, hydrostatic pressure effects, shape mode to volume mode coupling, and a fluid jet associated with the collapse of the neck of air formed during bubble creation. Using bubbles released from a nozzle as a model system, we have determined that sound production is excited by a sudden decrease in bubble volume driven by the collapse of the neck of gas joining the bubble to its parent. A simple analytical model of neck collapse driven by surface tension energy is in agreement with high speed photographic measurements, and sufficient to explain the details of acoustic excitation. [Work supported by ONR and NSF]

11:00

**4aPac9. Characterizing microbubble interactions with ultrasound using flow cytometry.** Thomas Matula (Center for Industrial and Medical

Ultrasound, Applied Physics Lab., University of Washington, 1013 NE 40th St., Seattle, WA 98105, USA, matula@apl.washington.edu), Jarred Swalwell (Center for Industrial and Medical Ultrasound, Applied Physics Lab., University of Washington, 1013 NE 40th St., Seattle, WA 98105, USA, jarred@u.washington.edu)

Characterizing the fundamental interaction of ultrasound with microbubbles is challenging because of the small spatial and temporal scales. High speed optical imaging is perhaps the most well-known method, as it provides direct information about their response. Although the image is in a plane, the image quality can be sufficient to obtain important information about bubble response. However, high-speed cameras are expensive, data is very limited, and difficult to process. We previously showed how light scattering can be used to obtain similar information - volume oscillations, destruction, even shell properties. Light scattering can be an inexpensive method for probing microbubbles. The difficulties with light scattering (also with optical imaging) are alignment and signal/noise. In this talk we will describe a technique to use commercially-available light-scattering systems to investigate the interaction of pulsed ultrasound with microbubbles. In particular, we developed a technique to insonify microbubbles flowing through the focal region of a flow cytometer. Attached to the quadrature side of a flow cuvette is a small piezoelectric transducer, driven in pulsed mode at various voltages to induce a bubble response. The light scattered from the bubbles can be used for sizing, destruction thresholds, and to assess volume oscillations. Funded by NIH #5R01EB000350

### Invited Paper

11:20

**4aPac10. Determination of cavitation bubble lifetimes using bubble-bubble coalescence data.** Franz Grieser (Department of Chemical and Biomolecular Engineering, The University of Melbourne, Parkville, Victoria, 3010 Melbourne, Australia, franz@unimelb.edu.au), Devi Sunarto (Department of Chemical and Biomolecular Engineering, The University of Melbourne, Parkville, Victoria, 3010 Melbourne, Australia, franz@unimelb.edu.au), Muthupandian Ashokkumar (Department of Chemical and Biomolecular Engineering, The University of Melbourne, Parkville, Victoria, 3010 Melbourne, Australia, masho@unimelb.edu.au)

The effect that surface-active solutes, such as aliphatic alcohols and sodium dodecylsulfate, have on the extent of bubble coalescence in liquids under different sonication conditions has been investigated by measuring the volume change of the solution following a period of sonication. The data obtained led to the conclusion that SDS does not reach equilibrium adsorption level at the bubble/solution interface. On this basis, a method is proposed for estimating nonequilibrium surface excess values for solutes that do not fully equilibrate with the bubble/solution interface during sonication. For the case of SDS in the presence of excess NaCl, the method was further employed to estimate the maximum lifetime of bubbles in a multibubble field. Data obtained from this study suggests that an acoustic bubble in a multibubble field has a finite lifetime, and that this lifetime decreases with increasing applied frequency, ranging from up to  $0.35 \pm 0.05$  ms for 213 kHz to  $0.10 \pm 0.05$  ms for 1062 kHz. These estimated lifetimes equate to a bubble in a multibubble field undergoing an upper limit of 50-200 oscillations over its lifetime for applied acoustic frequencies between 200 kHz and 1 MHz.

### Contributed Papers

11:40

**4aPac11. Interpretation of the pressure waves radiated by oscillating bubbles.** Karel Vokurka (Technical University of Liberec, Physics Department, Studentska 6, 461 17 Liberec, Czech Republic, karel.vokurka@tul.cz), Silvano Buogo (CNR-Istituto di Acustica 'O.M. Corbino', Via del Fosso del Cavaliere, 100, 00133 Rome, Italy, silvano.buogo@idac.rm.cnr.it)

An oscillating bubble is an excellent acoustic radiator. In a pressure wave emitted by the oscillating bubble information about the bubble properties and behavior is present. Hence, when using a suitable method, this information could be extracted and used to improve our understanding of the physical processes accompanying the bubble oscillations. However, to be able to extract this information, a number of prerequisites must be met. First, the measuring apparatus should be able to record a faithful copy of the pressure wave. Second, a large, statistically representative set of pressure records must be available for the analysis. Third, a suitable method must be used to analyze the recorded waves. All these requirements will be discussed in detail at the conference. Presented results are based on experience gained during evaluation of a large set of pressure records obtained recently in ex-

periments with spark generated bubbles. [Work has been partly supported (K.V.) by the Czech Ministry of Education as the research project MSM 4674787501.]

12:00

**4aPac12. Dynamics and radiation of single cavity in an abnormal compressible bubbly media.** Valeriy K. Kedrinskiy (Lavrentyev Institute of Hydrodynamics, Siberian Division of the Russian Academy of Sciences, Lavrentyev prospect 15, 630090 Novosibirsk, Russian Federation, kedr@hydro.nsc.ru)

The equation of a pulsation of a single cavity in the equilibrium (on pressure) bubbly medium was suggested. The state of such medium is described by Lyakhov's equation which at the condition of pressure equilibrium in the both phases (gas/liquid) becomes essentially simpler. The numerical analysis of features of cavity dynamics and the acoustic losses was executed. The notion "acoustic losses" mean a radiation generated by cavity. The analysis of radiation parameters was restricted by the vicinity of cavity wall from a liquid side. The studies of cavity behavior, the structure and

amplitude of a radiation have shown that the degree of a cavity compression by a stationary shock wave goes down when the volumetric concentration of gas phase K in the medium increases. The amplitude of pulsation essentially decrease and function R(t) (radius cavity vs. time) asymptotically (without oscillations) tends to the equilibrium state when K is equal approximately

3%. The equilibrium state is defined by amplitude of an incident shock wave and does not depend on K-value. The structure of a radiation wave takes the "soliton" form, its amplitude is essentially lesser and the width is much more in the comparison with corresponding parameters for a single-phase liquid. (RFBR 06-01-00317a financial support).

## 12:20-1:40 Lunch Break

### Contributed Papers

1:40

**4aPac13. Equilibrium state of a multi-size bubble population in a liquid.** Svetlana Kovinskaya (Mechmath LLC, 14530 Bluebird Trail, Prior Lake, MN 55372, USA, mechmath@mechmath.com)

A propagation of a linear pressure wave in the liquid with multi-size bubble population is investigated. The wave in the bubbly mixture is modeled as wave in the waveguide interacting with distributed resonators having different resonance frequencies. Each bubble sub-population has the frequency range of effective influence on the wave propagation. This range is started from the resonance frequency of the individual bubble and ended in dependency on the partial void fraction of this sub-population. Mutual influence of sub-populations leads to additional sound attenuation that in essence is a Landau damping. The dispersive equation for the propagating wave lets introduce a criterion for the equilibrium state of the bubbly mixture with multi-size bubble population. The equilibrium distribution which meets this criterion is found analytically. It is shown that the equilibrium distribution is an exact result from the resonant acoustical absorption theory. This theory is employed to find the bubble distribution from measured attenuation of the acoustical wave with assumption that only resonating bubbles contribute attenuation at the frequency of their resonance (neglecting off-resonance contributions). The deviation from the equilibrium contributes Landau damping into the resonant absorption. A dependency of Landau damping on the bubble size distribution is presented.

2:00

**4aPac14. Optical measurements of the hot spot and incandescent shock from high pressure cavitation in water.** Robert Hiller (Impulse Devices, Inc., 13366 Grass Valley Av. Unit H, Grass Valley, CA 95945, USA, asa@roberthiller.com), D. Felipe Gaitan (Impulse Devices, Inc., 13366 Grass Valley Av. Unit H, Grass Valley, CA 95945, USA, gaitan@impulsedevices.com)

Spontaneous acoustic cavitation in water at static pressure up to 300 bar has been experimentally investigated. Cavities are initiated by negative pressure and then collapse due to both acoustic pressure and shock waves reflected from the inner surface of the spherical resonator. The implosions result in intense (Mbar) shock waves and bright (1 nJ) light flashes which last from 5 to 40 nanoseconds. The optical spectrum of the flash is measured with a grating monochromator and intensified array detector for high wavelength resolution (~5 nm) but slow time resolution, and with a multiple-anode microchannel plate photomultiplier tube along with bandpass filters for fast time resolution (~1 ns) but poor wavelength resolution. The spectrum is generally broad-band and featureless, matching roughly to a Planck spectrum at 5000 to 8000K. The spectral and temporal structure of the flashes is matched to hydrocode simulations. The model suggests the flashes are due to a shell of hot, opaque, shocked water which surrounds and obscures the central hot core. (SMDC contract W9113M-07-C-0178)

2:20

**4aPac15. Investigation of bubble dynamics and sonoluminescence in megasonic fields.** Andrea Otto (Göttingen University, Friedrich-Hund-Platz 1, 37077 Göttingen, Germany, aotto@physik3.gwdg.de), Till Nowak (Göttingen University, Friedrich-Hund-Platz 1, 37077 Göttingen, Germany, tnowak@physik3.gwdg.de), Robert Mettin (Göttingen University, Drittes Physikalisches Institut, Friedrich-Hund-Platz 1, 37077 Göttingen, Germany, R.Mettin@physik3.gwdg.de), Frank Holsteins (SEZ AG,

SEZ-Strasse 1, 9500 Villach, Austria, F.Holsteins@at.sez.com), Alexander Lippert (SEZ AG, SEZ-Strasse 1, 9500 Villach, Austria, A.Lippert@at.sez.com)

Cavitation bubble motion and bubble structures in water are investigated for standing wave fields in the megasonic range by high-speed imaging. Larger degassing bubbles and small bubbles with high translation speeds can be resolved. Groups of bubbles arrange in lines or arrays, as reported earlier by Miller [Miller, JASA 62, 1977]. Additional, sonoluminescence is measured in overall long-term and phase-resolved (gated) long-term exposures. Several distinct luminescing islands can be detected. The findings seem to be strongly related to the standing wave nature of the pressure field in our setup. Conclusions on bubble distributions and for cleaning applications are drawn.

2:40

**4aPac16. Development of an acoustics-based instrument for bubble measurement in liquids.** Georges L. Chahine (Dynaflow, Inc., 10621-J Iron Bridge Rd, Jessup, MD 20794, USA, glchahine@dynaflow-inc.com), Xiongjun Wu (Dynaflow, Inc., 10621-J Iron Bridge Rd, Jessup, MD 20794, USA, xiongjun@dynaflow-inc.com), Xiaozhen Lu (Dynaflow, Inc., 10621-J Iron Bridge Rd, Jessup, MD 20794, USA, xiaozhen@dynaflow-inc.com)

The acoustic bubble spectrometer (ABS) is an acoustics-based device that provides bubble size distribution in a bubbly liquid through measurement at various frequencies of the sound speed and attenuation and solution of an inverse problem. Acoustic bursts of varying frequencies are emitted by one hydrophone and detected by another. A PC and data boards control signal generation, detection, signal processing, inverse problem solution, and results display. Extensive validation experiments were conducted against high speed-video optical measurements. The two methods give very close results for void fractions up to 3e-3, with the ABS possessing the significant advantage of enabling near real-time measurements. The field of application is being expanded to media other than water, and the technique improved to detect larger void fractions, with the help of numerical simulations of non-linear behavior of bubble clouds in acoustic fields.

3:00

**4aPac17. An acoustic resonator for determining the void fraction of bubbly mercury flows.** Ronald A. Roy (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, ronroy@bu.edu), Christopher E. Ormonde (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, chrisceo@bu.edu), Parag V. Chitnis (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, pchitnis@bu.edu), Robin O. Cleveland (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, robinc@bu.edu), R. Glynn Holt (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, rgholt@bu.edu)

An acoustic resonator for measuring free-gas void fraction of a helium-mercury mixture is investigated. We employ a vertical, stainless steel cylindrical waveguide with a 5.08-cm i.d., a 1.27-cm wall thickness, a 40-cm length, and pressure-release boundary conditions at both ends. A bubble injection flow loop produces 2-phase mixtures of varying void fraction that flows upwards through the tube, spills over, and recirculates. The resonator is driven from the top by a 2.54-cm diameter circular piston affixed to an



electrodynamic shaker. A hydrophone mounted 1 cm above the tube bottom is used to measure the frequency response of the system. Sound speed is inferred by assuming a linear dependence of axial mode number on mode frequency, and void fraction is calculated assuming a mixture sound speed for a bubble population with maximum sized much smaller than the resonant sizes in the modal frequency range (Wood's limit). The system was validated using non-bubbly water and water-air mixtures of different void fractions. Void fraction measurements for Helium-Mercury mixtures will be presented. [Supported by the ORNL Spallation Neutron Source, which is managed by UT-Battelle, LLC, under contract DE-AC05-00OR22725 for the U.S. Department of Energy.]

### 3:20

**4aPac18. Cavitation bubbles as microfluidic actuators.** Claus-Dieter Ohl (University of Twente, P.O. Box 217, Department of Science and Technology, 7500 AE Enschede, Netherlands, c.d.ohl@utwente.nl), Rory Dijkink (University of Twente, Physics of Fluids, Building Meander, Postbus 217, 7500 AE Enschede, Netherlands, r.j.dijkink@tnw.utwente.nl), Zwaan Ed (University of Twente, Physics of Fluids, Building Meander, Postbus 217, 7500 AE Enschede, Netherlands, e.v.zwaan@casema.nl), Séverine Le Gac (University of Twente, BIOS The Lab-on-a-Chip group, 7500 AE Enschede, Netherlands, S.LeGac@ewi.utwente.nl), Albert Van Den Berg (University of Twente, BIOS The Lab-on-a-Chip group, 7500 AE Enschede, Netherlands, A.vandenBerg@ewi.utwente.nl), Kinko Tsuji (Shimadzu Europa GmbH, 47269 Duisburg, Netherlands, kts@shimadzu.de)

In this talk we give an overview on the usage of single cavitation bubbles to pump, mix, and manipulate cells in microfluidics. The bubbles are generated with a laser pulse in optically transparent lab-on-a-chip devices. The bubble pulsations is inherently fast, thus although the characteristic dimensions are small high Reynolds numbers flow can be achieved.

### 4:00-5:20 Posters

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

### Contributed Papers

### 5:20

**4aPac20. Modeling of the effect of boundaries on ultrasound contrast agent microbubbles response.** Benjamin Dollet (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, benjamin.dollet@univ-rennes1.fr), Leen Van Wijngaarden (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, l.vanwijngaarden@tnw.utwente.nl), Nico De Jong (Erasmus MC, Dr Molewaterplein 50 room Ee2302, 3015GE Rotterdam, Netherlands, n.dejong@erasmusmc.nl), Detlef Lohse (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, d.lohse@utwente.nl), Michel Versluis (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, m.versluis@utwente.nl)

Ultrasound contrast agents are coated microbubbles currently extensively studied to target endothelial cells, for local drug delivery. It is therefore important to distinguish acoustically free-floating bubbles from bubbles located close, or targeted to blood vessels. Here, we propose a theoretical study to understand the effect of boundaries on bubble response to ultrasound. We consider the hydrodynamic interaction of a single bubble with a wall, including all possible bubble motions: volumetric oscillations, translation, and nonspherical deformations. We also include the friction in the viscous boundary layer along the wall. We derive the coupled equations of the bubble dynamics using a Lagrangian approach. We predict the bubble response to ultrasound, as a function of various parameters (applied frequency and amplitude, bubble size and coating, bubble/wall distance). We show that our new model predicts a decrease of the resonance frequency as a bubble gets closer to a wall, in agreement with experiments. We reproduce correctly the observed decrease of oscillation amplitude for a close to the wall, showing that it is due to the coupling between oscillation and trans-

Experiments show that depending on the channel height 2-dimensional or 3-dimensional fluid flow is generated. Interestingly, there exists a regime which can be described with the Laplace equation, thus it is essentially a potential inviscid flow. We will present the current work (of others and our group) on cavitation assisted pumping using the jetting effect, mixing flows through the creation of vorticity, and the interaction of a bubble with suspension cells. The first results on the latter promise a fruitful future for biologic relevant applications in integrated lab-in-chip devices.

### 3:40

**4aPac19. Ultrasonic synthesis of enzyme coated microbubbles.** Muthupandian Ashokkumar (Department of Chemical and Biomolecular Engineering, The University of Melbourne, Parkville, Victoria, 3010 Melbourne, Australia, masho@unimelb.edu.au), Francesca Cavalieri (Department of Chemical and Biomolecular Engineering, The University of Melbourne, Parkville, Victoria, 3010 Melbourne, Australia, masho@unimelb.edu.au), Franz Grieser (Department of Chemical and Biomolecular Engineering, The University of Melbourne, Parkville, Victoria, 3010 Melbourne, Australia, franz@unimelb.edu.au)

Gas-filled polymer coated microbubbles are intrinsically ultrasound responsive systems and when tailored with targeting features are promising candidates for smart drug delivery. We have ultrasonically synthesised stable, versatile, biodegradable and biocompatible microbubbles using the enzyme, lysozyme. The synthesis of lysozyme microbubbles has been achieved by sonicating an aqueous solution containing denatured lysozyme. The microbubbles have been characterised using a number of imaging techniques such as, SEM, AFM and light microscopy. We have observed that the experimental parameters such as, length of sonication, DTT concentration and denaturation time deeply affect the yield and the size of the microbubbles. We have also investigated the secondary structure and the enzymatic activity of lysozyme coated microbubbles. The lysozyme microbubbles have retained their enzymatic antimicrobial activities.

lation rather than the friction in the boundary layer. The threshold for non-spherical oscillations is also discussed and compared to experimental measurements.

### 5:40

**4aPac21. Experimental investigation of the effect of heating rate on pre-existing gas nuclei in a viscoelastic medium.** Ian Webb (University of Oxford, Medical Engineering Unit, 43 Banbury Road, OX2 6PE Oxford, UK, ian.webb@eng.ox.ac.uk), Manish Arora (University of Oxford, Medical Engineering Unit, 43 Banbury Road, OX2 6PE Oxford, UK, manish.arora@eng.ox.ac.uk), Stephen Payne (University of Oxford, Dept. of Engineering Science, Parks Road, OX1 3PJ Oxford, UK, stephen.payne@eng.ox.ac.uk), Ronald A. Roy (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, ronroy@bu.edu), Constantin C. Coussios (University of Oxford, Medical Engineering Unit, 43 Banbury Road, OX2 6PE Oxford, UK, constantin.coussios@eng.ox.ac.uk)

Inertial cavitation is known to play a key role in thermal HIFU therapy, both from the point of view of treatment safety and delivery, and as a potential tool for treatment monitoring. However, bubble behaviour in rapidly changing temperature fields remains poorly understood. Using a theoretical model, we have previously shown that, for a given initial bubble radius, a critical heating rate exists, above which the bubble will grow, and below which it will dissolve. In order to test this hypothesis, an electrical resistor embedded in 0.5% Xanthan Gum solution is used to impose a known temperature profile, measured by an array of thermocouples, on a series of embedded bubbles. An optical arrangement employing a 10 Megapixel CCD and a macro lens is used to image the bubbles within the gel at sufficient resolution for accurate sizing. The radius-time profiles for a range of bubble

sizes are thus extracted using image analysis techniques, allowing comparison with model predictions and subsequent refinement of the theoretical model. Future work will focus on the incorporating the effects of acoustic excitation, such as rectified diffusion, to develop a unified model of bubble behaviour in viscoelastic media under the effect of a HIFU field.

6:00

**4aPac22. Damage to single biological cells induced by laser-induced tandem microbubbles.** Georgy Sankin (Duke University, Department of Mechanical Engineering and Materials Science, Durham, NC 27708, USA, gns@duke.edu), Fang Yuan (Duke University, Department of Mechanical Engineering and Materials Science, Durham, NC 27708, USA, fang.yuan@duke.edu), Pei Zhong (Duke University, Department of Mechanical Engineering and Materials Science, Durham, NC 27708, USA, pzhong@duke.edu)

Recent studies have highlighted the potential for using laser-induced micro-cavitation in lab-on-a-chip devices. Shear stress in a liquid can be controlled and significantly enhanced by bubble-bubble interaction, providing new options for in situ cell treatment. Two micro-bubbles (10  $\mu$ s life time) are generated in a 25- $\mu$ m liquid layer using 5 ns tandem laser pulses delivered through the objective of a microscope. Bubble-bubble interaction in nearly two-dimensional flow is observed using high-speed video cameras. Two liquid micro-jets moving in opposite directions can be generated when the second bubble is produced at the maximum size of the first one. The jet velocity is estimated about 35 m/s. Particle imaging velocimetry reveals vortex flow motion around the oscillating bubble lasting for about 200  $\mu$ s. Cell lyses produced by jetting from asymmetric oscillation of tandem microbubbles are investigated at various bubble-cell distances and compared with the results from single symmetric bubble oscillation. The interaction of tandem microbubbles can produce microjetting, leading to damage of adjacent single biological cells.

6:20

**4aPac23. Introduction of a compliant gas-layer serves to mitigate damage to solid surfaces from the collapse of cavitation bubble clouds.** Parag V. Chitnis (Boston University, Dept. of Aersp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, pchitnis@bu.edu), Nicholas J. Manzi (Boston University, Dept. of Aersp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, nmanzi@bu.edu), Robin O. Cleveland (Boston University, Dept. of Aersp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, robinc@bu.edu), Ronald A. Roy (Boston University, Dept. of Aersp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, ronroy@bu.edu), R. Glynn Holt (Boston University, Dept. of Aersp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, rgholt@bu.edu)

The collapse of transient bubble clouds near a boundary was investigated. Transient cavitation bubbles were created using a shock-wave lithotripter. A porous ceramic disk (flow-pressure 7.5psi) was placed at the lithotripter focus. Air was forced through the disk to alter the boundary condition at the ceramic disk's proximal face. Gas pressure below 7.5psi resulted in a ceramic disk partially filled with fluid (rigid boundary); gas pressure over 7.5psi resulted in active bubbling at the proximal face (compliant boundary). Cavitation dynamics of bubble clouds near ceramic disks were studied for varying gas pressures (0-10psi). Images of the collapse were obtained from a high-speed camera. Additionally, a passive cavitation detector (3.5MHz focused transducer) was aligned with the lithotripter focus. Both the images and the acoustic measurements indicated that bubble clouds near

a rigid boundary collapse onto the boundary, forming a re-entrant liquid jet whose impact leads to surface erosion. When a compliant boundary is introduced, bubble clouds collapse away from the surface, thus mitigating cavitation damage. The damage to the ceramic disks was quantified using micro-CT imaging. [Supported by the ORNL Spallation Neutron Source, which is managed by UT-Battelle, LLC, under contract DE-AC05-00OR22725 for the U.S. Department of Energy.]

6:40

**4aPac24. Stability and simulations of pulsating contrast agents.** Nikos A. Pelekasis (Dept. Mechanical Engineering, University of Thessaly, Pedion Areos, 38334 Volos, Greece, pel@uth.gr), Kostas Tsiglifis (Dept. Mechanical Engineering, University of Thessaly, Pedion Areos, 38334 Volos, Greece, kotsigl@uth.gr)

The encapsulating membrane of ultrasound contrast agents, UCA's, is treated as a viscoelastic thin shell whose deviation from linear Hookean behavior is modeled as a strain softening or strain hardening effect via a parameter measuring the degree of membrane softness. As the amplitude of sound increases it controls the shift in resonance frequency until it hits the forcing frequency in which case an abrupt increase in the microbubble response takes place. Only strain softening shells exhibit this behavior. Deviations from sphericity are modeled via an additional parameter, namely the scalar bending modulus. This parameter controls static buckling of the shell, the onset of parametric instability and dynamic buckling. In this fashion phase diagrams can be constructed for a specific UCA that map regions of subharmonic growth of shape modes. Stability analysis and numerical simulations are employed in order to capture the onset, growth and break-up or saturation of shape modes. The above two parameters are added to the area dilatation modulus and viscosity of the membrane in order to construct a model that can be used to design new agents that behave optimally in different diagnostic or therapeutic modalities.

7:00

**4aPac25. Acoustic characterization of an ultrasound surgical transmitter in the linear and nonlinear regime of working.** Antonio Petosic (Faculty of Electrical Engineering and Computing, Unska 3, 10000 Zagreb, Croatia, antonio.petosic@fer.hr), Bojan Ivančević (Faculty of Electrical Engineering and Computing, Unska 3, 10000 Zagreb, Croatia, bojan.ivanecvic@fer.hr), Dragoljub Svilar (Brodarski Institut, Avenija Većeslava Holjevca bb, 10000 Zagreb, Croatia, dsvilar@hrbi.hr)

The method for measurement of a derived acoustic power of an ultrasound surgical transducer has been suggested in the free field conditions. The pressure field of the transmitter, immersed in depth of quarter wavelength and vibrating at the fundamental frequency ( $\approx 25$ kHz), has been measured with calibrated hydrophone at different excitation levels. In the linear regime, the transmitter has been theoretically described as an acoustic dipole, the source parameters have been found and good agreement between theoretical and experimental results is obtained. When transmitter is excited at higher excitation levels, the nonlinear behaviour in loading medium appears, with strong cavitation activity. In the averaged power spectrum of the recorded acoustic pressure signal, is evident the presence of harmonics ( $n \cdot f$ ), subharmonics ( $f/q$ ), ultraharmonics ( $n \cdot f/q$ ) of excitation frequency. The spatial pressure distribution of each discrete frequency component in the free acoustic field has been measured and its contribution to total acoustic power has been calculated. The total acoustic power in the cavitation noise signal is estimated integrating the averaged pressure power spectrum with appropriate contributions of each frequency component in the signal.

## Session 4aPPa

## Psychological and Physiological Acoustics: General Topics in Psychological and Physiological Acoustics IV

Elizabeth Strickland, Cochair

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## Contributed Papers

8:00

**4aPPa1. Circumaural transducer arrays for binaural synthesis.** Raphaël Greff (A-Volute, 4120 route de Tournai, 59500 Douai, France, raphael.greff@a-volute.com), Brian F. Katz (LIMS-CNRS, B.P. 133, 91403 Orsay, France, brian.katz@lmsi.fr)

Binaural cues such as the interaural time and level differences are the primary cues for estimation of the lateral position of a sound source, but are not sufficient to determine elevation or the exact position on the "cone of confusion". Spectral content of the head-related transfer function (HRTF) provides cues that permit this discrimination, notably high frequency peaks and notches created by diffraction effects within the pinnae. For high quality binaural synthesis, HRTFs need to be individualized, matching the morphology of the listener. Typical means for this are to measure or calculate the HRTF of the listener, but these lengthy and costly methods are not feasible for general public applications. This paper presents a novel approach for HRTF individualization, separating the head and torso effect from that of the pinnae. The head/torso component is numerically modelled while the pinnae component is created using a multiple transducer array placed around each pinna. The philosophy of this method consists in trying to excite the correct localization cues provided by the diffraction of the reconstructed wave front on the listener's own pinnae. Simulations of HRTF reconstruction with various array sizes and preliminary auditory localization tests are presented.

8:20

**4aPPa2. Interaural-time-difference sensitivity to acoustic temporally-jittered pulse trains.** Matthew J. Goupell (Austrian Academy of Science / Acoustics Research Institute, Wohllebengasse 12-14, 1040 Vienna, Austria, matt.goupell@gmail.com), Piotr Majdak (Austrian Academy of Science / Acoustics Research Institute, Wohllebengasse 12-14, 1040 Vienna, Austria, piotr@majdak.com), Bernhard Laback (Austrian Academy of Science / Acoustics Research Institute, Wohllebengasse 12-14, 1040 Vienna, Austria, Bernhard.Laback@oeaw.ac.at)

Bandpass-filtered pulse trains in acoustic hearing have been used to understand the high-rate pulse trains used in electrical stimulation strategies. In a left-right discrimination test, sensitivity to interaural time differences (ITD) in 600-pulses-per-second (pps) periodic pulse trains and aperiodic (temporally-jittered) pulse trains was tested with six normal-hearing listeners. It was found that jitter significantly and systematically increased ITD performance. A second experiment using 1200-pps pulse trains was performed to show that listeners were not solely benefiting from the longest interpulse intervals and thus the instances of reduced rate by adding jitter. To better understand the effect of jitter, the output of a basilar membrane model and a higher-level physiologically-based model was observed. Results from the modeling were reminiscent of an effect called "release from binaural adaptation" where the binaural system is reactivated by a temporal irregularity and this release possibly occurs at the level of the cochlear nucleus or lower. These results help understand pulse-rate limitations of ITD in cochlear-implant listeners.

8:40

**4aPPa3. From sounds to melodies: Memory for sequences of pitch and loudness.** Marion Cousineau (CNRS UMR 8158; Univ. Paris Descartes; Ecole Normale Supérieure, 29 rue d'Ulm, 75005 Paris, France, mcousineau@ens.fr), Daniel Pressnitzer (CNRS UMR 8158; Univ. Paris Descartes; Ecole Normale Supérieure, 29 rue d'Ulm, 75005 Paris, France, Daniel.Pressnitzer@ens.fr), Laurent Demany (CNRS UMR 5227; Univ. Victor Segalen, 146 rue Léo Saignat, 33076 Bordeaux, France, laurent.demany@psyac.u-bordeaux2.fr)

In order to understand speech or appreciate music, listeners have to process and remember patterns of sounds that vary along many perceptual dimensions. Here we investigated the perception of pitch sequences and loudness sequences, using a psychophysical method that uncouples discriminability and memory capacity. Pitch could be produced by either resolved or unresolved harmonics. Random sequences were constructed for which a single attribute (pitch or loudness) could take only two different values. These values were selected individually for each participant to produce equal discriminability ( $d'$ ) for isolated sounds. The participants then had to perform Same-Different judgments on pairs of sequences of two, four or eight elements each. We found that performance decreased rapidly with the number of elements for the loudness and pitch of unresolved harmonics conditions. With sequences of four and eight elements, performance was markedly better for the pitch of resolved harmonics condition. These findings show that short-term auditory memory capacity changes with the type of attribute that is varied within a sequence. For pitch, resolved harmonics yield a higher capacity than do unresolved harmonics; this could explain part of the difficulties encountered by cochlear implant users when listening to music.

9:00

**4aPPa4. Local-pitch identification accuracy depending on the trajectory of frequency-modulated tones.** Yuki Hiruma (School of Media Science, Tokyo Univ. of Technology, 1404-1 Katakuracho, Hachioji, 192-0982 Tokyo, Japan, m01043572a@mss.teu.ac.jp), Kiyooki Aikawa (School of Media Science, Tokyo Univ. of Technology, 1404-1 Katakuracho, Hachioji, 192-0982 Tokyo, Japan, aik@media.teu.ac.jp)

Local-pitch identification accuracies were analyzed for frequency-modulated (FM) tones. The problem was whether every portion of a continuous sound was perceived at the same accuracy or not. Psychophysical experiments revealed that the local-pitch identification accuracies were significantly different among the nodes of continuous FM tones. Also, the accuracies were dependent on the frequency trajectory shape. The stimuli included two types of piecewise-linear FM tones of up-down-up and down-up-down glide sequences. Each tone had four nodes; the initial, two intermediate points, and the final. The duration of each linear glide was 100 ms. The frequency range was between 1000 and 1500 Hz. A pair of FM tone was presented with one-second interval. The frequency was shifted up or down at one of the nodes in either of the tones. The shift amounts were 0%,



4%, and 8%. The subjects were requested to answer whether two pitch sequences were the same or different. The pitch identification accuracy was low at the initial for both types of FM tones. The accuracy at the final was highest for the up-down-up tone. The intermediate high frequency node showed the highest accuracy for the down-up-down tone. These results indicated that the local-pitch identification accuracies were trajectory-dependent.

9:20

**4aPPa5. Pitch discrimination: Combination of information across frequency.** Hedwig E. Gockel (MRC CBU, 15 Chaucer Rd., CB2 7EF Cambridge, UK, hedwig.gockel@mrc-cbu.cam.ac.uk), Robert P. Carlyon (MRC CBU, 15 Chaucer Rd., CB2 7EF Cambridge, UK, bob.carlyon@mrc-cbu.cam.ac.uk), Christopher J. Plack (Psychology Department, Lancaster University, LA1 4YF Lancaster, UK, c.plack@lancaster.ac.uk)

Performance ( $d'$ ) for fundamental frequency ( $F_0$ ) discrimination was measured for two complex tones (A and B) presented either individually or simultaneously. The objective was to investigate how information is combined in pitch processing. For most subjects, tones A and B were filtered from 1350-1650 and 3300-4200 Hz, respectively, such that both contained only unresolved components (components were added in sine phase). The tones had identical nominal  $F_0$ s of 75 Hz, and the difference in  $F_0$  between the two intervals of a 2AFC trial was the same for A and B. When A and B were tested individually, performance was equal for the two tones ( $d'$  value  $\sim 1$ ). For some subjects, to achieve equal  $d'$  for the two tones, all components were added in alternating phase,  $F_0$  was increased to 90 Hz, and tones A and B were filtered from 1375-1875 and 3900-5400 Hz, respectively. Presenting the tones simultaneously should increase  $d'$  by a factor of 1.41, if information from both were combined optimally. The results showed no increase in  $d'$  above that measured for the individual tones for three subjects, and only a very small improvement for another three subjects (factor of about 1.1) [supported by EPSRC Grant EP/D501571/1].

9:40

**4aPPa6. The role of compression in forward masking by Schroeder-phase complexes.** Magdalena Wojtczak (University of Minnesota, Department of Psychology, 75 E. River Road, Elliott Hall N218, Minneapolis, MN 55455, USA, wojtc001@umn.edu), Andrew J. Oxenham (University of Minnesota, Department of Psychology, 75 E. River Road, Elliott Hall N218, Minneapolis, MN 55455, USA, oxenham@umn.edu)

Waveforms with flat temporal envelopes can produce more forward masking than waveforms with more modulated, or peakier, temporal envelopes after auditory filtering, even when the rms amplitude of the two waveforms is equal. This has been explained in terms of basilar-membrane non-linearity, which can result in a higher rms amplitude for the flatter than for the peakier temporal waveform after compression. Here, forward masking was measured as a function of the phase curvature of two Schroeder-phase complexes, one with components around the signal frequency (on-frequency masker) and the other with components well below the signal frequency (off-frequency masker). The experiment tested the hypothesis that since the basilar-membrane response to the off-frequency complex at the signal-frequency place is presumably linear, masking should not depend on the phase curvature of the complex, whereas compression of the on-frequency masker should produce phase-dependent thresholds, with the minimum corresponding to the peakiest internal representation of the masker. The results replicate prominent phase effects using on-frequency maskers, but also show some phase effects with off-frequency maskers, which are not predicted by our current understanding of basilar-membrane compression. Other possible influences, such as efferent effects and neural compression, are considered. [Supported by NIH grant R01DC03909].

10:00

**4aPPa7. Additivity of auditory masking using Gaussian-shaped tones.**

Bernhard Laback (Austrian Academy of Science / Acoustics Research Institute, Wohllebengasse 12-14, 1040 Vienna, Austria, bernhard.laback@oeaw.ac.at), Peter Balazs (Austrian Academy of Science / Acoustics Research Institute, Wohllebengasse 12-14, 1040 Vienna, Austria, peter.balazs@oeaw.ac.at), Gwenael Toupin (Austrian Academy of Science / Acoustics Research Institute, Wohllebengasse 12-14, 1040 Vienna, Austria, toupin@kfs.oeaw.ac.at), Thibaud Necciari (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, necciari@lma.cnrs-mrs.fr), Sophie Savel (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, savel@lma.cnrs-mrs.fr), Sabine Meunier (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, meunier@lma.cnrs-mrs.fr), Solvi Ystad (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, ystad@lma.cnrs-mrs.fr), Richard Kronland-Martinet (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, kronland@lma.cnrs-mrs.fr)

Both temporal and spectral masking have been studied extensively in the literature. Mostly, they have been regarded as separate phenomena. Very little is known about the interaction between these two effects, i.e. masking in the time-frequency domain. Data on the time-frequency spread of masking evoked by a single Gaussian-shaped tone pulse are presented in an accompanying study at the same conference (Necciari et al.). The current study gathers data on the additivity of masking by up to four, approximately equally effective Gaussian maskers ( $ERB=600$  Hz), separated either along the time or the frequency axis. For temporal separation, the amount of masking increases with the number of maskers, with excess masking (exceeding linear additivity) of up to 25 dB. For frequency separation (preliminary data) excess masking amounts up to 15 dB, and the higher-frequency masker (relative to the target) contributes more to the additivity than the lower-frequency maskers. Experiments with multiple maskers combining both temporal and frequency separation are underway. Combined with the single masker data, these data may serve as a basis for modeling time-frequency masking effects in complex signals. Work partly supported by OEAD (WTZ project AMADEUS) and ANR

10:20-10:40 Break

10:40

**4aPPa8. Auditory masking using Gaussian-windowed stimuli.**

Thibaud Necciari (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, necciari@lma.cnrs-mrs.fr), Sophie Savel (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, savel@lma.cnrs-mrs.fr), Sabine Meunier (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, meunier@lma.cnrs-mrs.fr), Solvi Ystad (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, ystad@lma.cnrs-mrs.fr), Richard Kronland-Martinet (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, kronland@lma.cnrs-mrs.fr), Bernhard Laback (Austrian Academy of Science / Acoustics Research Institute, Wohllebengasse 12-14, 1040 Vienna, Austria, bernhard.laback@oeaw.ac.at), Peter Balazs (Austrian Academy of Science / Acoustics Research Institute, Wohllebengasse 12-14, 1040 Vienna, Austria, peter.balazs@oeaw.ac.at)

This study investigates auditory masking with Gaussian-windowed tones as target and masker stimuli. On the purpose of developing a time-frequency masking model, such stimuli minimize the time-frequency uncertainty. Also, as proposed by van Schijndel *et al.* (1999), they activate a single spectro-temporal observation window of the auditory system. The study presented here measured auditory masking with Gaussian-windowed stimuli with an ERB of 600 Hz and an effective duration of 9.6 ms. The masker was centered at 4 kHz. Its level was 60 dB SL. Four experiments were conducted. (1) Absolute thresholds for Gaussian-windowed and 300-ms-sinusoidal targets were measured and compared for 11 frequencies. (2) Masking patterns were obtained with targets of various frequency separations from the masker. (3) Forward masking functions with 4-kHz targets were measured at 5 temporal separations. (4) Forward masking was measured for different frequency separations between masker and target. These data are compared with those typically obtained with stimuli that are broader either in the frequency or time domain. A modelling attempt is made. A accompanying article



on multiple masker additivity based on the present results is presented at the same conference (Laback *et al.*). Work partly supported by EGIDE (PAI Amadeus) and the ANR.

11:00

**4aPPa9. Gaussian-noise discrimination as a tool to investigate auditory object formation.** Tom L. Goossens (Technische Universiteit Eindhoven, Den Dolech 2, 5600 MB Eindhoven, Netherlands, tomgoos@gmail.com), Steven Van De Par (Philips Research Europe, Digital Signal Processing (MS WO02), High Tech Campus 36, 5656 AE Eindhoven, Netherlands, steven.van.de.par@philips.com), Armin Kohlrausch (Philips Research Europe, Digital Signal Processing (MS WO02), High Tech Campus 36, 5656 AE Eindhoven, Netherlands, armin.kohlrausch@philips.com)

In the present study we show that, in a same/different experiment, listeners are good at discriminating 50-ms Gaussian-noise tokens with a spectral range of 350-850 Hz. However, when an identical 200-ms noise fringe, with the same statistical properties as the 50-ms target tokens, is appended to both target tokens, listeners show very poor discrimination performance. Apparently, these identical fringes cannot be ignored and these extra non-informative fringes impair the discrimination of the target tokens. It seems that a target token and the appended fringe form one auditory object and that access to subparts of these tokens is not possible. When a perceptual cue is introduced that can lead to the segregation of the target token and noise fringe, e.g., a temporal gap between target and fringe, the ability to discriminate improves implying that the non-informative noise can be (partly) ignored when it is part of a different auditory object than the target token. This method is used as a new approach to investigate the influence of cues such as spectral range, level, interaural level difference, and interaural time delay, on the formation of auditory objects.

11:20

**4aPPa10. Evidence for Poisson processes in change detection.** Christian Kaernbach (Institut für Psychologie, Universität Kiel, Olshausenstr. 62, 24098 Kiel, Germany, chris2008@kaernbach.de)

Comparisons between two stimuli (e.g., "which stimulus was louder?") and change detection ("same or different?") are often assumed to operate on the same decision basis. In the Gaussian signal detection theory, each of the two stimuli to be compared is transformed into a number, and the comparison is then made between these two numbers. If both stimuli are well above absolute threshold, the numbers to be compared have to be large, which motivates the use of normal distributions. The present study tests this assumption by measuring same-different ROC curves for the detection of small changes in the intensity of sinusoids. In contrast to previous studies, change detection was measured not only when the possible direction of change was a priori unknown, but also in two conditions where changes had a fixed direction. The obtained ROCs are asymmetric. This points to Poisson processes with low means. Moreover, the sensitivity to increments was significantly higher than the sensitivity to decrements. The results put into question the Gaussian theory of sensory comparisons and change detection.

11:40

**4aPPa11. Low level auditory processing of simple and complex sounds in autism.** Anna C. Bonnel (McGill University, Faculty of Educational and Counselling Psychology, 3700 McTavish Street, Montreal, QC H3A1Y2, Canada, anna.bonnel@mail.mcgill.ca), Stephen McAdams (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, smc@music.mcgill.ca), Bennett K. Smith (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill

Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, bennett.smith@music.mcgill.ca), Armando Bertone (McGill University, Faculty of Educational and Counselling Psychology, 3700 McTavish Street, Montreal, QC H3A1Y2, Canada, armando.bertone@mcgill.ca), Jake A. Burack (McGill University, Faculty of Educational and Counselling Psychology, 3700 McTavish Street, Montreal, QC H3A1Y2, Canada, jake.burack@mcgill.ca), Valter Ciocca (University of British Columbia, School of Audiology and Speech Sciences, 5804 Fairview Avenue (J.M. Mather Building), Vancouver, AB V6T 1Z3, Canada, vvciocca@audiospeech.ubc.ca), Laurent Mottron (Pervasive Developmental Disorders Clinic, Riviere-des-Prairies Hospital and Fernand Seguin Research Center, University of Montreal, Montreal, QC H3C 3J7, Canada, mottronl@istar.ca)

The documented dissociation between enhanced and diminished auditory processing performance in autistic listeners may be linked to the neural complexity required to process auditory stimuli. To test this hypothesis, four discrimination experiments were designed targeting pitch, spectral envelope, vocal timbre, and loudness. A range of pure- and complex-tone stimuli, with or without frequency or amplitude modulation, varied along spectral and temporal dimensions. An adaptive procedure was used to assess the auditory discrimination thresholds of groups of high-functioning participants with autism (HFA), Asperger syndrome (ASP), and typically developing individuals (TDs). Our research question was whether increasing the level of spectral and/or temporal complexity would have a detrimental impact on autistic listeners' ability to discriminate between acoustic stimuli. Preliminary results suggest that auditory discrimination performance of the latter group is not as dependent on levels of spectro-temporal complexity as originally predicted. The results will be interpreted in the context of current perceptually based models of enhanced and diminished perceptual functioning in autism.

12:00

**4aPPa12. Roughness detection in fricative-like noise and tone stimuli.**

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)

Audio (spectral) and modulation (envelope) frequencies both carry information in a speech signal. While low modulation frequencies (2-20Hz) convey syllable information, higher modulation frequencies (80-400Hz) allow for assimilation of perceptual cues, e.g., the roughness of amplitude-modulated noise in voiced fricatives, considered here. Psychoacoustic 3-interval forced-choice experiments measured AM detection thresholds for modulated noise accompanied by a tone with matching fundamental frequency at 125Hz: (1) tone-to-noise ratio (TNR) and phase between tone and noise envelope were varied, with silence between intervals; (2) as (1) with continuous tone throughout each trial; (3) duration and noise spectral shape were varied. Results from (1) showed increased threshold (worse detection) for louder tones (40-50dB TNR). In (2), a similar effect was observed for the in-phase condition, but out-of-phase AM detection appeared immune to the tone. As expected, (3) showed increased thresholds for shorter tokens, although still detectable at 60ms, and no effect for spectral shape. The phase effect of (2) held for the short stimuli, with implications for fricative speech tokens (40ms-100ms). Further work will evaluate the strength of this surprisingly robust cue in speech.

12:20

**4aPPa13. Measurement of equal-loudness contours using eardrum pressure as reference signal.**

Sebastian Schmidt (Institute of Communication Acoustics, Ruhr-Universität Bochum, IC 1/142, Universitätsstr. 150, 44780 Bochum, Germany, sebastian.schmidt@rub.de), Herbert Hudde (Institute of Communication Acoustics, Ruhr-Universität Bochum, IC 1/142, Universitätsstr. 150, 44780 Bochum, Germany, herbert.hudde@rubr-uni-bochum.de)

Equal-loudness contours represent the relationship of loudness perception and sound pressure at the ear. Usually, the reference pressure is defined with a standardised calibration procedure. Individual ear canal characteristics significantly influence the contours, resulting in peaks and notches. By

choosing the eardrum pressure as reference for the perception measurements, individual ear canal features are cancelled, when loudness and pressure are related. Thus, flatter contours can be achieved. Finite-element simulations of the sound field in the canal have shown that signals at remote positions may differ significantly from the eardrum pressure. Thus, to achieve eardrum related measurements, it is necessary to insert probe microphones sufficiently deep into the canal. However, the desired signal has

to be estimated from a distance, since measurements directly at the drum are not practical. A method is shown to determine both the probe tip distance and the transformation terms to calculate the drum pressure estimate. In this contribution, the results of equal-loudness contour measurements are discussed. The positioning and transformation processes are verified by simulations with one-dimensional and finite-element models and experiments.

THURSDAY MORNING, 3 JULY 2008

ROOM 242A, 8:40 A.M. TO 7:00 P.M.

## Session 4aPPb

### Psychological and Physiological Acoustics: General Topics in Psychological and Physiological Acoustics V

Elizabeth Strickland, Cochair

*Purdue University, 500 Oval Drive, West Lafayette, IN 47907, USA*

Armin Kohlrausch, Cochair

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Alain De Cheveigne, Cochair

*CNRS, Universite Paris 5, Ecole Normale Supérieure, 29 rue d'Ulm, Paris, 75230, France*

#### Contributed Papers

8:40

**4aPPb1. A fast FFT-based integral-equation solver for simulation of elastoacoustic wave propagation in human head.** Elizabeth Bleszynski (Monopole Research, 739 Calle Sequoia, Thousand Oaks, CA 91360, USA, elizabeth@monopoleresearch.com), Marek Bleszynski (Monopole Research, 739 Calle Sequoia, Thousand Oaks, CA 91360, USA, marek@monopoleresearch.com), Thomas Jaroszewicz (Monopole Research, 739 Calle Sequoia, Thousand Oaks, CA 91360, USA, tomek@monopoleresearch.com)

We describe formulation, implementation, and representative applications of a fast integral-equation solver for modeling wave propagation in inhomogeneous visco-elastic and visco-acoustic media (e.g. in biological tissues). The present approach is an extension of our work on fast integral equation solvers in pure acoustics. It is based on Lippmann-Schwinger (L-S) integral equations. It incorporates: (i) FFT-based compression of the stiffness matrix and the corresponding fast iterative method resulting in the solution complexity proportional to the number of unknowns, and ability to solve problems of several million unknowns, (ii) piecewise-linear basis functions supported on tetrahedra, representing displacement field, and corresponding efficient algorithms for evaluation of Galerkin matrix elements, (iii) rigorous two-stage solution scheme applicable to scatterers composed of high contrast materials and consisting of transforming the L-S equations into a system of two well-conditioned problems (conventional solution schemes, when applied to such scatterers, lead to stiffness matrix of large condition number). We present applications of the developed approach to simulation of elastic wave propagation in realistic models of the human head, with the goal of comparing sound transmission through the normal auditory airways and through bone conduction, and in the presence and absence of noise-protective devices. This work is supported by the Air Force Office of Scientific Research.

9:00

**4aPPb2. Towards a transfer function used to adjust audio for bone-conduction transducers.** Raymond M. Stanley (Georgia Institute of Technology, Georgia Tech, School of Psychology, 654 Cherry St, Atlanta,

GA 30332-0170, USA, rms@gatech.edu), Bruce N. Walker (Georgia Institute of Technology, Georgia Tech, School of Psychology, 654 Cherry St, Atlanta, GA 30332-0170, USA, bruce.walker@psych.gatech.edu)

Bone-conduction transducers may effectively replace normal air-conduction headphones in cases where the ears need to be plugged, or else remain unoccluded. However, sounds designed to be presented via air conduction need to be adjusted to maintain optimal perception via bone conduction. This study sought to find bone-to-air amplitude and phase shifts, as preliminary data for a complete transfer function between the bone-conduction and air-conduction pathway. The variability or stability of the shift data can indicate the feasibility of making effective adjustments to sounds to account for the bone-conduction pathway. Listeners cancelled air-conducted and bone-conducted tone pairs by method of adjustment at three frequencies (500, 3150, and 8000 Hz). The amplitude adjustments were relatively consistent, while the phase adjustments were quite variable. Further analysis revealed that the variability in phase adjustments came from differences between people, but were relatively consistent within a person. Together, these data suggest that generalized adjustments for the bone-conduction pathway may not be effective, but that individualized adjustments may be both necessary and potentially quite effective. These results can be extended to continuous transfer functions applied to sounds before they are presented via bone-conduction transducers ("bonephones") in an auditory display.

9:20

**4aPPb3. Variability of hearing protection devices attenuation as a function of source location.** Hugues Nelisse (IRSST, Service de la recherche, 505 Boulevard de Maisonneuve O, Montréal, QC H3A3C2, Canada, hugnel@irsst.qc.ca), Marc-André Gaudreau (École de Technologie Supérieure, Département de Génie Mécanique, 1100, rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, gaudream@cdrommond.qc.ca), Jérémie Voix (École de Technologie Supérieure, Département de Génie Mécanique, 1100, rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, jvoix@jerevox.com), Jérôme Boutin (IRSST, Service de la recherche, 505 Boulevard de Maisonneuve O, Montréal, QC H3A3C2, Canada, jerbou@irsst.qc.ca), Frédéric Laville (École de Technologie Supérieure,

4a THU. AM

It is of common knowledge, and well documented, that laboratory-measured noise attenuation values of most hearing protection devices (HPD) exceed significantly the attenuation values obtained in real-world workplace environments. Various reasons may explain such discrepancies (lack of training, wearing time, lack of comfort, bad fitting, noise environments, etc.) but very few of them have been studied in details due to the complexity of the problem. This study focuses on the variability of the attenuation of HPDs as a function of the location of the noise source. Laboratory measurements were performed where subjects, wearing a HPD and facing a loudspeaker, were asked to rotate slowly on a rotating chair to simulate different angular positions of the head relative to the source. The protected and unprotected sound pressure signals for both ears were recorded as time signals using miniature microphones placed respectively inside and outside the HPD (F-MIRE technique). The microphones signals were processed to obtain attenuation values for the different angular positions. Results for different type of HPD (ear-muffs and ear-plugs) are presented and discussed.

#### 9:40

**4aPPb4. Assessment of exposure to impulsive and continuous noise by auditory brainstem response.** Jan Zera (Centr. Inst. for Labour Prot. - Natl. Res. Inst., Czerniakowska 16, 00-701 Warsaw, Poland, jazer@ciop.pl), Krzysztof Kochanek (The Institute of Physiology and Pathology of Hearing, Pstrowskiego 1, 01-943 Warsaw, Poland, k.kochanek@ifps.org.pl), Adam Pilka (The Institute of Physiology and Pathology of Hearing, Pstrowskiego 1, 01-943 Warsaw, Poland, a.pilka@ifps.org.pl), Rafal Mlynski (Centr. Inst. for Labour Prot. - Natl. Res. Inst., Czerniakowska 16, 00-701 Warsaw, Poland, rmlynski@ciop.pl)

The aim of the work was to measure the wave V latency or thresholds in the auditory brainstem responses (ABRs) produced by impulsive and continuous noise. A forward-masking paradigm was used in which the ABR was evoked by a 4-kHz tone pip. The tone pip was masked by a preceding 201-ms or 501-ms interval of click trains or band-pass noise presented at various levels. The inter-click interval ranged from 20 ms (50 clicks/s) to 100 ms (10 clicks/s). The center frequencies of the noise bands ranged from 250 to 4000 Hz. Results show that changes in wave V latency may be used as an indicator of the equivalence of the effect of various kinds of noise on the human auditory system. [Work supported by the Polish Ministry of Science and Higher Education, grant 4T07B00428].

#### 10:00

**4aPPb5. Evaluation of auditory processing disorder and auditory efferent system functionality in adult dyslexics: towards a unification theory of auditory-language processing impairments?** Michel Hoen (Laboratoire d'Etude des Mécanismes Cognitifs (EMC). EA 3082 CNRS, Université Lumière Lyon 2, 5, Avenue Pierre Mendès-France, 69676 Bron Cedex, France, Michel.Hoen@univ-lyon2.fr), Claire Grataloup (Laboratoire Dynamique du Langage (DDL), UMR 5596 CNRS, Université de Lyon et Lyon 2, Institut des Sciences de l'Homme - 14 avenue Berthelot, 69363 Lyon Cedex 07, France, Claire.Grataloup@pse.unige.ch), Evelyne Veuillet (Univ. Lyon 1 - Lab. Neurosciences, Service Pr Collet, Pavillon U, Hôpital Edouard Herriot, F-69003 Lyon, France, iakhoun@olfac.univ-lyon1.fr), Hung Thai-Van (Univ. Lyon 1 - Lab. Neurosciences, Service Pr Collet, Pavillon U, Hôpital Edouard Herriot, F-69003 Lyon, France, iakhoun@olfac.univ-lyon1.fr), Lionel Collet (Univ. Lyon 1 - Lab. Neurosciences, Service Pr Collet, Pavillon U, Hôpital Edouard Herriot, F-69003 Lyon, France, iakhoun@olfac.univ-lyon1.fr), Fanny Meunier (Laboratoire Dynamique du Langage (DDL), UMR 5596 CNRS, Université de Lyon et Lyon 2, Institut des Sciences de l'Homme - 14 avenue Berthelot, 69363 Lyon Cedex 07, France, fanny.meunier@univ-lyon2.fr)

In a recent paper, Veuillet et al., (2007) suggested that some auditory processing mechanism could be impaired in children with dyslexia. They reported a link between children's ability to perceive phonemic boundaries and the physiological functionality of their medial olivocochlear system (MOC), an auditory efferent pathway functioning under central control. In the present experiment, we extended this observation by comparing speech-in-speech comprehension performances in a group of control participants

(N=40) and a group of adults who had been diagnosed dyslexic as children (N=49). Confirming the idea that patients with dyslexia present auditory processing disorders (APD), we show that adult dyslexics exhibit greater difficulty in comprehending speech in noise. Data moreover suggest a link between speech-in-noise comprehension difficulty and the MOC functionality in these participants. More precisely, it appears that the absolute functionality of the left and right MOC bundles does not differ in dyslexic patients compared to controls. What appears to be differing is the functional asymmetry in the MOC functionality between both ears, normal readers showing a classical functional asymmetry, while dyslexic adults show an absence of, or reduced functional asymmetry between MOC bundles. These results will be discussed in the context of current models of APD and Dyslexia.

#### 10:20-10:40 Break

#### 10:40

**4aPPb6. Training of English vowel perception by Finnish speakers to focus on spectral rather than durational cues.** Maria Uther (Brunel University, School of Social Sciences, UB8 3PH Uxbridge, UK, maria.uthers@brunel.ac.uk), Sari Ylinen (Cognitive Brain Research Unit, University of Helsinki, PO BOX 9, 00014 Helsinki, Finland, sari.ylinen@helsinki.fi), Antti Latvala (Cognitive Brain Research Unit, University of Helsinki, PO BOX 9, 00014 Helsinki, Finland, antti.latvala@helsinki.fi), Reiko Akahane-Yamada (ATR Promotions, Human Information Sciences Laboratory, -2-2 Hikaridai Keihanna Science City, Seika-cho, Soraku-gun, 619-0288 Kyoto, Japan, yamada@atr.jp), Paul Iverson (University College London, Department of Phonetics and Linguistics, 4, Stephenson Way, NW1 2HE London, UK, p.iverson@ucl.ac.uk)

This study used the High-Variability Phonetic Training (HVPT) technique to train Finnish speakers to distinguish English vowels. It was found that Finnish speakers tend to use durational cues (which are phonemically relevant in their own language) to make a vowel category distinction rather than the relevant spectral cues. We used duration-modified stimuli with a HVPT program to 'force' the use of spectral cues. We focused on the /i/ (as in 'feet') vs /I/ (as in 'fit') vowel contrast and tested behavioural performance using a perceptual identification task. We also measured the Mismatch negativity (MMN) component of auditory event-related potential (ERP) before and after the training to look at changes in brain responses. The worst pre-test performance was for the 'modified duration' condition (i.e. where the learner had no choice but to rely on spectral cues). There were also asymmetries in vowel perception in both behavioural and MMN tasks, with the detection of /i/ being more difficult compared to the detection of /I/. Nevertheless, training did result in marked improvement of the most problematic contrasts for Finnish speakers.

#### 11:00

**4aPPb7. Effects of voice familiarity and age on perceptual organization of sound from two competing talkers.** Ingrid S. Johnsrude (Queen's University, Dept Psychology, 62 Arch Street, Kingston, ON K7L 3N6, Canada, ingrid.johnsrude@queensu.ca), Allison Mackey (The Rotman Research Institute, Baycrest, 3560 Bathurst Street, 938, Toronto, ON M6A 2E1, Canada, allison.mackey@hotmail.com), Elizabeth M. Alexander (Queen's University, Dept Psychology, 62 Arch Street, Kingston, ON K7L 3N6, Canada, 3ema@queensu.ca), Heather Macdonald (Queen's University, Dept Psychology, 62 Arch Street, Kingston, ON K7L 3N6, Canada, hpmacdonald@gmail.com), Robert P. Carlyon (MRC CBU, 15 Chaucer Rd., CB2 7EF Cambridge, UK, bob.carlyon@mrc-cbu.cam.ac.uk)

We used the Coordinate Response Measure (CRM) procedure (Bolia et al, 2000) to examine whether a non-acoustic characteristic of speech -voice familiarity- can affect a listener's ability to separate competing voices. We tested 27 listeners, aged 45-79, with their spouse's and two novel voices (other listeners' spouses). Couples had been living together at least 18 years. On each trial, two different talkers produced two of four callsigns (one being the target 'Baron'), two of four colours and two of eight numbers, and the participant responded by indicating the colour-number combination to which 'Baron' was told to go. We tested three conditions: spouse-voice target phrase with a novel-voice masker phrase (F/N); novel-voice target with



spouse-voice masker (N/F); novel-voice target with (different) novel-voice masker (N/N). Compared to N/N, performance was significantly better in F/N, and, crucially, N/F conditions, indicating that listeners can exploit learned characteristics of a masking voice to help them track a novel voice. Furthermore, whereas the younger participants (under 60) benefited from having a familiar voice as target or as masker, the older group (over 60) benefited only when the familiar voice was the focus of attention, suggesting that the ability to use voice familiarity to segregate sounds changes with age.

11:20

**4aPPb8. Detection threshold and prominence of background music with interfering speech.** Mark M. Houben (TNO, Kampweg 5, P.O. Box 23, 3769 ZG Soesterberg, Netherlands, mark.houben@tno.nl)

Spoken radio and television messages are often accompanied by background music. When automating identification of broadcast music for handling copyright exploitation, it may be questioned to what extent music at very low sound levels should be taken into account. Should it just be perceivable by listeners who attend to the speech, or should the music be present more prominently? We studied detection and level of prominence of music with interfering speech. For this, Dutch radio and television broadcasts were recorded and speech fragments without background music were selected. To the fragments, music was added with various level differences between speech and music. Subjects were presented with these stimuli and answered a random generic question about the speech after each stimulus. As a second task, the prominence of the background music had to be scored on a scale (including 'not present'). Results show that, on average, 50% of the music is detected if the level of the music is 45 dB lower than the speech. A threshold based on a criterion of 'moderate prominence' results in a level difference of 26 dB. This level happens to coincide with a detection threshold of 95%.

11:40

**4aPPb9. Emotion and meaning in interpretation of sound sources.** Penny Bergman (Chalmers University of Technology, Division of Applied Acoustics - Chalmers Room Acoustics Group, Sven Hultins gata 8a, 41296 Gothenburg, Sweden, penny@chalmers.se), Daniel Vastfjäll (Chalmers University of Technology, Division of Applied Acoustics - Chalmers Room Acoustics Group, Sven Hultins gata 8a, 41296 Gothenburg, Sweden, daniel.vastfjall@psy.gu.se), Niklas Fransson (Chalmers University of Technology, Division of Applied Acoustics, Sven Hultinsgata 8a, 41296 Gothenburg, Sweden, niklas.fransson@psy.gu.se), Anders Sköld (Chalmers University of Technology, Division of Applied Acoustics, Sven Hultinsgata 8a, 41296 Gothenburg, Sweden, anders.skold@chalmers.se)

Research regarding the perception of sound focuses in large on the acoustical properties of the sound. We argue that, for a more complete picture of sound perception, one must take the non-physical properties into

account. By changing the emotional descriptor of a sound the perception in terms of level of annoyance will change. The present study investigates how a priming picture placing the origin of the sound in either a positive or negative environment affects the level of annoyance to same sound. Three different sounds were used in the experiment, all based on pink noise. The participants were, in the beginning of each sound, exposed to a picture telling where the sound originated. The picture was either a positive environment (a picture of a waterfall) or a negative environment (a picture of a larger factory). While listening to the sounds the participants completed different performance tasks. In the end of each sound the participants rated the level of annoyance. Results show that the annoyance ratings are significantly lower when primed with a positive picture. Results also indicate that for more attention demanding tasks this correlation is stronger. The findings are discussed in relation to theories of sound perception.

12:00

**4aPPb10. Fast Detection for Natural Animal Sounds.** Clara Suied (CNRS - UPMC UMR 7593, Pavillon Clérambault, Hôpital de la Salpêtrière, 47, Bd de l'Hôpital, 75013 Paris, France, clara.suied@ircam.fr), Marie Magnin (IRCAM - UMR CNRS 9912, Equipe Perception et Design Sonores, 1 place Igor Stravinsky, 75004 Paris, France, magnin.marie@gmail.com), Sabine Langlois (Renault, 1 avenue du golf, 78288 Guyancourt, France, sabine.langlois@renault.com), Patrick Susini (IRCAM - UMR CNRS 9912, Equipe Perception et Design Sonores, 1, place Igor Stravinsky, 75004 Paris, France, susini@ircam.fr), Stephen McAdams (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, smc@music.mcgill.ca)

Human listeners seem remarkably good at identifying complex and natural sounds. Relatively little attention, however, has been paid to the simple detection of such sounds. The present experiments measured the time course of sound detection in human listeners. Two categories of sounds were used: a train of 1-kHz pure tone pulses varying along an Inter Onset Interval dimension (termed here "IOI" sounds) and animal call sound samples. All sounds were equalized in loudness in a preliminary experiment. The task consisted of a simple reaction time, in which listeners were presented with interleaved IOI and animal sounds and had to manually press a button as fast as possible when they heard any stimulus. Two main results were observed. First, reaction times were significantly shorter for IOIs of 50, 33 and 25 ms than for an IOI of 100 ms, with no significant differences between IOIs of 33 and 25 ms, highlighting a threshold at 33 ms. Second, average reaction times were significantly shorter for animal sounds than for IOI sounds. Differences in terms of spectral content and temporal modulation characteristics might explain part of these effects. These results also suggest a fast detection advantage for natural animal sounds.

12:20-2:00 Lunch Break

## Contributed Papers

2:00

**4aPPb11. The effect of postural information on the perceived velocity of moving sound sources.** Shuichi Sakamoto (R.I.E.C., Tohoku University, 2-1, Katahira, Aoba-ku, 980-8577 Sendai, Japan, saka@ais.riec.tohoku.ac.jp), William L. Martens (McGill University, Schulich School of Music, 555 Sherbrooke Street West, Montreal, QC H3A 1E3, Canada, wlm@music.mcgill.ca), Yôiti Suzuki (R.I.E.C., Tohoku University, 2-1, Katahira, Aoba-ku, 980-8577 Sendai, Japan, yoh@ais.riec.tohoku.ac.jp)

In this study, the effect of postural information on the perceived velocity of moving sound sources was investigated. Auditory information was presented via five surround speakers and five subwoofers. Two sounds were presented first in front of participants, and then these moved past them toward the rear. At the same time, rearward pitch motion was presented via a

motion platform on which participants were situated. Duration of stimulus was 8 s. Initial distance of sound sources was set to 2.0, 4.0, 6.0, 8.0 and 10.0 m. Amplitude of pitch variation was 0, 0.2, 0.4, 1.0 and 2.0 degrees. Participants made magnitude estimates for speed of moving sound images, and judgment of goodness of movement matching between both auditory and postural information. The resulting speed magnitude estimates showed that pivot magnitude significantly affected the estimated velocity of sound sources except when velocity of sound sources was extremely high. Moreover, participants judged the multimodal match to be poor when the velocity of the moving sound sources was extremely high or low. These results suggest that the strongest multimodal interaction occurs when auditory information and postural variation are well matched, and are consistent with self motion.



2:20

**4aPPb12. Auralization of an orchestra with phase-shifted string sections.** Michelle C. Vigeant (Univ. of Nebraska - Lincoln, Architectural Eng. Program, 1110 S. 67th St., Omaha, NE 68182-0681, USA, michelle.vigeant@gmail.com), Lily M. Wang (University of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0681, USA, LWang4@UNL.edu)

Orchestra auralizations have been created in ODEON using multi-channel individual instrument anechoic recordings of two symphonies; however, only one or two string instruments were recorded to represent each string section. To simulate the chorus effect of an entire string section more accurately, the anechoic tracks of the single string instruments have been mixed with other versions of the same signal, each with some phase shift in time. Two groups of phase shifts were used: one with shorter delays of up to 23 ms, and one with longer delays of up to 47 ms. A maximum of seven differently phase-shifted signals were combined with the original to create a final anechoic recording for use in the auralizations, depending on the number of players each source represented. Using paired comparisons, test subjects were asked to identify the auralization that sounded most similar to the experience of listening to an orchestra in an actual concert hall: one having none, short or long phase-shifts. Results show that subjects have difficulty differentiating between these three types of auralizations, indicating that phase shifting may not be required for such multi-source multi-channel orchestra auralizations. [Work supported by the National Science Foundation.]

2:40

**4aPPb13. Individual differences in auditory localization of real sources in the horizontal plane.** Sophie Savel (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, savel@lma.cnrs-mrs.fr)

Recent studies on spatial hearing showed that individual variability is greater with virtual sources than with real ones, but that sensorial feedback provides an efficacious and rapid learning procedure. This study evaluated the degree of inter-individual variability with real sound sources in the absence of feedback. Fifty normal-hearing listeners, aged 18-62, either left- or right handed, participated. The experiment was conducted in an anechoic room. One of twelve loudspeakers placed in the frontal horizontal plane and hidden by a curtain emitted a train of low-pass pulses. Listeners had to indicate the estimated direction of that source by placing a pointer on a screen. No feedback was given to them. Each listener completed 480-720 trials. Results indicated that variability between individual was great in both resolution in bias. Precisely, individuals differed (1) in the size of their maximum error, (2) in the azimuthal region in which this maximum error occurred, and (3) in the spatial symmetry of their performance. Indeed, 25% of the listeners showed significant left-right differences in their performance, these differences always favoring the left side of space. Such asymmetries have been noticed earlier but attributed to greater front-back confusions on the right, which cannot explain the present results.

3:00

**4aPPb14. Investigating effects of spatially disparate visual stimuli on auditory localization in VR environments.** Khoa-Van Nguyen (IRCAM, 1 Place Igor Stravinsky, 75004 Paris, France, nguyen@ircam.fr), Clara Suied (CNRS - UPMC UMR 7593, Pavillon Clérambault, Hôpital de la Salpêtrière, 47, Bd de l'Hôpital, 75013 Paris, France, clara.suied@ircam.fr), Isabelle Viaud-Delmon (CNRS - UPMC UMR 7593, Pavillon Clérambault, Hôpital de la Salpêtrière, 47, Bd de l'Hôpital, 75013 Paris, France, ivd@ext.jussieu.fr), Olivier Warusfel (IRCAM, 1 Place Igor Stravinsky, 75004 Paris, France, Olivier.Warusfel@ircam.fr)

Investigating the time and spatial constraints under which visual and auditory stimuli are perceived as a unique percept or as spatially coincident has been a topic of numerous researches in neuroscience. However, these findings have been derived up to now in extremely simplified stimulation context consisting in the combination of elementary auditory and visual stimuli usually displayed in dark and anechoic conditions. The present experiment is conducted in a VR environment using a stereoscopic passive screen and binaural audio rendering. Auditory stimuli are displayed on headphones using individualized head-related transfer functions and visual stimuli are integrated in a visual background in order to convey visual perspective. The experiment investigates the effect of a spatially disparate visual stimulus on the auditory localization judgments (crossmodal bias), as well as the relation between the magnitude of the crossmodal bias and the perception of a unified bi-modal stimulus. The present study will indicate whether previous findings (Hairston et al., Journal of Cognitive Neuroscience, 2003) still hold in more complex audio-visual contexts such as those offered by VR environments.

3:20

**4aPPb15. The Role of Auditory-Visual Integration in Object Recognition.** Clara Suied (CNRS - UPMC UMR 7593, Pavillon Clérambault, Hôpital de la Salpêtrière, 47, Bd de l'Hôpital, 75013 Paris, France, clara.suied@ircam.fr), Nicolas Bonneel (REVES/Inria Sophia-Antipolis, 2004 route des lucioles, BP 93, FR-06902 Sophia Antipolis, France, Nicolas.Bonneel@sophia.inria.fr), Isabelle Viaud-Delmon (CNRS - UPMC UMR 7593, Pavillon Clérambault, Hôpital de la Salpêtrière, 47, Bd de l'Hôpital, 75013 Paris, France, ivd@ext.jussieu.fr)

Combining auditory and visual information about the same external event enhances perception and behavioural performance. Numerous factors have been shown to contribute to the integration of visual and auditory stimuli, like spatial or semantic relationships between the two stimuli. We studied the influence of spatial disparity between the auditory and the visual stimuli on bimodal object recognition in a go/no-go task, under realistic virtual environment. Participants were asked to react as fast as possible to a target object, presented in the visual and/or the auditory modality, and to inhibit a distractor object. Reaction times were significantly shorter for semantically congruent bimodal stimuli than would be predicted by independent processing of information about the auditory and the visual targets. Moreover, reaction times were significantly shorter for semantically congruent bimodal stimuli (i.e. visual and auditory targets) than for semantically incongruent bimodal stimuli (i.e. target represented in only one sensory modality). Importantly, these results were not altered by a large spatial disparity between the auditory and the visual targets. Altogether our findings suggest that rules governing multisensory integration vary according to the purpose for which auditory and visual stimuli are combined.

### 3:40-5:20 Posters

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

5:20

**4aPPb16. Why is sharp-limited low-frequency noise extremely annoying?** Detlef Krahe (Univ. of Wuppertal, Rainer-Gruenter-Str. 21, 42119 Wuppertal, Germany, krahe@uni-wuppertal.de)

Sharp-limited low-frequency noise having only weak components of higher frequencies, such as noise of an air condition or traffic noise attenuated by a thick window, has a very annoying effect on persons also at low levels. The strong fluctuation, which is specific to this kind of sound, is a frequently used explanation for this effect possibly caused by adaptation in the inner ear. Another or additional explanation could be a strong synchronism in the activities on the nerve fibers. Computer models of the auditory system show this synchronism. If some components at higher frequencies are added, the synchronism disappears and the noise is judged less uncomfortable. This raises the question, if noise protection resulting in a sound as described can not be even counterproductive. Differently sharp-limited sounds are investigated by an auditory model. The results are compared with results of judgment by hearing.

5:40

**4aPPb17. Comparison of subjective and objective evaluation methods for audio source separation.** Josef Kornycky (I-Lab Multimedia and DSP Research Group, Centre for Communication Systems Research, University of Surrey, GU2 7XH Guildford, UK, J.Kornycky@surrey.ac.uk), Banu Gunel (I-Lab Multimedia and DSP Research Group, Centre for Communication Systems Research, University of Surrey, GU2 7XH Guildford, UK, B.Gunel@surrey.ac.uk), Ahmet Kondo (I-Lab Multimedia and DSP Research Group, Centre for Communication Systems Research, University of Surrey, GU2 7XH Guildford, UK, A.Kondo@surrey.ac.uk)

The evaluation of audio separation algorithms can either be performed objectively by calculation of numerical measures, or subjectively through listening tests. Although objective evaluation is inherently more straightforward, subjective listening tests are still essential in determining the perceived quality of separation. This paper aims to find relationships between objective and subjective results so that numerical values can be translated into perceptual criteria. A generic audio source separation system was modelled which provided varying levels of interference, noise and artifacts. This enabled a full spread of objective measurement values to be obtained. Extensive tests were performed utilising the output synthesised by this separation model. The relationships found were presented and the factors of prime importance were determined.

6:00

**4aPPb18. An exploration of attentional monitoring of isochronous asynchronous streams in deviant detection and sensorimotor synchronization.** Martine Turgeon (Lancaster University, Psychology Department, Fylde College, LA1 4YF Lancaster, UK, m.turgeon@lancaster.ac.uk)

Past studies of auditory selective attention focused on location and/or pitch cues in stream segregation. In this study, two perceptually-segregated streams competed for attention in a perceptual-judgment task based on timbre cues and a sensorimotor task (tapping to the sounds of one stream). Both streams shared the same temporal structure (isochronous 100-ms events at 1.67 Hz) but were asynchronous (target-distractor asynchronies of -200, -100, 100, 200 ms), differed in pitch (712 or 1000 Hz) and could share or not their perceived location (diotic vs. dichotic). Deviants were amplitude modulated at 25 or 50 Hz and varied in temporal envelopes. Attentional filtering (distracting stream constant) was done using a discrimination task requiring pressing one of two keys upon hearing one of two deviants (Experiment 1) and a synchronization task requiring selective tapping to the

high-pitch or low-pitch sounds (Experiment 3). Attentional monitoring (distracting stream varies) required maintaining attention to the same stream upon hearing slow-modulated deviants and switching attention to the other stream upon hearing the fast-modulated deviants. This was done for the discrimination task (Experiment 2) and synchronization task (Experiment 4). Perceptual and perceptuo-motor results suggest attentional-filtering and attentional-monitoring costs. Surprisingly, the mere presence of a non-concurrent stream in a different frequency band interferes with deviant-discrimination and synchronization-tapping performance.

6:20

**4aPPb19. Normalization in count-comparison model of interaural time difference decoding.** Ville Pulkki (Helsinki University of Technology, P.O. Box 5400, 02015 TKK, Finland, Ville.Pulkki@tkk.fi)

Recent neurophysiological studies suggest that binaural decoding is based on count comparison for both ITD and ILD. In such mechanisms, the neural signals are stronger in the auditory pathways leading to the ipsilateral hemisphere when a signal is presented earlier, or with higher level, to the contralateral ear. A computational model is described implementing binaural cue decoding based on count-comparison principles for ITD decoding, which is assumed to occur in medial superior olive (MSO). Pooled response of MSO is modeled as running multiplication between inputs, which are derived from ear canal signals with GTFB filtering and phase-locked impulse generator. The contralateral and ipsilateral inputs to MSO are then convolved with different responses. The model output corresponds well to neurophysiological results. In the earlier version of the model, the MSO output was normalized only with the contralateral signal, as suggested by the neuroanatomy. It has been later found out that the output of MSO model depends on ILD, which is in contradiction with psychoacoustics studies. In this study, it is proposed that the ipsilateral input to MSO is self-normalized, which provides ILD-independent ITD decoding.

6:40

**4aPPb20. Representation of Harmonic Sounds in the Helix of the Lateral Lemniscus.** Gerald Langner (Neuroacoustics, TUD, Zool. Institut, Schnittpahnstr. 3, 64287 Darmstadt, Germany, gl@bio.tu-darmstadt.de), Claudia Simonis (Schnittpahnstr. 3, 64287 Darmstadt, Germany, simonis@bio.tu-darmstadt.de), Antje Sauck (Schnittpahnstr. 3, 64287 Darmstadt, Germany, antjesalg@freenet.de), Ralf Galuske (Schnittpahnstr. 3, 64287 Darmstadt, Germany, galuske@mpih-frankfurt.mpg.de)

The percept of pitch of harmonic sounds is based on temporal processing. This explains also our ability to recognize harmonic relationships between different sounds, because as a result of a neuronal correlation analysis in the auditory midbrain neurons are tuned not only to a certain pitch but to a certain degree also to integer multiples of that pitch. The responses to harmonics of a pitch are suppressed within 30 ms after signal onset by inhibition with the likely source being the ventral nucleus of the lateral lemniscus (VNLL). An investigation of spatial representation of periodicity information with the 2-Deoxyglucose method in gerbils showed that low pitch is represented dorsally and high pitch ventrally along the length-axis of the VNLL. Three-dimensional computer reconstructions of the VNLL (program AMIRA) gave evidence for a helical periodicity map with 7 to 8 turns, reminiscent of the pitch helix known from music psychology. Moreover, the spatial organization of the VNLL suggests that it is organized as a double-helix representing musical octaves and fifths. Reconstructions of the VNLL of Nissl-stained human brains gave evidence of a similar organization and therefore of a similar functional role of the VNLL for pitch processing in humans.

## Session 4aSAa

**Structural Acoustics and Vibration, Computational Acoustics, and EURONOISE: Fluid—Structure Interaction II**

Nouredine Atalla, Cochair

*Univ. de Sherbrooke, Mechanical Engineering Depart., 2500 Boulevard de l'Université, Sherbrooke, QC J1K 2R1, Canada*

Vicente Cutanda Henriquez, Cochair

*Institute of Sensors, Signals and Electrotechnics, University of Southern Denmark, Niels Bohrs Allé 1, Odense S, 5230, Denmark*

Stefan Schneider, Cochair

*Laboratoire de Mécanique et d'Acoustique, UPR 7051 CNRS, 31, Chemin Joseph-Aiguier, Marseille, 13402, France***Contributed Papers**

8:00

**4aSAa1. Fluid-structure interaction and computational aeroacoustics of the flow past a thin flexible structure.** Frank Schäfer (University Erlangen-Nuremberg, Institute of Fluid Mechanics, Cauerst. 4, 91058 Erlangen, Germany, frank.schaefer@lstm.uni-erlangen.de), Thomas Uffinger (University Erlangen-Nuremberg, Institute of Fluid Mechanics, Cauerst. 4, 91058 Erlangen, Germany, thomas.uffinger@lstm.uni-erlangen.de), Stefan Becker (University Erlangen-Nuremberg, Institute of Fluid Mechanics, Cauerst. 4, 91058 Erlangen, Germany, stefan.becker@lstm.uni-erlangen.de), Jens Grabinger (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, jens.grabinger@lse.eei.uni-erlangen.de), Manfred Kaltenbacher (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, manfred.kaltenbacher@lse.eei.uni-erlangen.de)

In many technical applications the interaction between a fluid flow and a thin flexible structure leads to the generation of acoustic noise which is caused by flow induced structural vibrations. Examples for such applications are coverings and panelings of cars and airplanes. In many cases the generated noise is unwanted so that noise reduction is a topic of major interest. In the present work we investigate the acoustic field resulting from the interaction of a thin flexible structure with a turbulent flow field by means of numerical simulation. Two different model configurations are considered: one is the flow over a flexible plate, in the second case the flexible plate is located in the wake of a square cylinder. The major aim of this work is to provide a better understanding of the noise generation processes in these flow cases. The numerical methodology applied is utilized for a decomposition of the acoustic field into one part generated by the structural vibrations and another part which is due to stream noise. Finally, comparisons to experimental data available at our institute are provided.

8:20

**4aSAa2. An Equivalent-Acoustic Finite Element Method for Modeling Sound Absorbing Materials.** Donald J. Nefske (General Motors R&D Center, Vehicle Development Research Laboratory, 30500 Mound Rd., Warren, MI 48090-9055, USA, DNefske@aol.com), Shung H. Sung (General Motors R&D Center, Vehicle Development Research Laboratory, 30500 Mound Rd., Warren, MI 48090-9055, USA, Shung.H.Sung@gm.com)

An equivalent-acoustic finite element method is developed for modeling sound absorbing materials, such as seats and interior trim in the automobile passenger compartment. The equivalent-acoustic method represents the sound absorbing material using acoustic finite elements with frequency-dependent material properties determined from the measured acoustic impedance of sound absorbing material samples. Solution of the equivalent-acoustic model within the Nastran computer capability and coupling of the

model with an acoustic finite element model of a surrounding enclosure, such as the passenger compartment, are developed. The accuracy of the equivalent-acoustic method is assessed for modeling a sound absorbing material in a one-dimensional impedance tube, a foam layer in a rectangular box enclosure, and an automotive seat in a semi-reverberant enclosure.

8:40

**4aSAa3. Finite element modelling of transient elastic wave propagation in an inhomogeneous anisotropic fluid/solid multilayer medium: A time-domain method.** Guillaume Haiat (CNRS, Laboratoire de Recherches Orthopédiques, 10, Avenue de Verdun, 75010 Paris, France, haiat@univ-paris12.fr), Salah Naili (Université Paris 12, B2OA, 61, Avenue du Général de Gaulle, 94010 Créteil, France, naili@univ-paris12.fr), Quentin Grimal (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, quentin.grimal@lip.bhdc.jussieu.fr), Maryline Talmant (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, talmant@lip.bhdc.jussieu.fr), Christophe Desceliers (Université de Marne la Vallée, 5, Boulevard Descartes, 77454 Marne la Vallée, France, christophe.desceliers@univ-mlv.fr), Christian Soize (Université de Marne la Vallée, 5, Boulevard Descartes, 77454 Marne la Vallée, France, soize@univ-mlv.fr)

The axial transmission technique is used clinically for cortical bone assessment. However, ultrasonic propagation in this multiscale medium remains unclear, in particular because of the heterogeneous nature of cortical bone. The aim of this work is to evaluate the effect of spatial gradients of elastic moduli on the ultrasonic response of the bone structure. Therefore, a 2D finite element time-domain method is developed to simulate transient wave propagation in a three-layer medium constituted of an inhomogeneous transverse isotropic solid layer sandwiched between two acoustic fluid layers and excited by an acoustic linear source located in one fluid layer delivering broadband ultrasonic pulses. The model couples the acoustic propagation in both fluid media with the elastodynamic response of the solid. The conditions of continuity are used to model the fluid-structure interaction. A constant spatial gradient of material properties in the direction perpendicular to the layer is considered in the solid structure. In the presence of a gradient, the first arriving signal (FAS) velocity depends on the average material properties when the thickness is smaller than the wavelength (guided wave modes) whereas the FAS velocity depends on the velocity at the surface when the thickness is larger than the wavelength (lateral wave).



9:00

**4aSAa4. Dynamic behavior analysis of vehicle acoustic trim using FEM.** Norimasa Kobayashi (Toyota Motor Corporation, 1, Toyota-cho, 471-8572 Toyota, Aichi, Japan, koba@nori.tec.toyota.co.jp), Yoshihisa Uno (Toyota Motor Corporation, 1, Toyota-cho, 471-8572 Toyota, Aichi, Japan, yoshi@uno.tec.toyota.co.jp), Hiroo Yamaoka (Toyota Motor Corporation, 1, Toyota-cho, 471-8572 Toyota, Aichi, Japan, yamaoka@giga.tec.toyota.co.jp)

Multilayer vehicle acoustic trim containing poroelastic materials affects noise and vibration phenomena not only in the high-frequency range but also in the low and mid-frequency range. However, there are few established technology to analyze the vehicle model which includes the characteristics of the trim. In this paper, a methodology for the analysis of multilayer acoustic trim within a vehicle FEM model is derived based on Biot theory. Using this methodology, dynamic vibration, radiation and absorption behavior of three types of multilayer trim including conventional isolative type, absorptive type and isolative type with absorptive top layer backed by rectangular metal plates are analyzed. The trim model is also applied to a vehicle FEM model and its effects on interior sound pressure level are validated with experimental data.

9:20

**4aSAa5. Noise reduction prediction of Ariane 5 fairing with acoustic protection made of porous-elastic material.** Slaheddine Frikha (ESI France, 99 rue des Solets, Parc des affaires SILIC - BP 80112, 94513 Rungis, France, slaheddine.frikha@esi-group.com), Mohamed Ali Hamdi (ESI GROUP & UTC, 20, Rue Du Fonds Pernant, 60471 Compiègne, France, Mohamed-Ali.Hamdi@esi-group.com), Philippe Roux (CNES, Rond Point de L'Espace, 91 023 Courcouronnes, France, philippe.roux@cnes.fr), Lassen Mebarek (ESI GROUP & UTC, 20, Rue Du Fonds Pernant, 60471 Compiègne, France, Lassen.Mebarek@esi-group.com), Abderrazak Omrani (ESI GROUP & UTC, 20, Rue Du Fonds Pernant, 60471 Compiègne, France, Abderrazak.Omrani@esi-group.com)

ESI Group (Formerly STRACO) has been involved during the last two decades in the numerical prediction of noise reduction index of ARIANE 5 fairing. During the 1990's, STRACO developed an axi-symmetric, boundary element model of the fairing where the fairing protection made of distributed Helmholtz Resonator is modeled by equivalent local impedance. Recently, the fairing acoustic protection has been replaced by a foam-made insulator. Such porous materials are widely used by transportation industries to improve the payload acoustic comfort. In collaboration with automotive industry, ESI group developed RAYON-VTM, a powerful tool allowing the predicting of fully trimmed vehicle vibroacoustic response up to 500 Hz. RAYON VTM model the porous-elastic material using a 3D Finite Element (PEM), based on the Modified-Biot-Equations. This new module of RAYON software has been applied to predict the vibroacoustic response of the ARIANE 5 fairing allowing a detailed modeling of the acoustic protection. A fully 3-D model of the fairing has been developed. The results show the influence of intrinsic modeling of the porous-elastic protection as well as the influence of non-axi-symmetric details usually neglected in the axi-symmetric approach.

9:40

**4aSAa6. Influence of solid phase elasticity in poroelastic liners submitted to grazing flows.** Benoit Nennig (Université de Technologie de Compiègne, Lab. Roberval UMR 6253, Dept. Acoustique, BP 60319, 60203 Compiègne, France, benoit.nennig@utc.fr), Jean-Daniel Chazot (Université de Technologie de Compiègne, Lab. Roberval UMR 6253, Dept. Acoustique, BP 60319, 60203 Compiègne, France, jean-daniel.chazot@utc.fr), Emmanuel Perrey-Debain (Université de Technologie de Compiègne, Lab. Roberval UMR 6253, Dept. Acoustique, BP 60319, 60203 Compiègne, France, emmanuel.perrey-debain@utc.fr), Mabrouk Ben Tahar (Université de Technologie de Compiègne, Lab. Roberval UMR 6253, Dept. Acoustique, BP 60319, 60203 Compiègne, France, mabrouk.bentahar@utc.fr)

In the present work, we study the sound propagation in a duct treated with a poroelastic liner exposed to a grazing flow. Acoustic propagation in

the liner and in the fluid domain is respectively governed by Biot's model and Galbrun's equation. Here, the coupling between Galbrun's and Biot's equation is carried out with a mixed pressure-displacement FE. On one hand, a mixed formulation is used in Galbrun's equation to avoid numerical locking. And on the other hand, in poroelastic media, the description of both phases involves the displacement of the solid phase and the pressure in the fluid phase. In addition of using the complete Biot's model, simplified models are also tested. A fluid equivalent model that does not take into account solid phase elasticity and a model that neglects only the shear stress are hence used. These two simplified models enable to evaluate the contributions of the compressional and shear waves in the solid phase. Finally, validity of each simplified model in the specific case of a liner exposed to a grazing flow is discussed for different materials.

10:00-10:20 Break

10:20

**4aSAa7. An Evolutionary Optimization Method applied to Absorbing Poroelastic Systems.** Francisco Silva (Unité de Mécanique (UME), ENSTA, Chemin de la Hunière, 91761 Palaiseau, France, francisco-ilson.dasilva@ensta.fr), Renato Pavanello (Depart. of Computational Mechanics - FEM - Unicamp, POBOX 6122, Rua Mendeleiev, s/n - Cidade Universitária, 13083-970 Campinas, Brazil, pava@fem.unicamp.br)

Poroelastic materials can be used in engineering applications such as: noise control of automobiles, acoustical insulation systems for aircrafts, industrial, environmental and domestic sound quality control, etc. The insulating systems must be as light as possible and the acoustic absorption in the low frequency domain must be maximized for certain gaps. Topology Optimization is frequently used to design structures and acoustic systems in a large range of engineering applications. In this work, we propose one method to maximize the absorbing performance of insulation poroelastic systems using a coupled finite element model and Evolutionary strategies. The goal is find the best distribution of poroelastic material on insulating systems. The absorbing performance of the poroelastic material samples in a Kundt tube is simulated using a coupled poroelastic and acoustic finite element model. The Biot-Allard Coupled Model is used to represent the foam material. The porous material model is coupled to a waveguide using a modal superposition technique. A sensitivity number indicating the optimum locations for porous material removing is derived and used in a numerical hard kill scheme. The sensitivity number is used to form an evolutionary poroelastic optimization (EPO) algorithm which is verified through numerical examples.

10:40

**4aSAa8. Acoustic frequency wave propagation and fluid-loading in composite sandwich panels.** Jennifer F. Crossley (Swansea University Engineering Department, Singleton Park, SA2 0DS Swansea, UK, 437441@swansea.ac.uk), Johann Sienz (Swansea University Engineering Department, Singleton Park, SA2 0DS Swansea, UK, J.Sienz@swansea.ac.uk)

Composite sandwich panels comprising glass fibre reinforced epoxy skins with nomex honeycomb cores have found application as aircraft interior fixtures and partitions. This is due to their low density combined with static strength and stiffness. In this complex acoustic and vibration environment however, the same properties can lead to unwanted sound transmission through the structure. We analyse, in this paper, fluid-loading of such a panel by an acoustic fluid using an elementary theory which has been developed in the study of structural dynamics and vibration, based on a generalisation of Timoshenko theory for homogeneous beams. The dispersion relations derived from this theory, with and without fluid-loading terms, are used to quantify the effect of fluid-loading by air on shear and flexural waves at acoustic frequencies in such stiff, lightweight structures. The most appropriate method of fluid-structure coupling to be applied in modelling an internal acoustic field enclosed by such a structure is discussed.

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11:00

**4aSAa9. A formulation based on modal optimization for predicting sound radiation from fluid-loaded aircraft structures.** Olivier Coltery (Airbus France - Acoustics and Environment Department, 316 route de Bayonne, F-31060 Toulouse Cedex 09, France, olivier.coltery@insa-lyon.fr), Jean-Louis Guyader (INSA de Lyon - LVA, Bâtiment St. Exupéry, 25 bis avenue Jean Capelle, F-69621 Villeurbanne Cedex, France, jean-louis.guyader@insa-lyon.fr)

This study is led in the context of understanding sound transmission through fluid-loaded aircraft structures with non-uniform damping. The fluid/structure coupling is here highlighted. Classic approaches using modal radiation impedances to formulate the fluid/structure coupling do lead to complexity and are computationally time-consuming. A method defining

rigorously this coupling and avoiding the calculation of the modal radiation impedances is proposed. This method aims at developing a simple formulation taking advantage of current technical progress in optimization algorithms. Sound radiation from a simply supported, baffled, fluid-loaded plate excited mechanically or acoustically is here solved in optimizing the modal amplitudes so that they fit the governing equation with fluid loading. To perform this optimization, a sampling of the plate into observation points is first done, then a modal decomposition into in vacuo modes is led. Comparison with results from the literature over [10-1000Hz] for 3 reference cases of steel plate immersed in air (critical frequencies equal to  $f_{C1}=1.2$  kHz and  $f_{C2}=12$  kHz) show excellent agreement within 1dB. The simplicity and computation time allow an extension to non-uniform damped aircraft structures and a prediction over a large frequency band. As perspectives, results from multi-layered plates with a local damping patch are presented.

THURSDAY MORNING, 3 JULY 2008

ROOM 252B, 11:40 A.M. TO 7:00 P.M.

### Session 4aSAb

## Structural Acoustics and Vibration and EURONOISE: Distributed Active Noise and Vibration Control I

Kenneth Cunefare, Cochair

*Woodruff School of Mech. Eng., The Georgia Inst. of Technol., Atlanta, GA 30332-0405, USA*

Manuel Collet, Cochair

*FEMTO-ST UMR CNRS, 23 chemin de l'Épitaphe, Besançon, 25000, France*

### Invited Papers

11:40

**4aSAb1. Smart panels with active wedges.** Paolo Gardonio (ISVR, University of Southampton, Highfield, SO17 1BJ Southampton, UK, pg@isvr.soton.ac.uk), Yohko Aoki (ISVR, University of Southampton, Highfield, SO17 1BJ Southampton, UK, ya@isvr.soton.ac.uk), Stephen J. Elliott (ISVR, University of Southampton, Highfield, SO17 1BJ Southampton, UK, sje@isvr.soton.ac.uk)

This paper presents an experimental study comparing the vibration and sound radiation control performance obtained from six different rectangular panels with various geometries of active wedge. Each panel is equipped with sixteen triangular piezoceramic actuators along the panel border and accelerometer sensors located at the top vertex of the triangular actuators. These sensor-actuator pairs are used to implement decentralized velocity feedback loops that produce active damping on the plate. The primary objective of this paper is to investigate the effect of the size and geometry of the triangular actuators on stability and control performance. Narrow frequency band measurements highlight that vibration reductions at the first few resonant frequencies are significantly improved by increasing the height of the triangular actuators. However, the increase of the height also results in spillover effects at frequencies higher than around 700Hz. In contrast, an increase of the base length of the triangular actuators improves the overall control performance up to 1 kHz without increasing the spillover effect.

12:00

**4aSAb2. A new hybrid passive/active cell to realize a complex impedance boundary condition.** Marie-Annick Galland (Centre Acoustique du LMFA, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, 69134 Ecully cedex, France, marie-annick.galland@ec-lyon.fr), Jean-Baptiste Dupont (Centre Acoustique du LMFA, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, 69134 Ecully cedex, France, jean-baptiste.dupont@ec-lyon.fr)

Hybrid passive/active cells previously developed at the LMFA have proved their efficiency for global noise reduction. A hybrid cell combines passive absorbent properties of a porous layer and active control at its rear face. Active control is mainly used to increase absorption at low frequencies by cancelling the imaginary part of the surface impedance presented by the absorber. A specific decentralized control algorithm allowed realizing large panels to reduce multi-tone noise radiated by a flow duct. Generally, the design of such absorbers starts by determining the optimal impedance for the targeted application, defined as that which, for each frequency, produces the highest reduction. However, in many cases results show that this optimal impedance is complex and that both real and imaginary parts are frequency dependent. This paper investigates the potential of a new hybrid absorber intended to realize any of impedance. The new cell uses one microphone on each side of a resistive cloth. Normal velocity can then be deduced by a simple pressure difference, which allows an easy synthesis of the error signal. First experiments carried out in a standing wave tube are described and different control strategies are examined, allowing us to validate this promising concept.

12:20

**4aSAb3. The performance trade off of decentralised, distributed and centralised controllers.** Oliver N. Baumann (Institute of Sound and Vibration Research, University of Southampton, Highfield, SO171BJ Southampton, UK, onb@isvr.soton.ac.uk), Kenneth Frampton (ISVR, University of Southampton, Highfield, SO17 1BJ Southampton, UK, kdf@soton.ac.uk), Paolo Gardonio (ISVR, University of Southampton, Highfield, SO17 1BJ Southampton, UK, pg@isvr.soton.ac.uk)

Direct velocity feedback control of structures is well known to increase structural damping and thus reduce vibration. In multichannel systems the way in which the velocity signals are used to inform the actuators ranges from decentralised controller through distributed or clustered controllers to the fully centralised controller. The objective of distributed controllers is to exploit the anticipated performance advantage of the centralised controller whilst maintaining the ease of implementation and robustness of the decentralised controller. It has been observed, however, that in many vibration control systems the centralised controller struggles to perform significantly better than a decentralised controller. This paper compares a number of distributed controllers and optimisation techniques for the reduction of kinetic energy and radiated sound power and identifies the conditions under which the centralised and distributed controllers offer a significant performance advantage.

12:40

**4aSAb4. Reduced models for elastoacoustic problems with intelligent interfaces.** Mohamed Ichchou (Centre Acoustique du LMFA, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, 69134 Ecully cedex, France, mohamed.ichchou@ec-lyon.fr), Aida A. Kacem (Centre Acoustique du LMFA, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, 69134 Ecully cedex, France, aida.hadj-kacem@ec-lyon.fr)

Effective and predictive noise and vibration modelling and design tools as well as efficient and high-performance materials are essential to produce world-leading products with regards to the noise and vibration quality. In one hand, the modern trend towards virtual design and prototyping requires good analysis and modelling tools in the entire noise frequency band. In another hand, new passive, adaptive or active materials offer improved technologies issues for the vibration and noise treatment. However, there is still a lack of efficient numerical tools in the low frequency band, and an absence of tools in the mid-frequency range. Moreover, although new materials potential can be demonstrated in prototype structures, their performances are still insufficient in integrated applications. To bring advances in noise and vibration treatment using new materials, the challenge is to be capable of supporting specific aspects related to advanced materials and virtual prototyping methods. An essential requirement for this is a multi-disciplinary analysis and coupled simulation tools where effects on a very different scale need to be interconnected. The paper proposes reduced models of elastoacoustic problems with localised intelligent interface. The main focus is development of fast simulation tool for the design of new systems. Theoretical as well as numerical experiments are offered in this work.

1:00-2:00 Lunch Break

### *Invited Papers*

2:00

**4aSAb5. Decentralised active control of single - frequency panel vibrations using piezoelectric actuator - sensor pairs.** Alain Berry (Univ. de Sherbrooke, Mechanical Engineering Depart., 2500 Boulevard de l'Université, Sherbrooke, QC J1K 2R1, Canada, alain.berry@usherbrooke.ca), Philippe Micheau (Univ. de Sherbrooke, Mechanical Engineering Depart., 2500 Boulevard de l'Université, Sherbrooke, QC J1K 2R1, Canada, Philippe.Micheau@USherbrooke.ca), Rémi Louviot (Univ. de Sherbrooke, Mechanical Engineering Depart., 2500 Boulevard de l'Université, Sherbrooke, QC J1K 2R1, Canada, remi.louviot@usherbrooke.ca), Yvonnick Brunet (Univ. de Sherbrooke, Mechanical Engineering Depart., 2500 Boulevard de l'Université, Sherbrooke, QC J1K 2R1, Canada, yvonnick.brunet@usherbrooke.ca)

This paper addresses active control of the bending response of a panel using independent PZT (piezoceramic) actuator - PVDF (piezopolymer) sensor pairs distributed on the panel. Previous work showed that under the assumption of collocated, dual actuator - sensor pairs, decentralised static gain control is stable due to plant matrix passivity at all frequencies. However, duality is not guaranteed for collocated PZT - PVDF pairs because of the coupling of piezoelectric transducers with both bending and extensional modes of the panel. Moreover, the spatially local nature of PZT actuator to PVDF sensor transfers on the panel can lead to a diagonal - dominant FRF matrix but is detrimental to global control of the panel vibration or acoustic radiation; hence, a non-diagonal dominant plant matrix is more likely to result in global control for this problem. In the case of single-frequency disturbance, stability analysis shows that plant matrix passivity is only required at the disturbance frequency and that a phase - shift compensation, identical for all independent units, can ensure stable decentralized control. Guidelines for the design of decentralised PZT - PVDF pairs are provided, with the objective of global vibroacoustic control of a panel.

2:20

**4aSAb6. Broad-Band Vibration Attenuation in Plates With Periodic Arrays of Shunted Piezoelectric Patches.** Massimo Ruzzene (Georgia Institute of Technology, School of Aerospace Engineering, 270 Ferst Drive, Atlanta, GA 30332-0150, USA, massimo.ruzzene@ae.gatech.edu), Luca Airoldi (Dipartimento di Ingegneria Aerospaziale, Via La Masa 34, 20156 Milano, Italy, luca.airoldi@gmail.com)

Periodic arrays of shunted piezoelectric patches are employed for the control of waves propagating over the surface of plate structures. The shunted piezoelectric patches act as sources of impedance mismatch, which yield interference phenomena resulting from the interaction between incident, reflected and transmitted waves produced by the mismatch. The impedance mismatch corresponding to the shunted piezos can be tuned to achieve strong attenuation over frequency bands which are defined by the shunting circuit connected to the patches. Broad-band vibration attenuation can be achieved through the application of series and parallel of multiple resonant

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circuits, or through the implementation of negative capacitance configurations. The ability of plates with periodic shunted piezoelectric patches to attenuate vibrations over extended frequency bands is demonstrated numerically, through Finite Element models of the considered electromechanical structures. The effect of number of patches and of their periodic distributions is investigated together with the analysis and the comparison of performance achieved with various shunting strategies.

2:40

**4aSAb7. Use of head mounted microphone arrays for active control.** Marty Johnson (Virginia Tech, Mechanical Engineering, 143 Durham 0238, Blacksburg, VA 24061, USA, martyj@vt.edu)

There has been an effort recently to create head mounted or user worn microphone arrays. These arrays pose challenges in their design and characterization but can be used for a number of different purposes. Initially these arrays were intended for sound source localization and natural hearing restoration (where the hearing of a user whose ears are obstructed by an encapsulating helmet or headset is restored). Once in place, and assuming the user is wearing headphones, these arrays can be used for a number of different applications including voice isolation for communication, focused listening and noise cancellation. Specifically this paper investigates the use of these arrays for the active control of noise both at the user's ears and in the communications/voice signals sent from the user. A numerical study, using both data generated numerically and experimentally, demonstrates that large reductions in noise can be achieved using adaptive active control methods.

3:00

**4aSAb8. Active control of multimodal tonal noise propagated in circular duct with axial subsonic mean flow until  $M=0.3$ .** Martin Glessier (CNRS - LMA, 31 Chemin Joseph Aiguier, 13009 Marseille, France, glessier@lma.cnrs-mrs.fr), Emmanuel Friot (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, friot@lma.cnrs-mrs.fr), Muriel Winninger (CNRS - LMA, 31 Chemin Joseph Aiguier, 13009 Marseille, France, winninger@lma.cnrs-mrs.fr), Cédric Pinhède (CNRS - LMA, 31 Chemin Joseph Aiguier, 13009 Marseille, France, pinhede@lma.cnrs-mrs.fr), Alain Roure (CNRS - LMA, 31 Chemin Joseph Aiguier, 13009 Marseille, France, roure@lma.cnrs-mrs.fr)

As new generation of aircraft engine with lower blade passing frequency appeared in the 1990's, the fan tones radiated from the inlets had become one of the dominant source of sound. Efforts have then been made to develop active noise control. Encouraging results have been obtained but the physical limitation of the fan tones reduction have not been clearly determined, owing mainly to the complexity of the experimental rigs. This paper present an experimental investigation of the control of multimodal tonal noise propagated in circular duct in presence of a mean flow ( $M \leq 0.3$ ). A laboratory wind tunnel has been implemented for this purpose. Two limiting factors for the sound reduction are underlined: (i) the degradation of the secondary transfer matrix conditioning as the number of propagating modes increases in the duct and (ii) the degradation of the hypothesis of the time-invariance of the system to control as the flow velocity is increased. The effect of those limiting factors on the control efficiency are evaluated.

### *Contributed Paper*

3:20

**4aSAb9. New prospects in implementing distributed control strategies for mechanical structures optimization.** Manuel Collet (FEMTO-ST UMR CNRS, 23 chemin de l'Épitaphe, 25000 Besançon, France, manuel.collet@univ-fcomte.fr)

The research activities in the fields of smart materials and structures today are very important and represent a large potential for the technological innovations. New methods are now available which allow active transducers and their driving electronics to be directly integrated in otherwise passive structures. Today the main research challenge deals with the development of new multi-functional structures integrating their own electro-mechanical controlling systems. In the past few years, a technological revolution has

occurred in the fields of integrated MEMS that offers new opportunities for smart structures design and optimization. By using such as integrated active or hybrid distributed set of electromechanical transducers, we could also control the material's intrinsic mechanical behavior for building new desired functionalities. We can also speak of "integrated distributed smart structures". Through two examples, this paper aims at showing what can be the main advantages in developing such of integrated distributed control strategies in comparison with passive and "classical" active systems. The adopted point of view takes also into consideration energy balance assessments, absolute efficiency and of course robustness. The first introduced example deals with the vibroacoustic impedance control of tube for wave power flow cancellation. The second one treats of the optimization of hybrid shunted piezoelectric distributed patches for mechanical power flow control.

3:40-5:20 Posters

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

### *Contributed Papers*

5:20

**4aSAb10. Online adaptive distributed control of vibro-acoustic systems.** Kenneth Frampton (ISVR, University of Southampton, Highfield, SO17 1BJ Southampton, UK, kdf@soton.ac.uk)

One of the primary difficulties in distributed, real-time vibro-acoustic control is the difficulty of high-bandwidth network communication of sensor data. The necessary communications rate has proven overwhelming to existing communications protocols. In order to overcome this limitations an adaptive control technique has been developed that requires only occasional inter-node communication of sensor data in order to maintain optimal performance. The technique is based on iterative feedback tuning (IFT)

which has the advantage of not requiring a system model. In this system, each control node collects its sensor data over a period of time. The data is then communicated to other nodes in the system. Once a node has received the required sensor data the local adaptation algorithm is initiated. Local feedback gains are adjusted based on an estimate of the cost function gradient and the new control law is implemented until the next adaptation cycle. This work presents the theory behind this adaptive control technique and simulation results of the control performance are presented. It is demonstrated that this approach to distributed control can perform as well as model based optimal distributed control, and nearly as well as centralized control.

**4aSAb11. Damping of a device with piezoelectric materials and negative capacitance electrical circuits.** Christian S. Granger (IEMN, 41 boulevard Vauban, 59046 Lille, France, christian.granger@isen.fr), Anne-Christine Hladky (IEMN, UMR CNRS 8520, avenue Poincaré, BP 60069, 59652 Villeneuve d'Ascq, France, anne-christine.hladky@isen.fr), Jean-Claude Debus (IEMN, 41 boulevard Vauban, 59046 Lille, France, jean-claude.debus@isen.fr)

In this paper, the damping of a device is obtained by a transfer of the vibratory energy into electrical energy and then into thermal energy (dissipation in an electrical resistance). The transfer is carried out by using piezoelectric materials (PZT piezoelectric plates, macro fibre composite MFC) and it is improved by charging the piezoelectric materials by an electrical circuit having a negative capacitance impedance. Two devices are considered: a clamped plate, which is an academic case, and a large aluminium plate ( $0.85\text{m} \times 0.78\text{m} \times 2\text{mm}$ ). The optimal position and the geometry of the ceramics are determined using an analytical method and a numerical method with the help of the ATILA finite element code. The equivalent electrical circuit of the device is conceived. Damping of the device charged by the electrical circuits is measured using a laser vibrometer. Damping can be performed on a given frequency range, covering several bending modes, by using several ceramics on the plate and several external electrical circuits, connected to the active material. Finally, tests in an anechoic chamber show the efficiency of the device for the reduction of the noise transmitted through the plate.

**4aSAb12. Decentralized control of sound radiation using a high-authority/low-authority control strategy with anisotropic actuators.** Noah H. Schiller (NASA Langley Research Center, 2 North Dryden Street, Hampton, VA 23681, USA, noah.h.schiller@nasa.gov), Randolph H. Cabell (NASA Langley Research Center, 2 North Dryden Street, Hampton, VA 23681, USA, randolph.h.cabell@nasa.gov), Chris R. Fuller (Virginia Tech, 131 Durham Hall, Blacksburg, VA 24061, USA, christopher.r.fuller@nasa.gov)

This paper describes a combined control strategy designed to reduce sound radiation from stiffened aircraft-style panels. In particular, the control approach uses robust active damping in addition to high-authority LQG control. Active damping is achieved using direct velocity feedback with triangularly shaped strain actuators and point velocity sensors [P. Gardonio and S.J. Elliott, JASA 117(4), 2046-2064 (2005)]. However unlike previous work, anisotropic actuators are used since they outperform traditional isotropic actuators in this application. While active damping is simple and robust, stability is guaranteed at the expense of performance. Therefore, this approach is often referred to as low-authority control. In contrast, LQG control strategies can achieve substantial reductions in sound radiation. Unfortunately, the unmodeled interaction between neighboring control units can destabilize decentralized control systems. Numerical simulations show that combining active damping and decentralized LQG control can be mutually beneficial. In particular, augmenting the in-bandwidth damping supplements the performance of the LQG control strategy and reduces the destabilizing

interaction between neighboring control units. Therefore, the performance of the combined system can be better than the sum of each individual strategy.

**4aSAb13. An impedance matching technique for active-passive vibration control.** Jonas L. Svensson (Chalmers University of Technology, Division of Applied Acoustics, SE-41296 Gothenburg, Sweden, jonas.svensson@chalmers.se), Patrik B. Andersson (Chalmers University of Technology, Division of Applied Acoustics, SE-41296 Gothenburg, Sweden, patrik.andersson@chalmers.se), Wolfgang Kropp (Chalmers University of Technology, Division of Applied Acoustics, SE-41296 Gothenburg, Sweden, wolfgang.kropp@chalmers.se)

Impedance matching techniques have shown potential for active vibration control of structures in bending. Such structures are commonly described by Euler-Bernoulli theory. Previous studies concerning impedance matching of these structures have only considered scalar quantities. However, for an Euler-Bernoulli beam four field variables are involved which implies that a scalar impedance is insufficient. The purpose of this study is therefore to expand the technique to include full  $2 \times 2$  matrices. This is achieved by first deriving the reflection matrix as a function of the characteristic impedance matrices of an Euler-Bernoulli beam and an arbitrary termination impedance. An active impedance load is then introduced in order to manipulate the reflection matrix. A theoretical example is given where the approach is utilized to match the junction between an Euler-Bernoulli beam and a sandwich composite. This proposed active-passive damping configuration employs active control to enclose all incident power in the sandwich composite. Results show that the active impedance load is responsible for the main part of the power absorption over a broad frequency range.

**4aSAb14. Hybrid Foams for passive/active control of transmission loss of panels.** Morvan Ouisse (FEMTO-ST UMR CNRS, 23 chemin de l'Épitaphe, 25000 Besançon, France, morvan.ouisse@univ-fcomte.fr), Manuel Collet (FEMTO-ST UMR CNRS, 23 chemin de l'Épitaphe, 25000 Besançon, France, manuel.collet@univ-fcomte.fr)

Absorbing materials are widely used in transport industry. Classical foams can efficiently improve transmission loss of panels, but only in the higher frequency range. For the lower part of the frequency range, structural active control can be a good strategy to reduce the vibratory level and induced noise. Trying to take advantage of those two points is the objective of the development of hybrid foams. Many practical and scientific aspects are related to their design: which passive material should be used? Which active material, and what shapes should be better? In order to understand the basics of the hybrid foams, some specific multiphysics models are developed, including vibro-acoustic modelling of foams, coupled with elasto-dynamics for structural parts and piezo-elastic models for active parts. The objective of the strategy is to try to take advantage of distributed actuators within the foam in order to modify the energy transmission path. The physical strategy is presented in the paper, together with the full description of the model, with numerical results. The main difficulty for the numerical part is the very high number of degrees of freedom which are required by the model. Some model reduction strategies are then investigated.



## Session 4aSCa

**Speech Communication: Acoustics of Speech Production: Aeroacoustics and Phonation**

David A. Berry, Cochair

*The Laryngeal Dynamics Laboratory, Division of Head & Neck Surgery, David Geffen School of Medicine at UCLA, 31-24 Rehab Center, 1000 Veteran Ave., Los Angeles, 90095-1794, USA*

Xavier Pelorson, Cochair

*Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, Grenoble Cedex, 38031, France*

**Invited Papers**

8:40

**4aSCa1. Implications of the fluctuating drag force voice source.** Richard S. McGowan (CReSS LLC, 1 Seaborn Place, Lexington, MA 02420, USA, rsmcgowan@earthlink.net), Michael S. Howe (College of Engineering, Boston Univ., 110 Cummington Street, Boston, MA 02215, USA, mshowe@bu.edu)

A recently published paper on the aeroacoustics of the voice source calculates the acoustic source of voice caused by fluctuation drag forces [Howe, M. S. and McGowan, R. S., *J. Fluid Mech.*, 592, 367-92]. There are two extensions to this calculation that will be presented: 1) inclusion of the ventricular folds downstream of the vocal folds, and 2) the fluid-structure interaction at the vocal folds. For the ventricular folds, the effect of tissue shape on the drag forces will be investigated in terms of the shape's effect on the relation between the Lamb vector and the Kirchhoff vector. Regarding the fluid-structure interaction at the vocal folds, the two-mass model will be examined in light of the recently published calculation of drag forces. [Work supported by grant NIDCD-004688 to Dr. G. S. Berke of UCLA.]

9:00

**4aSCa2. Investigation of the mechanisms of voicing offset.** Anna Barney (ISVR, Univ. of Southampton, SO17 1BJ Southampton, UK, ab3@soton.ac.uk), Luis Jesus (Escola Superior de Saúde da Universidade de Aveiro, Universidade de Aveiro, Campus Universitário de Santiago, 3810-193 Aveiro, Portugal, lmtj@ua.pt), Ricardo Santos (Escola Superior de Saúde da Universidade de Aveiro, Universidade de Aveiro, Campus Universitário de Santiago, 3810-193 Aveiro, Portugal, ricardo.santos@ua.pt)

During voiced speech the source of sound production arises from the vibrations of the vocal folds within the larynx. In order to terminate these vibrations it is necessary either to adduct the vocal folds to produce a forced closure or to reduce the pressure drop across them to below a threshold level. A reduction in pressure drop might be achieved by reducing the sub-glottal pressure, by increasing the supra-glottal pressure or by abduction of the folds. Two human subjects were asked to achieve phonation offset by each of these strategies in turn. Acoustic and EGG signals measured on the subjects were compared with the output of a simple theoretical vocal fold model. A further comparison was made with corresponding signals recorded during the production of selected short phrases by a European Portuguese speaker; a language where devoicing of both vowels and voiced consonants is particularly prevalent.

9:20

**4aSCa3. Numerical study of volume sources associated with displacement flow during phonation.** Jong Beom Park (McGill University, 817 Sherbrooke St. West, Montreal, QC H3A 2K6, Canada, jong.b.park@mail.mcgill.ca), Luc Mongeau (McGill University, 817 Sherbrooke St. West, Montreal, QC H3A 2K6, Canada, luc.mongeau@mcgill.ca)

The glottal displacement flow is a volume flow induced by the motion of the vocal folds in the absence of transglottal pressure. This source may eventually contribute to the radiated sound, in particular during glottal opening. The volume source strength was numerically computed for a time-varying rigid M5 geometry, and decomposed into monopole and dipole components. The air mass entrained by the dipole moment was found to have a length equivalent to the glottal dimension based on the net force exerted on the fluid by the vocal folds wall. The dipole moment, associated with asymmetric geometries and/or motion, is significant even for the case of motion of an incompressible deformable solid with constant volume, but having a time-varying shape.

9:40

**4aSCa4. Aeroacoustic production of speech sounds.** Michael Krane (ARL Penn State, PO Box 30, State College, PA 16804-0030, USA, mkh5@only.arl.psu.edu)

Speech sound production is described in terms of its essential physics by focusing on the aeroacoustics of jets in the vocal tract. Aeroacoustic theory is used to show that the primary sources of sound may be expressed in terms of unsteady aerodynamic forces on the vocal tract walls, especially where a jet is formed, and where the jet performs a strong interaction with the walls. The theory further clarifies which details of jet structure and vocal tract geometry contribute to sound production. This information is used to guide useful

approximations concerning the aerodynamics. With the assistance of measurement, these approximations are used to develop a model of the sound source in the time and frequency domains. The contribution of jet structure and vocal tract geometry to sound source characteristics is then discussed. Model predictions of the resulting acoustic field are compared to acoustic measurements.

## Contributed Papers

10:00

**4aSCa5. Increasing the complexity of glottal flow models: in-vitro validation for steady flow conditions.** Julien Cisonni (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, julien.cisonni@gipsa-lab.inpg.fr), Annemie Van Hirtum (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, annemie.vanhirtum@gipsa-lab.inpg.fr), Xiao Yu Luo (Dept. of Mathematics, Univ. of Glasgow, University Gardens, G12 8QW Glasgow, UK, x.luo@maths.gla.ac.uk), Xavier Pelorson (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, pelorson@icp.inpg.fr)

Quasi one-dimensional glottal flow descriptions predict vocal folds oscillations characteristics which are qualitatively relevant to in-vitro and in-vivo experimental data. The current paper considers the resolution of the 2D Navier-Stokes equations in order to obtain a refined description of the flow phenomena adapted to more realistic glottal geometry. The pressure and flow rate predictions obtained from quasi one-dimensional flow models and the resolution of the 2D Navier-Stokes equations are examined for steady flows within a rigid glottis. The models predictions are validated against in-vitro measurements performed on rigid constriction replicas comparable to the geometrical conditions of the glottis and mounted in a suitable set-up. The confrontation between the experimental and computed data tends to show that the accuracy of the estimated pressures increases with the complexity of the flow model whereas the inverse tendency can be observed for the estimated flow rates. A focus is made on the flow separation point which is predicted by the resolution of the Navier-Stokes equations and appears to be a crucial parameter of the quasi one-dimensional flow models. The use of a variable separation criterion obtained from the 2D flow modelling in the quasi one-dimensional models makes the different models predictions more similar.

10:20

**4aSCa6. Calculation model of the influence of the vocal fold shape on the vocal fold oscillation form.** Andreas Gömmel (RWTH Aachen, Civil Engineering Dept., Structural Statics and Dynamics, Mies-van-der-Rohe-Str. 1, 52064 Aachen, Germany, goemmel@lbb.rwth-aachen.de), Christoph Butenweg (RWTH Aachen, Civil Engineering Dept., Structural Statics and Dynamics, Mies-van-der-Rohe-Str. 1, 52064 Aachen, Germany, butenweg@lbb.rwth-aachen.de), Malte Kob (RWTH Aachen, Dept. of Phoniatrics, Pedaudiology, and Communication Disorders, Pauwelsstr. 30, 52074 Aachen, Germany, mkob@ukaachen.de)

Vocal fold (VF) oscillation is driven by fluid-structure interaction effects. A possible way of modeling these effects is the finite-element (FE) method. The presented FE model consists of two coupled domains: A fluid domain representing the air, and a structural domain representing the VFs. In principle, each of the domains is a stand-alone simulation model. In the current implementation a thin three-dimensional frontal slice of the vocal folds and the sub- and supraglottal areas is modeled. Flow calculation is done using the standard Navier-Stokes equations. The air is modeled as a transient, viscous, and laminar flow. Constant physiologic values of pressure are used as driving force. For structural analysis, linear volume elements are used. There are two different models which differ in the VF shape. The first one is an assumed shape of a normal voice while the second shape was measured at an excised larynx and resembles more a falsetto voice. The results support two observations and assumptions: During normal phonation the VF touch each other in a constantly changing converging and diverging shape while during the more falsetto-like phonation, no converging/diverging shape is visible and no closure occurs.

10:40-11:00 Break

11:00

**4aSCa7. Intraglottal pressure distributions for oblique glottal angles.** Ronald C. Scherer (Bowling Green State University, Department of Physics and Astronomy, Bowling Green, OH 43403, USA, ronalds@bgsu.edu)

Asymmetric vocal fold oscillation occurs for both normal and pathological phonation. The pressures on the right and left glottal walls receive different pressures when the glottis is oblique, as we have shown previously. Different driving forces may affect the motion, although to what extent is not yet known. We have continued empirical modeling of various oblique glottal shapes using a Plexiglas model (M5), and report the findings of those studies. In general, an oblique glottis tends to receive higher pressures on the more convergent side, and the cross-channel pressure differences may shift polarity between upstream and downstream sections. The effects on intraglottal pressures for a wide range of obliquity, minimal glottal diameter, transglottal pressure, and included glottal angle will be shown and discussed relative to potential oscillatory effects. [supported by NIH R01DC03577]

11:20

**4aSCa8. Separated flow behavior in an in-vitro rigid model of the laryngeal channel.** Denisse Sciamarella (CNRS, Bâtiment 508, Université Paris-Sud, 91403 Orsay, France, denisse@limsi.fr), Elisa Chisari (LFD, Facultad de Ingenieria, Universidad de Buenos Aires, Av. Paseo Colon 850, C1063ACV Buenos Aires, Argentina, elisachisari@gmail.com), Guillermo Artana (LFD, Facultad de Ingenieria, Universidad de Buenos Aires, Av. Paseo Colon 850, C1063ACV Buenos Aires, Argentina, gartana@fi.uba.ar), Lucie Bailly (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, Lucie.Bailly@gipsa-lab.inpg.fr), Xavier Pelorson (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, pelorson@icp.inpg.fr)

Flow through an in-vitro rigid model of the scaled-up laryngeal channel is measured using pressure sensors and visualized using the Schlieren technique for different geometrical configurations. Three downstream flow-conditions are considered: steady, quasi-impulsive and periodical using an electromechanical device controlling the inflow and producing the cyclic jet emerging from the glottis. The separated flow behavior in the presence of a ventricular constriction ("false vocal folds") is also examined. Direct theoretical flow predictions and numerical simulations are proposed to quantify the aerodynamic impact involved by the ventricular bands on the pressure distribution. Two parameters are investigated: the aperture of the ventricular bands and the distance between the vocal folds and the ventricular bands. The influence of both parameters are measured and compared to the theoretical outcome.

11:40

**4aSCa9. Aeroacoustic measurements in a vocal tract model.** Daniel J. Leonard (ARL Penn State, PO Box 30, State College, PA 16804-0030, USA, djl239@psu.edu), Michael Krane (ARL Penn State, PO Box 30, State College, PA 16804-0030, USA, mkh5@only.arl.psu.edu)

An experiment to clarify the relation between turbulent jet structure, vocal tract wall shape and the resulting sound is described. A life-scale model of the vocal tract (18cm length, 2.6cmx2.6cm square cross-section), fabricated from clear cast acrylic is used. A jet, formed at a constriction, passes over or against a simple obstacle, generating sound. Correlated aerodynamic and acoustic measurements are used to determine the transfer function between the obstacle and the measurement location outside the model and the aeroacoustic source spectrum. For comparison the source spectrum is also estimated using the model described in Krane (JASA, 2005), using aerodynamic measurements as empirical input.

12:00

**4aSCa10. Influence of vocal fold stiffness on phonation characteristics at onset in a body-cover vocal fold model.** Zhaoyan Zhang (School of Medicine, University of California, 31-24 Rehab Center, 1000 Veteran Ave., Los Angeles, CA 90095-1794, USA, zyzzhang@ucla.edu), Juergen Neubauer (School of Medicine, University of California, 31-24 Rehab Center, 1000 Veteran Ave., Los Angeles, CA 90095-1794, USA, jneubauer@mednet.ucla.edu)

The influence of the body and cover stiffnesses on phonation onset and the resulting vibration pattern was investigated in a body-cover continuum model of the vocal folds. An eigenvalue analysis was performed to obtain phonation onset characteristics. The analysis showed that, with increasing body-cover stiffness ratio, both the phonation threshold pressure and frequency (normalized by the Young's modulus and wave speed of the cover layer, respectively) first increased rapidly and then gradually approached a plateau. For a given glottal resting opening, a soft vocal fold body led to a larger prephonatory glottal opening, which had a negative effect on phonation onset pressure, and for certain vocal fold geometries, led to a local minimum in the phonation threshold pressure as a function of the body stiffness. Although the phonation threshold pressure was low for a vocal fold configuration with both a soft cover and a soft body, the vocal fold vibration at onset exhibited a significant whole-body vertical motion and a low sound production efficiency, and therefore it may not be desirable for voice production. For a large body-cover stiffness ratio, this vertical motion was suppressed and vibration was restricted to the cover layer and the medial surface, resulting in a more effective flow modulation and a better sound production efficiency.

12:20

**4aSCa11. Modeling of voice registers and bifurcation theory.** Marco Zemke (Institute for Theoretical Biology, Invalidenstr. 43, 10115 Berlin, Germany, marco.zemke@biologie.hu-berlin.de), Isao Tokuda (Japan Advanced Institute of Science and Technology, 1-1 Asahidai, Nomi-shi, 923-1292 Ishikawa, Japan, isao@jaist.ac.jp), Hanspeter Herzel (Institute for Theoretical Biology, Invalidenstr. 43, 10115 Berlin, Germany, h.herzel@biologie.hu-berlin.de)

The exact definition of registers in the human voice is still under debate. Especially the quantitative analysis of transitions between these registers have not been investigated in much detail yet. Experiments show different kinds of voice instabilities that appear close to the transition from chest to falsetto register. These instabilities include abrupt jumps between the two registers exhibiting hysteresis, aphonic episodes, subharmonics and chaos. To model these phenomena we start with a three mass cover model of the human larynx which can vibrate in chest and falsetto like patterns. We extend our model to a four mass body-cover polygon model with a supraglottal resonator. We more realistically make use of bifurcation theory to understand how small smooth changes of parameter values can cause sudden qualitative changes in the dynamical behavior of the larynx. Experiments with excised human larynges and untrained singers show comparable coexistence of both registers in an extended parameter region. Our study suggests that voice range profiles can be generalized as two dimensional bifurcation diagrams.

12:40-2:00 Lunch Break

### Contributed Papers

2:00

**4aSCa12. Theoretical and experimental results of phonation threshold pressure vs. oscillation frequency of the vocal folds.** Jorge C. Lucero (Univ. Brasilia, Dept. Mathematics, 70910-900 Brasilia - DF, Brazil, lucero@unb.br), Annemie Van Hirtum (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, annemie.vanhirtum@gipsa-lab.inpg.fr), Nicolas Ruty (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, nicolas.ruty@gipsa-lab.inpg.fr), Julien Cisonni (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, julien.cisonni@gipsa-lab.inpg.fr), Xavier Pelorson (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, pelorson@icp.inpg.fr)

The dynamical principles of the vocal fold oscillation at phonation were set forth by Titze (I. R. Titze, J. Acoust. Soc. Am. **83**, 1536-1552, 1988), by representing motion of the tissues as a surface wave propagating in the direction of the airflow. An important result of his work was an equation for the phonation threshold value of lung pressure, defined as the minimum value required to initiate the vocal fold oscillation. Titze's model assumed a small time delay for the mucosal wave to travel along the vocal folds, with the consequence that the phonation threshold pressure results independent of the oscillation frequency. Here, we consider an extension of his model for an arbitrary time delay. Our results show that the threshold pressure increases with oscillation frequency following a  $x/\sin(x)$  law. We investigate the validity of the theoretical equation by comparing it with pressure measures from a mechanical replica of the vocal folds, under various configurations. In general, the equation shows good agreement with the experimental data, and may find applications for building empirical relations of glottal aerodynamics, and for clinical studies of phonation. [Work supported by CAPES-Brazil]

2:20

**4aSCa13. Physical modelling of Vowel-Stop-Vowel sequences.** Claire Brutel-Vuilmet (GIPSA-Lab, INPG, 46 Av. Felix-Viallet, 38031 Grenoble, France, claire.brutel-vuilmet@gipsa-lab.inpg.fr), Xavier Pelorson

(Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, pelorson@icp.inpg.fr), Annemie Van Hirtum (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, annemie.vanhirtum@gipsa-lab.inpg.fr), Susanne Fuchs (ZAS, Schuetzenstr. 18, 10117 Berlin, Germany, fuchs@zas.gwz-berlin.de)

The physical phenomena involved in the production of vowel sounds are now well described and several effective physical models have been developed. On the contrary, only a few studies concern the physical aspects of the production of consonant sounds (fricatives, stops...) or the production of vowel-consonant-vowel sequences. Indeed, more complex acoustical and aerodynamical mechanisms should be considered (generation of aeroacoustical sources, dynamical description...). In this contribution, we focus on the production of vowel-stop-vowel sequences. A stop sound is produced after a complete occlusion of the vocal tract. In an intervocalic context, the closure entails an increase of the intraoral pressure, which results in the cessation of the vocal folds vibration. The opening is quickly followed by the generation of an aeroacoustical source, and entails a decrease of the intraoral pressure, which results in the resumption of the vocal folds motion. A physical model able to describe these aerodynamical observations and to predict the evolution of the intraoral pressure during the closure gesture is first detailed. Different interesting results are discussed (influence of the closure time, influence of the closure position, coordination of the different sources...). And a confrontation between theoretical results and experimental data is presented.

2:40

**4aSCa14. Analysis of frication noise modulation from a physical model.** Anna Barney (ISVR, Univ. of Southampton, SO17 1BJ Southampton, UK, ab3@soton.ac.uk), Philip J. Jackson (University of Surrey, Centre for Vision, Speech and Signal Processing, GU2 7XH Guildford, UK, p.jackson@surrey.ac.uk)

A physical model, built to investigate the aeroacoustic properties of voiced fricative speech, was used to study the amplitude modulation of the

turbulence noise it generated. The amplitude and fundamental frequency of glottal vibration, relative positions of the constriction and obstacle, and the flow rate were varied. Measurements were made from pressure taps in the duct wall and the sound pressure at the open end. The high-pass filtered sound pressure was analyzed in terms of the magnitude and phase of the turbulence noise envelope. The magnitude and phase of the observed modulation was related to the upstream pressure. The effects of moving the obstacle with respect to the constriction are reported (representative of the teeth and the tongue in a sibilant fricative respectively). These results contribute to the development of a parametric model of the aeroacoustic interaction of voicing with turbulence noise generation in speech.

3:00

**4aSCa15. Towards sibilant /s/ modelling: preliminary computational results.** Xavier Grandchamp (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, xavier.grandchamp@gipsa-lab.inpg.fr), Annemie Van Hirtum (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, annemie.vanhirtum@gipsa-lab.inpg.fr), Xavier Pelorson (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, pelorson@icp.inpg.fr), Kazunori Nozaki (Cyber Media Center, Osaka University Cybermedia, 5-1 Mihogaoka Ibaraki, 567-0047 Osaka, Japan, nozaki@cmc.osaka-u.ac.jp), Shinji Shimojo (Cyber Media Center, Osaka University Cybermedia, 5-1 Mihogaoka Ibaraki, 567-0047 Osaka, Japan, shimojo@cmc.osaka-u.ac.jp)

Acoustic effects of the turbulence in human sounds production is generally accepted in the case of fricatives. Nevertheless, this phenomenon is not taken into account in physical modelling of the speech production. Steady flow in a uniform two dimensional channel with a one side triangular

obstacle has been simulated using Large Eddy Simulation for different Reynolds numbers. The used geometrical and flow characteristics are severe simplifications of the human articulators during the production of the sibilant /s/. The impact of different geometrical configurations of the downstream obstacle representing the teeth on the computational results is searched in order to detect the aeroacoustic sources produced by turbulent flow, crucial in /s/ production.

3:20

**4aSCa16. The Influence of Constriction Geometry on Sound Generation in Fricative Consonants.** Gordon Ramsay (Haskins Laboratories, 300 George Street, New Haven, CT 06511, USA, ramsay@haskins.yale.edu)

Sound generation in fricative consonants is traditionally supposed to depend only on the Reynolds Number, usually defined in terms of the constriction area and the volume velocity at the constriction. The potential influence of the detailed three-dimensional geometry of the constriction is often ignored, even though previous empirical studies have shown this to have an important effect on the spectral shape of the source and the overall sound strength. At present, the physical processes governing turbulent jet formation and aeroacoustic source generation in fricative consonants are not fully understood. In this paper, we use large-eddy simulations of three-dimensional viscous incompressible flow to visualize the development of the turbulent flow field and aeroacoustic source distribution in an elliptical duct representing the vocal tract, for elliptical, laminar, and grooved constriction shapes that share the same cross-sectional area function. By contrasting results for these geometries, we test the hypothesis that turbulent jet formation is determined largely by the shape of the boundary layer where flow separates at the exit of the constriction, and that the perimeter of the constriction, rather than the cross-sectional area, may therefore be a more appropriate parameter for characterizing properties of the aeroacoustic source.

THURSDAY MORNING, 3 JULY 2008

ROOM 250B, 8:40 A.M. TO 12:00 NOON

## Session 4aSCb

### Speech Communication: Neurobiology of Speech Perception

Paul Iverson, Cochair

*University College London, Department of Phonetics and Linguistics, 4, Stephenson Way, London, NW1 2HE, UK*

Christophe Pallier, Cochair

*Cognitive Neuroimaging Unit INSERM 562, CEA/Neurospin, bat 145, point courier 156, Gif/Yvette, 91191, France*

### Invited Papers

8:40

**4aSCb1. Involvement of Auditory Cortex in Speech Production.** Frank H. Guenther (Boston University, 677 Beacon Street, Boston, MA 02215, USA, guenther@cns.bu.edu)

In addition to their role in speech perception, the auditory cortical areas of the superior temporal lobe are important for the formation and maintenance of motor commands for speech production. Using a combination of neural network modeling, neuroimaging, and auditory perturbation experiments, we have characterized the network of brain regions involved in auditory feedback control of segmental aspects of speech. This network involves auditory error cells in bilateral posterior superior temporal cortex which become active when the current auditory feedback mismatches the auditory target for the current speech sound. Projections from these auditory cortical areas to the right hemisphere ventral premotor areas, then on to primary motor cortex, transform perceived auditory errors into corrective movement commands for the speech articulators. The DIVA model of speech production produces a close quantitative fit to acoustic data collected during unexpected auditory perturbation of speech and during sensorimotor adaptation to sustained auditory perturbations. Neuroimaging results motivate a modification of the model to include right ventral premotor cortical areas in the auditory feedback circuit for speech production. [Work supported by NIDCD, NSF.]



**4aSCb2. Motor regions contribute to speech perception: awareness, adaptation and categorisation.** Matthew H. Davis (MRC CBU, 15 Chaucer Rd., CB2 7EF Cambridge, UK, matt.davis@mrc-cbu.cam.ac.uk), Ingrid S. Johnsrude (Queen's University, Dept Psychology, 62 Arch Street, Kingston, ON K7L 3N6, Canada, ingrid.johnsrude@queensu.ca), Alexis G. Hervais-Adelman (Centre for the Neural Basis of Hearing, Department of Physiology, Development and Neuroscience, University of Cambridge, Downing Site, CB23EG Cambridge, UK, alexis.hervais-adelman@mrc-cbu.cam.ac.uk), Jack C. Rogers (MRC Cognition and Brain Sciences Unit, 15 Chaucer Road, CB2 7EF Cambridge, UK, jack.rogers@mrc-cbu.cam.ac.uk)

Functional imaging and TMS studies show that motor and premotor cortex responds to heard speech though the functional significance of this response is unclear. Three recent fMRI studies, showing modulation of motor responses to heard speech in the absence of overt spoken or manual responses, may shed light on how regions typically associated with speech production contribute to perception. (1) Motor activity remains robust during light anaesthetic sedation, but is obliterated for deeply sedated participants who are no longer aware of speech. (2) Motor cortex responds more to distorted yet intelligible noise-vocoded words than to clear speech or unintelligible noise. This neural correlate of listening effort is also observed for "clear-then-vocoded" presentations that enhance perceptual adaptation compared to a matched "vocoded-then-clear" condition that doesn't enhance adaptation. (3) During paired priming of audio-morphed syllables we see a greater response to acoustic changes that cross phonological category boundaries compared to acoustically-equivalent within-category changes. Additional responses to between-category pairs extend to motor regions, suggesting that neural correlates of categorical perception include regions involved in speech production. These findings illustrate how perceptual awareness, adaptation and categorisation of speech all engage motor regions. Implications for neurobiological accounts of speech perception will be discussed.

**4aSCb3. From Speech to Language: Mapping the Auditory Comprehension Network.** Jonas Obleser (Max Planck Institute of Human Cognitive and Brain Sciences, Stephanstrasse 1A, 04103 Leipzig, Germany, jonas@obleser.de)

Speech comprehension is a complex perceptual and cognitive task that is fulfilled in a surprisingly robust manner. I will present a series of studies that aim at disentangling the interaction of sensory/acoustic and cognitive/linguistic factors driving the speech comprehension system: (i) How does the system in its entirety deal with noise at the auditory entry level? (ii) Which are key contextual influences that aid speech comprehension when the signal quality drops, and (iii) what is the functional circuitry within and across auditory cortex that copes with comprehension difficulties? Main results include: 1. Comprehension of intermediate signal quality based on semantic predictability engages a left-hemispheric, widely distributed array of brain structures. Also, functional connectivity amongst these areas appears enhanced. 2. Signal intelligibility gates and enables the expected semantic effects (cloze probability; BA44 BOLD response and EEG N400), whereas semantic effects shape the intelligibility modulation in anterolateral temporal cortex. 3. The angular gyrus (BA 39) enhances difficult yet successful speech comprehension whereas left posterior STG reflects higher computational effort (either poor signal quality or low cloze probability) in speech comprehension. Behavioural, EEG and fMRI data will be presented, and themes of (bi-)laterality and bottom-up/top-down directionality will be re-visited along the way.

**4aSCb4. An fMRI study of subliminal priming of spoken words.** Christophe Pallier (Cognitive Neuroimaging Unit INSERM 562, CEA/Neurospin, bat 145, point courier 156, 91191 Gif/Yvette, France, christophe@pallier.org), Sid Kouider (Laboratoire de Sciences Cognitives et Psycholinguistique, Ecole Normale Supérieure, 29 rue d'Ulm, 75005 Paris, France, sid.kouider@ens.fr), Vincent De Gardelle (Laboratoire de Sciences Cognitives et Psycholinguistique, Ecole Normale Supérieure, 29 rue d'Ulm, 75005 Paris, France, gardelle@biologie.ens.fr)

Repetition priming has been widely used to study spoken and written word recognition. Its physiological counterpart is repetition suppression, a reduction in neural activity resulting in a measurable decrease of the fMRI signal. By varying the representational level at which the repetition occurs, one can determine which properties are encoded in a given brain area, and which are not. We will report on an fMRI experiment using subliminal auditory priming of spoken words. Subliminal priming has been used, for example, by Dehaene et al. to study visual word recognition. Our experiment employs a technique developed by Kouider & Dupoux that allows subliminal presentation of auditory stimuli using temporal compression and forward masking. The participants perform a lexical decision task on the target item, which is preceded by a subliminal prime that can be phonetically identical or different from the target, and spoken or not by the same speaker. A fast-event related paradigm is used where each prime-target pair is presented during silent gaps between the acquisitions. The planned analyses will seek to identify the brain regions showing subliminal repetition suppression to word repetition regardless of speaker change, as well as other areas sensitive to speaker change regardless of linguistic content.

### *Contributed Paper*

**4aSCb5. The neural bases of normalising for accented speech: A repetition suppression functional magnetic resonance imaging study.** Patti Adank (F.C. Donders Centre for Cognitive Neuroimaging, Kapittelweg 29, 6525EN Nijmegen, Netherlands, patti.adank@fcdonders.ru.nl), Peter Hagoort (F.C. Donders Centre for Cognitive Neuroimaging, Kapittelweg 29, 6525EN Nijmegen, Netherlands, peter.hagoort@fcdonders.ru.nl)

A repetition suppression fMRI paradigm was employed to explore the neuroanatomical substrates of normalisation for accented speech in spoken sentence processing. Sentences were produced in two accents: in Standard Dutch and an artificial accent of Dutch. In the experiment, participants

listened to two sentences presented in quick succession. The second sentence was either spoken by the same speaker in the same accent, by the same speaker in a different accent, by a different speaker in the same accent, or by a different speaker in a different accent. This design allowed us to study neural responses to a change in speaker only, a change in accent only and a change in accent and speaker. Results showed small effects for a change of speaker only in right Superior Temporal Gyrus (STG). A change of accent only showed extensive activations in left and right STG and Superior Temporal Sulcus (STS). Finally, a change of speaker and accent showed extensive activations in left and right STG and STS, and increased activity in left Inferior Frontal Gyrus (IFG). The results indicate that normalisation processes for accented speech recruit a wide neural network. The role of left IFG in normalisation processes will be discussed.

*Invited Papers*

10:40

**4aSCb6. Streams of processing and hemispheric asymmetries in speech perception.** Sophie K. Scott (University College London, Institute of Cognitive Neuroscience, 17 Queen Square, WC1N 3AR London, UK, sophie.scott@ucl.ac.uk)

Studies in non-human primates have indicated that, as in the visual system, there are (at least) two streams of processing in the auditory system. These pathways are associated with different types of auditory processes - an anterior 'what' pathway and posterior 'how/where' pathway(s). In this talk I will use these neurophysiological theories as a framework for interpreting findings from a range of PET and fMRI studies of human speech perception and production, and present evidence that the anterior 'what' pathway in humans shows hierarchical processing of the speech signal, reflecting a move from acoustic/phonetic processing to a more abstract representation in the anterior superior temporal sulcus. In contrast, posterior auditory areas in humans are associated with sensory/motor interactions in speech, and with aspects of working memory processing. I will address how these systems are differentially recruited when speech perception is made difficult, due to different types of masking noise. Finally, I will outline differences in the processing of speech in left and right auditory areas.

11:00

**4aSCb7. The temporal analysis of spoken language.** David Poeppel (University of Maryland, 1401 Marie Mount Hall, College Park, MD 20742, USA, dpoeppel@umd.edu)

The (concurrent) construction of syllabic and phonemic representations forms the basis for creating interpretable representations of speech; therefore we look here to temporal attributes commensurate with their acoustic implementation. Based on a distributed model of the functional anatomy of speech perception (Hickok & Poeppel 2007) and on the assumption that the perception of speech requires multi-time resolution analysis (Poeppel 2003), electrophysiological data are shown that illustrate how auditory cortex makes use of one specific temporal mechanism, the processing of phase (Luo & Poeppel 2007). We hypothesized that the phase pattern of cortical rhythms associated with modulation rates mediating intelligible speech provide an encoding mechanism. We observed that the phase of the theta band response generated in auditory cortex tracks sentence-level acoustics with the sensitivity and specificity necessary for neuronal encoding. The data are consistent with the view that a ~200 ms temporal window (period of theta oscillation) segments the incoming signal, resetting and sliding to track speech dynamics. This hypothesized mechanism for cortical speech analysis is based on the stimulus-induced modulation of inherent cortical rhythms and provides supporting evidence implicating the syllable as a computational primitive for the representation of spoken language.

11:20

**4aSCb8. ERPs and speech sound perception - possibilities and restrictions.** Maija S. Peltola (University of Turku, Department of Phonetics, FIN-20014 Turku, Finland, maija.peltola@utu.fi)

Speech sound perception is a complex combination of attention independent and attention dependent processes which both contribute to the final goal of understanding the spoken message. The preattentive level has become more accessible to research and event related potentials (ERPs) can easily be used to study the automatic processing of the speech signal. In particular, the mismatch negativity (MMN) response offers a tool for investigating the manner in which speech sounds are encoded as neural representations. Cross-linguistic studies revealing different kinds of representations in native speakers of different languages form the core for further studies, which have shown the plasticity of the brain in forming new representations for non-native sounds in various types of learning environments. However, despite all these promising advances there are still some restrictions connected both with the methodology available and the conclusions that can be reached on the basis of the occasionally contradicting results. Also, since results obtained by using attention independent and dependent methods are not always compatible, some potentially significant results may never reach the attention that they deserve, even if this incompatibility may be one of the keys into the understanding of the complicated mechanisms underlying speech sound perception.

*Contributed Paper*

11:40

**4aSCb9. Cognitive control skills and speech perception after short-term second language experience during infancy.** Barbara Conboy (University of Washington, Dept. of Speech & Hearing Sciences, and Institute for Learning & Brain Sciences, Box 357988, Seattle, WA 98195, USA, bconboy@u.washington.edu), Jessica Somerville (University of Washington, Dept. of Speech & Hearing Sciences, and Institute for Learning & Brain Sciences, Box 357988, Seattle, WA 98195, USA, sommej@u.washington.edu), Patricia K. Kuhl (University of Washington, Dept. of Speech & Hearing Sciences, and Institute for Learning & Brain Sciences, Box 357988, Seattle, WA 98195, USA, pkkuhl@u.washington.edu)

Previous research has linked increasing cognitive abilities to reductions in sensitivity to nonnative phonemes toward the end of the first year, but found no association between cognitive skills and native speech perception (Conboy et al., 2006; Lalonde & Werker, 1995). The present study examined cognitive abilities and brain activity to second-language (L2) phoneme contrasts in infants who had short-term experience with the L2: we predicted better cognitive skills in infants with better discrimination of the L2 contrast. Seventeen infants from monolingual English homes completed event-related potential (ERP) speech perception testing and nonlinguistic tasks requiring attentional flexibility, memory, and inhibitory control at 11 months, after twelve Spanish play sessions from 9.5 - 10.5 months. An ERP oddball paradigm assessed discrimination of English and Spanish contrasts

(English: voiced /da/ vs. voiceless-aspirated [tha]; Spanish: prevoiced /da/ vs. voiceless-unaspirated /ta/). Infants showed broad mismatch negativity (MMN) discriminatory responses to both contrasts. Larger Spanish MMN amplitudes were linked to better performance on cognitive tasks (detour-

reaching object-retrieval and the A not B task) (Fisher's exact test,  $p=.01$ ), suggesting a role for specific cognitive abilities in the early stages of phonetic learning. There was no association between English MMN amplitudes and cognitive skills.

THURSDAY MORNING, 3 JULY 2008

ROOM 343, 8:00 TO 10:20 A.M.

### Session 4aSPa

## Signal Processing in Acoustics, Acoustical Oceanography, and ECUA: Model-Based Signal Processing II

Sean Lehman, Cochair

*Lawrence Livermore Natl. Lab., Livermore, CA 94551, USA*

Christian Pichot, Cochair

*Antennas & Telecommunications Laboratory, University of Nice-Sophia Antipolis, France*

### Contributed Paper

8:00

**4aSPa1. Water column tomographic inversion with a network of drifting buoys.** Sergio Jesus (ISR, Universidade do Algarve, PT-8005-139 Faro, Portugal, [sjesus@ualg.pt](mailto:sjesus@ualg.pt)), Cristiano Soares (ISR, Universidade do Algarve, PT-8005-139 Faro, Portugal, [csoares@ualg.pt](mailto:csoares@ualg.pt)), Nelson Martins (ISR, Universidade do Algarve, PT-8005-139 Faro, Portugal, [nmartins@ualg.pt](mailto:nmartins@ualg.pt))

The estimation of ocean environmental properties by means of the inversion of acoustic signals has in several occasions been performed using a single vertical array of acoustic receivers, with a towed acoustic source as an attempt to ensure a rapid spatial coverage of the area of interest, as only a single ocean transect is "seen" at each time. Ideally, one would like to obtain

an instantaneous picture of the complete area (volume) under observation. However, the resulting acoustic observations, hence environmental estimates, are not simultaneous in time. Using multiple acoustic receiving arrays appears to be a natural step towards both increasing the spatial coverage, and obtaining simultaneous environmental estimates of different ocean transects. It also gives a higher chance to capture spatial transient features, as for example solitons. Using multiple receiver arrays represents the addition of a new spatial dimension at the receiving end and opens up the number of possibilities to a Nx2D or full 3D view of the ocean. Taking support on the data set of the RADAR'07 experiment (July 9 - 16, 2007) where data was simultaneously collected on three vertical arrays, this paper explores space coherent processing of the several receiving arrays and Nx2D or 3D environmental constrained water column matched-field inversion.

### Invited Paper

8:20

**4aSPa2. Model based echo processing architectures for sonar target classification.** Manell E. Zakharia (French Naval Academy, BP 600, 29240 Brest-Armées, France, [manell.zakharia@ecole-navale.fr](mailto:manell.zakharia@ecole-navale.fr))

The discrimination between man-made and natural targets is faced to the problem of setting up appropriate processing architectures that extract relevant and robust parameters that could be used for classification. To be robust, signal models have to be associated to physical models and echo parameters have to be associated to physical ones. Several models already published are investigated: bright spots, generalized bright spots, resonances. Associated processing architectures are presented: matched filter, bank of filters, AR modeling. Their performance are compared on experimental data set obtained in tank. The discriminating performance are compared in the case of shells (man-made) and solid targets (natural) of the same shape insonified in a random incidence (monostatic). Following a detailed description of echo formation mechanisms in the time-frequency plane, an explicit time-frequency architecture is presented: the time-frequency filtering. Finally a new (all chirp) model based on velocity dispersion of surface waves, is proposed that could reduce the number of discriminating parameters and be robust to minor changes of shell characteristics.

### Contributed Papers

8:40

**4aSPa3. Automatic Acoustics Measurement of Audible Inspirations in Pathological Voices.** Eduardo Castillo-Guerra (University of New Brunswick, P.O. Box 4400, 15 Dineen Dr., D36 Head Hall, Fredericton, NB E3B 5A3, Canada, [ecastill@unb.ca](mailto:ecastill@unb.ca)), Williams Lee (University of New Brunswick, P.O. Box 4400, 15 Dineen Dr., D36 Head Hall, Fredericton, NB E3B 5A3, Canada, [wlee@unb.ca](mailto:wlee@unb.ca))

Audible inspiration (AI) is a type of speech perturbation commonly heard in pathologic voices. This acoustic parameter is used in conjunction with other acoustic observations to assess different types of pathologic conditions of speech associated with neurological or vocal cord disorders.

However, the perception of this speech perturbation is often very subjective and difficult to appraise in a quantitative and consistent form. This work reports an algorithm to estimate the severity of this perturbation using time-frequency characteristics. The algorithm is based on a linear combination of the frequency of occurrence, the duration and the intensity of the inspirations. An algorithm to segment the AIs in conversational speech is proposed. The AI index was first evaluated with the Massachusetts Eye and Ear Infirmary Voice Database and then with two other databases containing recording from motor speech disorders. The segmentation algorithms was 90% effective detecting the audible inspirations and the AI index showed a high correlation (80.8%) with respect to the average perceptual judgment obtained from three judges with experience evaluating disordered speech.

9:00

**4aSPa4. Comparative Study of Wideband Subspace Direction of Arrival (DOA) estimation methods.** Sheraz Khan (Laboratoire Ondes et Acoustique, ESPCI, 10 rue Vauquelin, 75231 Paris, France, sheraz.khan@polytechnique.edu)

Signal subspace Methods like ESPRIT, MUSIC and MATRIX PENCIL, provides high resolution Direction Of Arrival (DOA) estimation in comparison to traditional Delay and Sum and Capon methods, which are limited by sensor spacing. However underwater Acoustics signals are inherently wideband in nature and most of these Subspace methods works on narrowband signals. Currently modified version of these methods for wideband signals are emerging. These methods are broadly classified as Coherent and Incoherent methods depending upon how signals of different frequencies have been merged. Performances of these methods are evaluated using extensive Monte-Carlo simulations under various protocols by comparing there Mean Square Error in DOA estimation and by their respective resolution power. This comparison study is also complemented with real acoustic data from public domain.

9:20

**4aSPa5. Application of statistical methods in underwater signal classification.** Brett E. Bissinger (ARL Penn State, PO Box 30, State College, PA 16804, USA, beb194@psu.edu), Richard Lee Culver (ARL Penn State, PO Box 30, State College, PA 16804, USA, rlc5@psu.edu), Nirmal K. Bose (ARL Penn State, PO Box 30, State College, PA 16804, USA, bkn@enr.psu.edu), Colin W. Jemmott (ARL Penn State, PO Box 30, State College, PA 16804, USA, cwj112@psu.edu)

The overall goal of our work is to utilize knowledge of the ocean environment to improve sonar detection and classification performance. Source classification and localization in the underwater environment is a challenging problem in part because propagation through the space- and time-varying medium introduces multipath, variability, and decorrelation to the signal. Traditional underwater signal classification has relied on parametric methods such as the likelihood ratio tests. Recent research has explored non-parametric methods like maximum entropy and maximum likelihood with favorable results. This talk considers other, more contemporary non-parametric methods, e.g. principle component analysis, independent component analysis and support vector machines, and compares their structure and performance with previous results. Work supported by Office of Naval Research Undersea Signal Processing.

9:40

**4aSPa6. Estimation of acoustic directivity from microphone array measurements using parametric models.** Jean Bulté (ONERA, BP 72 - 29, avenue de la division Leclerc, 92322 Châtillon, France,

jean.bulte@onera.fr), Vincent Fleury (ONERA, BP 72 - 29, avenue de la division Leclerc, 92322 Châtillon, France, vincent.fleury@onera.fr), Renaud Davy (ONERA, BP 72 - 29, avenue de la division Leclerc, 92322 Châtillon, France, renaud.davy@onera.fr)

In this paper, we are interested in recovering the far-field acoustic pattern of a directive source from signals recorded in the near-field by an array of microphones with a reduced spatial extent. This question is particularly relevant in small test facilities where far-field acoustic measurements can not be carried out. A two-step approach is suggested. Firstly, the characteristics of sources are estimated from near-field measurements. Secondly, these characteristics are used to estimate the far-field radiation pattern. The main difficulty of this problem mainly resides in the first step. Due to the reduced spatial extent of the array, much information is lost about source characteristics, which mathematically leads to solve an ill-posed inverse problem. Our approach consists in using a parametric model based on physical assumptions, which has the virtue of regularizing the estimation problem. The suggested method is firstly evaluated with simulations, and then applied to experimental data recorded during aeroacoustic tests with a subsonic jet in an anechoic wind tunnel. It is shown that comparison between far-field measurements and estimated far-field pattern are in good agreement.

10:00

**4aSPa7. A volumetric interferometric synthetic aperture sonar reconstruction algorithm.** 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 synthetic aperture sonar (InSAS) bathymetric reconstruction is an inverse problem that is often simplified to a time delay estimation problem. This uses a simple system model of a continuous scattering surface with single scatterer per resolution cell. This model is violated by layover, multiple scattering, occlusions, or sea-surface multipath producing artefacts in the reconstructed image. While some artefacts, such as from occluded shadow regions, can be rejected by using a threshold on the correlation coefficient, this does not work in general. Moreover, since each pixel is reconstructed independently it is difficult to improve the reconstruction by adding prior information. In this paper we propose a reconstruction algorithm using a probabilistic volumetric model; similar to those used for photometric 3-D scene reconstruction from multiple cameras. While significantly slower than time delay estimation methods, the bathymetric reconstruction can be improved due to better scene modelling and the incorporation of priors such as surface continuity. Furthermore, an advantage of a volumetric model is that correction for the footprint shift is implicit. We demonstrate the algorithm using both simulated and real data.



**Session 4aSPb****Signal Processing in Acoustics, Acoustical Oceanography, and ECUA: Bayesian Signal Processing II**

Zoi-Heleni Michalopoulou, Cochair

*Department of Mathematical Sciences, New Jersey Institute of Technology, 323 M L King Blvd, Newark, NJ 07102, USA*

Simon J. Godsill, Cochair

*Trumpington Street, Cambridge, CB21PZ, UK***Invited Papers****10:40**

**4aSPb1. An Overview of Bayesian Computational methods for audio signal processing.** Simon J. Godsill (Trumpington Street, CB21PZ Cambridge, UK, [sjg@eng.cam.ac.uk](mailto:sjg@eng.cam.ac.uk)), Ali-Taylan T. Cemgil (Trumpington Street, CB21PZ Cambridge, UK, [atc27@eng.cam.ac.uk](mailto:atc27@eng.cam.ac.uk)), Paul Peeling (Trumpington Street, CB21PZ Cambridge, UK, [php23@cam.ac.uk](mailto:php23@cam.ac.uk))

In this talk we describe Bayesian computational models and methods for inference about noisy audio signals, with the aim of performing tasks such as musical transcription, source separation, automated annotation with content descriptors, noise reduction and object-based coding. The models are structured models of non-stationary sparsity in audio sources, usually expressed in the time-frequency plane. The computational inference methods are based around Monte Carlo techniques including the particle filter for rapid on-line inference and Markov chain Monte Carlo for batch problems of higher complexity. Examples will be given from the spheres of source separation, multiresolution noise reduction, denoising and interpolation of missing batches from audio. We will also describe their use for acoustical analysis of the properties of a musical instrument, focusing on the parameters of inharmonicity in struck or plucked strings.

**11:00**

**4aSPb2. Bayesian approach to signal detection, source localization and ocean environmental parameter estimation.** Loren W. Nolte (Duke University, Department of Electrical and Computer Engineering, Durham, NC 27708, USA, [lwn@ee.duke.edu](mailto:lwn@ee.duke.edu))

From a Bayesian perspective, signal detection, source localization and ocean environmental parameter estimation algorithms can be viewed as simultaneous signal processing operation. In these cases, the likelihood function provides a mechanism for incorporating physical models of the ocean environment. In addition, one can incorporate uncertainties of source location and ocean environmental parameters a priori, rather than dealing with them "after the processing", and this approach also provides a posteriori probabilities of these source and ocean parameters as outputs of the processing. In addition to affecting the structure of optimal signal processing algorithms, these uncertainties affect how well one can spatially localize the source of sound, or the accuracy with which one can estimate the ocean environmental parameters. In passive and active sonar detection, these uncertainties, along with the knowledge of the ocean physics that has been incorporated in the likelihood function, determine detection performance. In particular, this paper will illustrate how signal detection theory can provide quantitative upper bounds of sonar detection performance on the receiver operating characteristic (ROC) as a function of the amount of uncertainty in ocean environmental parameters, source location, and signal-to-noise ratio.

**11:20**

**4aSPb3. Desoloing for musical accompaniment systems.** Lawrence J. Raphael (Adelphi University, 1 South Avenue, Garden City, NY 11530, USA, [raphael@adelphi.edu](mailto:raphael@adelphi.edu))

I discuss ongoing work for musical accompaniment systems in which we remove the soloist from a full recording for soloist and accompanying ensemble (e.g. a concerto), leaving just the accompaniment. I will discuss briefly the score matching problem, which generates a correspondence between a symbolic music representation and the audio. Using this score match, straightforward masking leads to usable source separation results, since the nature of the accompaniment problem partially compensates for the damage done by masking. However, I will discuss methods of improving this separation process involving imputing unobserved audio. The presentation will include a live demonstration of the accompaniment system.

**11:40**

**4aSPb4. Bayesian linear regression onto data-driven dictionaries with application to music transcription.** Cédric Févotte (CNRS-Telecom ParisTech, 37-39, rue Dareau, 75014 Paris, France, [fevotte@tsi.enst.fr](mailto:fevotte@tsi.enst.fr))

In this work, we describe a linear regression technique where features of the dictionary (the collection of explanatory variables) are learnt from the data itself. Our Bayesian setting allows to add regularization constraints on both the explanatory variables and the regressors that fit physical properties of sound. More precisely, smoothness constraints can be imposed on the first ones while time-

persistence and/or sparse constraints can be imposed on the second ones. When applied to music, we observe that the retrieved explanatory variables bear a semantic value and that the overall process yields a compact data-driven object-based transcription of the original signal.

12:00

**4aSPb5. Bayesian formant tracking using conditionally linear Gaussian models.** Patrick Wolfe (Harvard University, Harvard University - SEAS, 33 Oxford St Rm MD-129, Cambridge, MA 02138-2901, USA, patrick@seas.harvard.edu)

Formants play a central role in the perception and analysis of speech. In this presentation we describe Bayesian approaches to estimating vocal tract resonances from speech waveforms, formulated as a statistical model-based tracking problem. In particular, approaches by Deng and colleagues have shown the promise of an extended Kalman filtering approach based on a robust linearization of the formant-to-cepstrum map. We describe recent extensions to model inter- as well as intra-formant correlation, and detail efficient inference schemes that preserve conditional Gaussianity. A database of formant trajectories provides a notion of ground truth by which estimator performance can be evaluated, and which demonstrates the efficacy of our approach relative to contemporary benchmark tools for formant analysis.

12:20

**4aSPb6. Bayesian tracking and geoacoustic inversion.** Stan E. Dosso (University of Victoria, School of Earth and Ocean Sciences, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca), Michael J. Wilmut (University of Victoria, School of Earth and Ocean Sciences, Victoria, BC V8W 3P6, Canada, mjwilmut@uvic.ca)

This paper describes a Bayesian approach to two related inverse problems in underwater acoustics: localizing/tracking an acoustic source when ocean environmental properties are unknown, and determining environmental properties using acoustic data from an unknown (moving) source. The goal of this work is not simply to estimate values for source and/or environmental parameters, but to determine parameter uncertainty distributions, thereby quantifying the state of knowledge and information content of the inversion. A common formulation is applied for both problems in which source parameters (location and spectrum) and environmental parameters are considered unknown random variables constrained by noisy acoustic data and by prior information on parameter values (e.g., physical limits for environmental properties) and on inter-parameter relationships (limits on horizontal and vertical source speed). Given the strong nonlinearity of the inverse problem, marginal posterior probability densities are computed numerically using efficient Markov-chain Monte Carlo importance sampling methods. Source tracking results are represented by joint marginal probability distributions over range and depth, integrated over unknown environmental parameters. The approach is illustrated with two examples representing tracking a quiet submerged source and geoacoustic inversion using noise from an unknown ship-of-opportunity. In both cases, source, seabed, and water-column parameters are unknown.

12:40-2:00 Lunch Break

### *Invited Papers*

2:00

**4aSPb7. Geoacoustic Environment Tracking Using Kalman and Particle Filters.** Peter Gerstoft (Marine Physical Laboratory, Scripps Institute of Oceanography, 8602 La Jolla Shores Drive, La Jolla, CA 92093-0238, USA, gerstoft@ucsd.edu), Caglar Yardim (Marine Physical Laboratory, Scripps Institute of Oceanography, 8602 La Jolla Shores Drive, La Jolla, CA 92093-0238, USA, cyardim@ucsd.edu), William Hodgkiss (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, wsh@mpl.ucsd.edu)

This paper addresses the problem of tracking the geoacoustic environmental parameters such as the sound speeds in water, sediment and the bottom, sediment attenuation, density and thickness. The tracking is based on using acoustic measurements within an extended Kalman (EKF), unscented Kalman (UKF), and particle filter (PF) framework. The acoustic field is computed using a normal mode code which introduced a varying degree of nonlinearity depending on the environmental parameter of interest. Posterior Cramer-Rao lower bounds (PCRLB) are used to compute the tracking performances of the filters, including the filter efficiencies, divergence statistics, and computational complexities. The results showed that some of the parameters such as the water column parameters can be tracked by Kalman filters, however, the tracking performance of the Kalman filters was limited by the highly nonlinear relation between the sediment/bottom parameters and the acoustic field and non-Gaussian densities of these parameters. Particle filters proved to be very promising in tracking sediment layer parameters, even in the abruptly changing environments.

2:20

**4aSPb8. Inference and learning in gamma chains for Bayesian audio processing.** Ali Taylan Cemgil (University of Cambridge, Trumpington street, CB2 1PZ Cambridge, UK, atc27@cam.ac.uk), Onur Dikmen (Bogazici University, Dept. of Computer Engineering, 80815 Istanbul, Turkey, onuro@boun.edu.tr)

Statistical description of complex phenomena encountered in many applications requires construction of nonstationary models. A first step in analysis of such nonstationary sources involves typically a traditional time-frequency analysis Short time Fourier transform. In all these techniques, the underlying implicit assumption is that the process is piecewise stationarity, however dependencies across frequency bands or time frames are not explicitly characterised. Here, we investigate a class of prior models, called Gamma chains, for modelling such statistical dependencies in the time-frequency representations of signals. In particular, we model the prior variance of transform coefficients using Markov chains, trees and fields of inverse Gamma random variables. This model class is Markovian and conditionally conjugate, so standard inference methods like Gibbs sampling, variational Bayes or sequential Monte Carlo can be applied

effectively and efficiently. We also show how hyperparameters, that determine the coupling between prior variances of transform coefficients, can also be optimised. We discuss the pros and cons of various inference schemata (variational Bayes, Gibbs sampler and particle filtering) in terms of complexity and optimisation performance for this model class. We illustrate the effectiveness of our approach in audio denoising and single channel audio source separation applications.

2:40

**4aSPb9. A nonlinear frequency-domain beamformer for underdetermined speech mixtures.** Michael Davies (University of Edinburgh, ICDOM, Kings Buildings, Mayfield Road, EH9 3JL Edinburgh, UK, mike.davies@ed.ac.uk), Mohammad Dmour (University of Edinburgh, Institute for Digital Communications, School of Engineering & Electronics, Alexander Graham Bell Building, Kings Buildings, Mayfield Road, EH9 3JL Edinburgh, UK, m.dmour@ed.ac.uk)

Extraction of a target speech source from among multiple interfering speech sources is challenging when there are fewer microphones than sources (the underdetermined case). Existing speech source separation techniques often suffer from artifacts as well as performance deterioration in reverberant environments, and in some cases also need to estimate the number of sources present. This paper introduces a frequency-domain non-linear beamformer that can perform speech source separation of underdetermined mixtures, is reasonably artifact free and does not require prior knowledge of the number of speakers. Our approach models the data via a Gaussian mixture distribution in the observation domain, which can be learned using the expectation maximization (EM) algorithm. A non-linear distortionless beamformer is then developed, based on this model. Simulations of the non-linear beamformer in underdetermined mixtures with room reverberation confirm its capability to successfully separate speech sources with virtually no artifacts.

3:00

**4aSPb10. Bayesian model selection applied to room-acoustic energy decay analysis.** Tomislav Jasa (Institute fur Neuroinformatik ETH/UNIZ, Winterthurerstrasse 190, 8057 Zurich, Switzerland, jasa@ini.phys.ethz.ch), Ning Xiang (Rensselaer Polytechnic Institute, Greene Building, School of Architecture, 110 8th Street, Troy, NY 12180, USA, xiangn@rpi.edu)

The previous work of Xiang et al. [Xiang & Goggans, J. Acoust. Soc. Am. 110 (2001), pp. 1415-1424; 113 (2003), pp. 2685-2697; Xiang & Jasa, J. Acoust. Soc. Am. 120 (2006) pp.3744-3749] successfully applied the Bayesian formalism to estimate multiple decay parameters from Schroeder decay functions measured or calculated in acoustically coupled spaces. In this work we consider a more difficult problem of determining the correct decay model in the presence of energy decay data within the Bayesian framework. We will compare the Annealing/Thermodynamic algorithm [Neal, R. M., Statistics and Computing 11 (2001), pp. 125-139], the Nested Sampling algorithm [Silvia & Skilling, Data Analysis: A Bayesian Tutorial, Oxford Science Publications (2006)], and a combined Variational/Markov Chain Monte Carlo approach in order to determine the correct decay model. The advantages/shortcomings of these methods are discussed in the context of the decay model selection and parameter estimation using experimentally measured data from real coupled spaces.

3:20

**4aSPb11. Efficient Bayesian inference for multiple pitch estimation of music audio.** Emmanuel Vincent (INRIA Rennes Bretagne Atlantique, Campus de Beaulieu, 35042 Rennes Cedex, France, emmanuel.vincent@irisa.fr)

Multiple pitch estimation consists of estimating the number of active notes and their fundamental frequencies on each time frame of a music signal. This is a core problem for several applications, including score transcription and source separation. Bayesian harmonic models are a promising approach, since they allow the joint exploitation of various priors on the model parameters. However existing Bayesian inference methods often rely on specific prior distributions and remain computationally demanding for realistic data. We propose a generic inference method based on adaptive factorization of the joint posterior that allows the application of such models to real-world data. We evaluate the results for the task of multiple pitch estimation using different levels of factorization.

3:40

**4aSPb12. Bayesian single channel blind speech dereverberation using Monte Carlo methods.** James R. Hopgood (Institute for Digital Communications, The University of Edinburgh, School of Engineering and Electronics, Alexander Graham Bell Building, The King's Buildings, Mayfield Road, EH9 3JL Edinburgh, UK, James.Hopgood@ed.ac.uk), Christine Evers (Institute for Digital Communications, The University of Edinburgh, School of Engineering and Electronics, Alexander Graham Bell Building, The King's Buildings, Mayfield Road, EH9 3JL Edinburgh, UK, c.evers@ed.ac.uk), Judith Bell (Heriot Watt University, School of Engineering and Physical Sciences, Riccarton, EH14 4AS Edinburgh, UK, j.bell@hw.ac.uk)

Audio signals in confined spaces exhibit reverberation due to reflections off surrounding obstacles. Moreover, the signal is distorted by noise, usually modeled as an additive signal observed within the room, independent of the microphone's location, and unaffected by the acoustics. Reverberation and noise cause significant deterioration of audio quality and intelligibility to signals recorded in acoustic environments. Bayesian blind dereverberation infers knowledge about the system by exploiting the statistical properties of speech and the acoustic channel. In the Bayesian framework, the reverberant and noisy signal can be enhanced by processing it either sequentially using online methods or in a batch using offline methods. This paper compares several distinct Bayesian approaches for single-channel blind speech dereverberation. These include Markov chain Monte Carlo methods for batch processing, and sequential Monte Carlo (particle filtering) methods for online processing. In the batch method, static parametric models are used for modeling the statistics of the speech and channel. Optimal parameter estimates are then used to enhance the observed signal. In the sequential approach, the clean speech signal is considered itself an unknown state. Various dynamical models and optimal sampling strategies are discussed for state estimation. The results demonstrate the superiority of the sequential method.

## Session 4aUWa

**Underwater Acoustics, Acoustical Oceanography, and ECUA: Impact of Internal Waves on Shallow Water Propagation**

James F. Lynch, Cochair

*Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA*

Thomas Folégot, Cochair

*NATO Undersea Research Center, Viale San Bartolomeo 400, La Spezia, 19126, Italy**Invited Paper*

8:00

**4aUWa1. Spatial and temporal sound field fluctuations due to propagating internal waves in shallow water.** Mohsen Badiey (University of Delaware, College of Marine and Earth Studies, S. College Street, Newark, DE 19716, USA, badiey@udel.edu), Boris Katsnelson (Voronezh State University, 1, Universitetskaya sq., 394006 Voronezh, Russian Federation, katz@phys.vsu.ru), James F. Lynch (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, jlynch@whoi.edu)

Fluctuating three dimensional acoustic wave propagation in shallow water presents a challenge due to the anisotropic nature of the environment. Temporal and spatial changes in the water column caused by the passage of internal waves are among the primary causes of the anisotropy. Three main mechanisms for the acoustic field variations have been determined: 1) adiabatic propagation, 2) horizontal refraction, and 3) mode coupling. All these mechanisms provide different characteristics of the sound fluctuations, and depend on the angle between the internal wave direction and the source receiver propagation path, as well as frequency and mode number. In a recent multi-institutional shallow water experiment (SW06) a series of source-receiver positions were created to examine the effects of 3D environmental fluctuations on the propagation of low to mid frequency (0.05-3 kHz) broadband acoustic signals while extensive environmental observations were made. These recent observational results confirm the above theoretical hypothesis. Work supported by ONR-3210A and RFBR.

*Contributed Papers*

8:20

**4aUWa2. Sound intensity variations in the presence of shallow-water internal waves passing through acoustic track.** Jing Luo (University of Delaware, College of Marine and Earth Studies, S. College Street, Newark, DE 19716, USA, luojing@udel.edu), Mohsen Badiey (University of Delaware, College of Marine and Earth Studies, S. College Street, Newark, DE 19716, USA, badiey@udel.edu), Entin A. Karjadi (College of Marine and Earth Studies, University of Delaware, Newark, DE 19716, USA, karjadi@udel.edu), Boris Katsnelson (Voronezh State University, 1, Universitetskaya sq., 394006 Voronezh, Russian Federation, katz@phys.vsu.ru), Alexander Tshoidze (Voronezh State University, 1, Universitetskaya sq., 394006 Voronezh, Russian Federation, tshoidze@phys.vsu.ru), James Moum (College of Oceanic and Atmospheric Sciences, Oregon State University, 104 COAS Administration Bldg, Corvallis, OR 97331, USA, jmoum@coas.oregonstate.edu), James F. Lynch (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, jlynch@whoi.edu)

Fluctuations of low frequency pulses (LFM signals in 270-330 Hz band) in the presence of internal solitary wave (ISW) packet during the SW06 experiment are analyzed to quantify the interaction of sound with ISW field. Three situations during approximately 2 hours (20:30 - 22:30 GMT of August 17, 2006) are considered: a period when ISW was absent, a period when ISW started to intersect the acoustic track, and a period when ISW occupied the acoustic track. The propagation direction, velocity, and amplitude of the ISW were estimated from the on-board radar images recorded by two research vessels along with temperature records of sensors moored at the source, at the receiver, and between the source-receiver track. Modal and frequency filtering of received pulses was carried out as well as analysis

of temporal variations of the field depth distribution. This analysis allows us to identify two different acoustic fluctuation mechanisms: horizontal refraction and adiabatic variability, and it confirms the previously proposed theory on the sound field fluctuations due to ISW. Work was supported by ONR and RFBR.

8:40

**4aUWa3. Short-range acoustic propagation through non-linear internal waves.** Daniel Rouseff (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, rouseff@apl.washington.edu), Dajun Tang (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, djtang@apl.washington.edu), Kevin L. Williams (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, williams@apl.washington.edu), James Moum (College of Oceanic and Atmospheric Sciences, Oregon State University, 104 COAS Administration Bldg, Corvallis, OR 97331, USA, jmoum@coas.oregonstate.edu), Zhongkang Wang (Hangzhou Applied Acoustics Research Institute, PO Box 1249, 310012 Hangzhou, China, zkwang@apl.washington.edu)

During the Shallow Water 2006 Experiment (SW06), mid-frequency acoustic transmission data were collected on a vertical array over a continuous 7-hour period at range 550 m. The relatively short range was deemed desirable for studying the effects of internal waves; individual waves in a packet of nonlinear internal waves might be isolated between the acoustic source and receiver. Of present interest are data immediately before, during and after the passage of a non-linear internal wave on 18 August 2006. Among other features, the data show a new acoustic path being generated as the internal wave passes the acoustic source. A ray-based model is developed for the observed effect that uses as input nearby oceanographic measurements. [Work supported by ONR.]



9:00

**4aUWa4. Intensity fluctuations of mid-frequency sound signals passing through moving nonlinear internal wave in experiment sw06.** Boris Katsnelson (Voronezh State University, 1, Universitetskaya sq., 394006 Voronezh, Russian Federation, katz@phys.vsu.ru), Valery Grigorev (Voronezh State University, 1, Universitetskaya sq., 394006 Voronezh, Russian Federation, grig@box.vsi.ru), James F. Lynch (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, jlynch@whoi.edu), Dajun Tang (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, dtang@apl.washington.edu)

Fluctuations of an intensity of the broadband pulses are studied in mid-frequency area (2 - 4.5 kHz) propagating in the shallow water in the presence of intensive internal waves (IW) moving approximately along an acoustic track. Theory elaborated earlier predicts that in this case specific features of fluctuations are provided by modes coupling (for low frequency sound) or ray scattering (high frequency area) and depend on direction of propagation of signals relative wave front of IW. The corresponding research was carried out during multi-institutional experiment sw06 in New Jersey shelf. We analyze temporal dependence of intensity for the sequence of the sound pulses radiated from the R/V Knorr during approximately one hour - 15:31 -16:20 GMT (August 13, 2006) and received by two separate single hydrophone units (SHRUs) placed at different distance from the source (~4 km and ~12 km). The corresponding acoustic tracks had a little different directions relative wave front of IW. Properties of IW were established using temperature records of sensors in different locations. It is shown that frequency spectra of fluctuations for these SHRUs have different predominating frequencies in accordance with mentioned directions of acoustic tracks. Results of measurements are compared with theoretical estimations and demonstrate good consistency.

9:20

**4aUWa5. Simultaneous nearby measurements of acoustic propagation and high-resolution sound speed structure containing internal waves.** Frank S. Henyey (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, frank@apl.washington.edu), Kevin L. Williams (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, williams@apl.washington.edu), Dajun Tang (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, dtang@apl.washington.edu)

On two occasions during the SW06 experiment, towed CTD chain measurements were made close to an acoustic propagation path. The acoustic path was 1 km long, oriented roughly in the direction of propagation of large nonlinear internal waves. On the first occasion, large nonlinear internal waves were absent, and on the second occasion, they were present. The CTD chain was towed in loops around the acoustic path, roughly 200 m on either side of the path. On the first occasion, 17 loops were made in about 5.5 hr, and on the second occasion, 7 loops were made in about 2.5 hr. Throughout these time periods, acoustic transmissions between 2 kHz and 10 kHz were carried out. The acoustic environment on the path is estimated by space and time interpolation between the tows on the two sides of the path. The acoustic data is compared with propagation modeling in this environment.

9:40

**4aUWa6. The effects of non-linear internal wave curvature on acoustic propagation.** Timothy Duda (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 107, MS-12, Woods Hole, MA 02543, USA, tduda@whoi.edu), James F. Lynch (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, jlynch@whoi.edu), Ying-Tsong Lin (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 107, MS-12, Woods Hole, MA 02543, USA, ytl@whoi.edu), Arthur Newhall (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, anewhall@whoi.edu), Hans Graber (RSMAS - University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149, USA, hgraber

@rsmas.miami.edu), Micheal Caruso (RSMAS - University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149, USA, mcaruso@rsmas.miami.edu)

It is well known that coastal nonlinear internal waves can have curved wavefronts, due to either being generated at a "point source" such as a submarine canyon or valley or due to the wave being refracted by large scale water column or bathymetric features. We will explore the first case here, with the emphasis on circular wave fronts. Three cases of acoustic propagation in the vicinity of a circular wavefront will be considered: 1) the source within a soliton train, 2) the source in front of the soliton train, and 3) the source behind the soliton train. Simple geometric forms are presented showing how the curved wavefronts can lead to mode number and frequency dependent dispersion and shadowing effects. Work sponsored by Office of Naval Research.

10:00

**4aUWa7. The effects of the internal waves on the statistic characteristics of the sound fields in the 2001 ASIAEX South China Sea Experiment.** Lixin Wu (National Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, NO.21, Bei-Si-huan-Xi Road, 100080 Beijing, China, wxl@mail.ioa.ac.cn), Jun Wang (National Lab. of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, NO.21, Bei-Si-huan-Xi Road, 100080 Beijing, China, wangj@mail.ioa.ac.cn), Shaoqiang Wang (National Lab. of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, NO.21, Bei-Si-huan-Xi Road, 100080 Beijing, China, wsq@mail.ioa.ac.cn), James F. Lynch (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, jlynch@whoi.edu), Timothy Duda (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 107, MS-12, Woods Hole, MA 02543, USA, tduda@whoi.edu)

Internal waves are one of primary sources of ocean variations in shallow water. The temporal-spatial stability of sound channel may be degraded by the activities of internal waves. We present analyses of statistic characteristics observed in acoustic signals transmitted by two 400Hz sources moored as part of ASIAEX 2001 South China Sea (SCS) experiment. One source was 31.3 km offshore from the receiving array, and the other was 20.6 km alongshore from the array. Time series of signal intensity measured at individual phones of a 16-element vertical line array and a 32-element horizontal line array. temporal-spatial correlations were observed from 2 May to 17 May 2001. The temporal-spatial decorrelation scales are closely related to internal wave properties. The largest internal wave, especially solitary wave packets, are the principal contributors to reduction of the temporal-spatial decorrelation scales. We also present elementary analyses of higher-order statistics (HOS), such as bispectrum, bicoherency coefficients, etc.

10:20-10:40 Break

10:40

**4aUWa8. Observed acoustic arrival structure and intensity variability induced by transbasin nonlinear internal waves in the South China Sea basin.** Chingsang Chiu (Naval Postgraduate School, Department of Oceanography, 833 Dyer Road, Rm 328, Monterey, CA 93943-5738, USA, chiu@nps.edu), D. Benjamin Reeder (Naval Postgraduate School, Department of Oceanography, 833 Dyer Road, Rm 328, Monterey, CA 93943-5738, USA, dbreeder@nps.edu), Christopher Miller (Naval Postgraduate School, Department of Oceanography, 833 Dyer Road, Rm 328, Monterey, CA 93943-5738, USA, cwmiller@nps.edu), Justin Reeves (Naval Postgraduate School, Department of Oceanography, 833 Dyer Road, Rm 328, Monterey, CA 93943-5738, USA, jmreeves@nps.edu), Steve Ramp (Naval Postgraduate School, Department of Oceanography, 833 Dyer Road, Rm 328, Monterey, CA 93943-5738, USA, sramp@nps.edu), Yiing-Jang Yang (Naval Academy, Department of Marine Science, P.O. Box 90175, Tsoying Kaohsiung, Taiwan, yjyang@cna.edu.tw), Ruey-Chang Wei (National Sun Yat-sen University, 70 Lien-Hai Road, Institute of Undersea Technology, 804 Kaohsiung City, Taiwan, rcwei@mail

During the 2005-2006 Windy Island Nonlinear Soliton Experiment (WISE), large-amplitude internal waves were observed by in situ oceanographic sensors to propagate eastward from the Luzon Strait, through the deep basin, onto the shelf and Dongsha Plateau of the South China Sea year round except winter. In a parallel effort to measure the effects of these trans-basin internal waves on long-range, low-frequency sound propagation, an acoustic source with a center frequency of 400-Hz and bandwidth of 100-Hz was moored on the west side of the deep basin transmitting a phase-modulated m-sequence signal every 15 min from February to October 2006. These periodic transmissions were recorded by a receiver moored ~166 km to the east of the source. The recording was processed to give the arrival structure of a pulse and its temporal changes over the eight-month period. The observed temporal variability in the arrival structure and in the acoustic intensity were analyzed using time-series techniques and models with emphasis to elucidate the connection to the observed sound speed variability induced by the nonlinear transbasin internal waves. Results from the analysis are presented and discussed. [The research is sponsored by the US ONR and the Taiwan NSC.]

#### 11:00

**4aUWa9. Spatial and temporal coherence of low-frequency acoustic field in shallow water: Experimental results.** Lianghao Guo (National Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, NO.21, Bei-Si-huan-Xi Road, 100080 Beijing, China, glh2002@mail.ioa.ac.cn), 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), Lixin Wu (National Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, NO.21, Bei-Si-huan-Xi Road, 100080 Beijing, China, wlx@mail.ioa.ac.cn), Xilu Li (National Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, NO.21, Bei-Si-huan-Xi Road, 100080 Beijing, China, glh2002@mail.ioa.ac.cn)

Spatial and temporal coherence of acoustic field has very important effects on applications of underwater acoustics. In this paper, recent experimental results of low-frequency signal coherence in shallow water are presented. For signals with low frequencies of 100~500Hz in shallow water, the vertical correlation has no distinct depth dependence, but it has obvious range dependence. The horizontal correlation length is greater than 40 wavelengths, the temporal correlation length is greater than 510s at frequency of 475Hz and greater than 1800s at frequency of 150Hz. These experimental results show that low-frequency acoustic field has strong spatial coherence and temporal stability in shallow water. [Work supported by the National Natural Science Foundation of China under Grant No.10734100]

#### 11:20

**4aUWa10. Temporal correlation of MFP with the presence of internal waves.** Zhenglin Li (Haikou Acoustic Lab. & National Lab. of Acoustics, Inst. of Acoustics, Chinese Academy of Sciences, 63 Binghaidadao, 570105 Haikou, China, lzhl@mail.ioa.ac.cn), Guihua Ji (Haikou Acoustic Lab. & National Lab. of Acoustics, Inst. of Acoustics, Chinese Academy of Sciences, 63 Binghaidadao, 570105 Haikou, China, jgh@mail.ioa.ac.cn), Qiongxing Dai (Haikou Acoustic Lab. & National Lab. of Acoustics, Inst. of Acoustics, Chinese Academy of Sciences, 63 Binghaidadao, 570105 Haikou, China, dqx66@163.com)

Internal wave is a dominant source of ocean uncertainties in shallow waters. The ability of passive source localization may be degraded due to mismatch between model predictions and measurements caused by the activities of internal waves. Using ocean environment measurements from an experiment, the effects of Garrett-Munk and solitary internal waves on the temporal correlation of matched-field processing (MFP) in shallow water for sources with different frequencies and different depths are numerically investigated. It is shown that the temporal correlation of MFP decreases as the amplitude of solitons or the average energy density of linear internal waves increased. For a source with lower frequency or located below the

thermocline, the temporal correlation of MFP is less affected by internal waves, and the length of which is longer. Moreover, the effects of the range between solitons and source on the temporal correlation of MFP are relatively small.

#### 11:40

**4aUWa11. Observations of noise generated by nonlinear internal waves on the continental shelf during the SW06 experiment.** Andrew N. Serebryany (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, aserebryany@whoi.edu), Arthur Newhall (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, anewhall@whoi.edu), James F. Lynch (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, jlynch@whoi.edu)

As part of the Shallow Water 2006 (SW06) experiment, simultaneous measurements of coastal internal wave oceanography and ocean acoustics were made over a two month period in late summer. The generation of noise by nonlinear internal waves propagating on the shelf during the SW06 experiment was observed, and is reported upon here. Three main types of noise were observed: bed noise, mid-column noise and noise from the sea surface. Surface noise is created due to an enhancement of surface wave breaking in the convergence zone created by the internal waves. Strong broadband bed noise was observed during the moments of internal wave passage above a horizontal array of hydrophones lying on the bottom. Appearance of bed noise in the form of several spikes we observed coinciding with the strongest bottom currents created by internal wave orbital currents. For the case of breaking internal waves, the near-bed spike-like noise disappeared and mid-column noise of a different character was seen instead. Mechanisms of the various types noise generation will be discussed. Work sponsored by ONR.

#### 12:00

**4aUWa12. Applying the Data Nullspace Projection Method to a Geoacoustic Bayesian Inversion in a Randomly Fluctuating Shallow-Water Ocean.** James F. Lynch (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, jlynch@whoi.edu), Ying-Tsong Lin (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 107, MS-12, Woods Hole, MA 02543, USA, ytlin@whoi.edu), Arthur Newhall (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, anewhall@whoi.edu)

Bayesian inversion techniques which are commonly used in geoacoustic inversion can suffer the effects of uncertain water-column fluctuations. To reduce these effects, one could also invert for the fluctuating water-column parameters; however, there are issues with this approach. One obvious problem is that the dimensions of parameter space will increase, so that Bayesian inversion may not be efficient. Another issue arises from the temporal and spatial randomness and variability of the water-column parameters; this requires extra effort in handling the randomness and variability in the inversion procedure. In this paper, we propose another approach to the problem. The data nullspace projection method, which has been applied to perturbative inversion, is extended to Bayesian inversion using acoustic modal wavenumbers and group velocities. The idea of this method is to project acoustic data onto a subspace that is insensitive to uncertain water-column fluctuations, and use the projected data to invert for bottom properties. The advantage of this approach is that we do not need to invert for water-column parameters, so that the inversion requires less operations than the previous approach. A numerical simulation demonstrates the feasibility of the projection approach. It is then applied to real data collected in the SW06 experiment.

#### 12:20

**4aUWa13. Acoustic propagation modeling in the presence of environmental uncertainty.** Yu Yu Khine (USNRL, 4555 Overlook Ave SW, Washington, DC 20375, USA, yuyu.khine@nrl.navy.mil), Steven

Finette(USNRL, 4555 Overlook Ave SW, Washington, DC 20375, USA, steven.finette@nrl.navy.mil), Roger Oba (USNRL, 4555 Overlook Ave SW, Washington, DC 20375, USA, roba@wave.nrl.navy.mil)

Due to incomplete knowledge of ocean environments, this research incorporates environmental uncertainty into an acoustic model representing wave propagation in order to quantitatively represent the uncertainty of the acoustic field. The waveguide considered here has a spatially varying, uncertain sound speed distribution with a known correlation length. Karhunen-Loeve and polynomial chaos expansions are used to represent the uncertainty in the environment and acoustic field, respectively, in a narrow-angle

parabolic wave equation. In this two-dimensional model, the water depth is 150 m and propagation is over a range of 20 km. The environmental uncertainty term in the wave equation is assumed to vary randomly in the range direction, and is characterized by an exponentially decaying correlation function. An implicit finite difference scheme is used to solve a set of coupled differential equations for the stochastic envelope function at different source frequencies in the range of a few hundred Hz. The simulated results will include probability density functions at selected spatial locations in the waveguide, first and second moments of the field, and these results will be compared with those obtained independently from Monte Carlo samplings from the same ocean environment. Work supported by ONR.

## 12:40-1:40 Lunch Break

### Contributed Papers

1:40

**4aUWa14. New Numerical Computation of Acoustic Propagation in the Ocean in the Presence of Internal Waves is 1000 Times Faster than Traditional Split-Step Fourier Approaches.** Alfred R. Osborne (University of Torino, Dipartimento di Fisica Generale, Via Pietro Giuria 1, 10124 Torino, Italy, al.osborne@gmail.com)

A new numerical model for acoustic propagation based upon the large angle parabolic type equations is found to execute about 1000 times faster than the FFT split step algorithm. The approach is applied to the imaging and nonlinear filtering of internal waves in shallow water regions for 3-dimensional propagation of acoustic waves. For appropriate array placement one can construct the inverse problem and hence the internal wave field from the acoustic measurements themselves. The unique nonlinear filtering method allows one to filter out the internal waves from the density field and hence to realized acoustic wave propagation in the absence of the internal waves. The hyperfast acoustic model has some of the following characteristics: (1) The work can be extended to the fully 3-dimensional Helmholtz equation. (2) The method is a kind of multi-dimensional Fourier analysis which exactly solves the wave equations (PE, large angle PEs and Helmholtz) so that the acoustic wave does not degrade with range as with more traditional numerical integrations. (4) Multi-dimensional Fourier transforms can be used to nonlinearly filter acoustic signals in a wide range of applications and hence the approach is quite robust to interference from background acoustic noise.

2:00

**4aUWa15. Modeling probability density functions for acoustic propagation through internal waves in shallow water environments.** Kevin D. Lepage (Naval Research Laboratory, 4555 Overlook Ave SW, Washington, DC 20375, USA, kevin.lepage@nrl.navy.mil)

Acoustic propagation through internal waves in shallow water introduces randomness to the acoustic field in that internal waves are generally incompletely measured and are therefore best understood as random realizations of a stochastic process. Due to the physics of internal waves these environmental perturbations are expressed in a finite number of modes, each of which

affects acoustic propagation differently. As acoustic propagation in shallow water is itself confined into modes, a matrix of interaction strengths for both the accumulated phase and the mode coupling may be derived which can be integrated forward along the acoustic path to account for the accumulation of uncertainty. A powerful ansatz for solving this problem is a Polynomial Chaos (PC) expansion of the complex modal amplitudes in the random variables which are the internal wave amplitudes. In this work the PC technique is used to derive the probability density functions of the complex modal amplitudes in the presence of a homogeneous internal wave field, showing good agreement with Monte-Carlo results. The pdfs of the complex modal amplitudes can then be combined to obtain the pressure field pdf as well as various statistics of interest such as the scintillation index. [Work supported by ONR]

2:20

**4aUWa16. A modified dnoidal model for internal solitary waves and its effect on sound transmission.** Shaoqiang Wang (National Lab. of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, NO.21, Bei-Si-huan-Xi Road, 100080 Beijing, China, wsq@mail.ioa.ac.cn), Lixin Wu (National Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, NO.21, Bei-Si-huan-Xi Road, 100080 Beijing, China, wx@mail.ioa.ac.cn), Changqing Hu (Shanghai Acoustics Lab., Chinese Academy of Sciences, No.456, Xiao-Mu-Qiao Road, Xu-Hui District, 200032 Shanghai, China, hchq@mail.ioa.ac.cn)

A new solution of KDV equation in the form of dnoidal function is developed in this paper. Based on this new solution, a modified solution of KDV equation with slowly varying parameters in the form of dnoidal function is derived. Numerical simulation shows that the wave packets scale of the modified solution is only 1/5 to the solution in the reference (JASA, 108 (3), pp 957-972), and also the equilibrium positions are different. Comparing the solutions with slowly varying parameters with the published data of internal solitary waves in the ocean indicates that the modified solution is better for the description of the dnoidal wave. Based on the modified solution of KDV equation with slowly varying parameters, a modified model for dnoidal wave is given and its effect on sound transmission is discussed in this paper.

**Session 4aUWb****Underwater Acoustics and ECUA: Determination of Acoustic Properties of Materials for Sonar Applications I**

Kenneth G. Foote, Cochair

*Woods Hole Oceanographic Institution, Woods Hole, MA 02543, USA*

Stephen P. Robinson, Cochair

*National Physical Laboratory, Hampton Road, Teddington, TW11 0LW, UK***Invited Papers****8:00**

**4aUWb1. Phase change measurement, sound speed and attenuation determination from underwater acoustic panel tests.** Jean Piquette (Naval Undersea Warfare Center, 1176 Howell Street, Newport, RI 02841, USA, paoleroae@npt.nuwc.navy.mil), Anthony Paolero (Naval Undersea Warfare Center, 1176 Howell Street, Newport, RI 02841, USA, paoleroae@npt.nuwc.navy.mil), Robert Drake (Naval Undersea Warfare Center, 1176 Howell Street, Newport, RI 02841, USA, Drakerm@npt.nuwc.navy.mil)

Material measurements in underwater acoustics are employed regularly for characterization of materials used as acoustic windows. Various techniques have been developed to measure the acoustic attenuation magnitudes of materials. Recently, there has been increased interest in window configurations of complex geometries and of phase response of that material to an acoustic wave. This latter phase parameter can be of high significance for acoustic devices composed of an array of elements. Thus, the phase response of the attenuation is needed to properly characterize material performance on the overall system. This translates into a primary need to measure the complex acoustic attenuation of panels at various angles of incidence. This paper summarizes the methodology to measure the complex attenuation of materials. Case examples of test materials measured will be presented; comparisons to theoretical response will be provided. Considerations given to rigging and required acoustic settling times are discussed. Expansion of the methodology that incorporates a causal relationship between the measured phase response and attenuation to the sound speed of the material will be developed. The model relating these parameters and their implementation via an iterative least squares fitting of the parameters to the measurement data will be discussed.

**8:20**

**4aUWb2. The underwater acoustic testing of modest sized panel materials using a multi-element array technique in a laboratory test vessel.** Michael J. Martin (QinetiQ Ltd, Rm 1146, Bldg A7, Cody Technology Park, Ively Road, GU14 0LX Farnborough, UK, mjmartin1@qinetiq.com), Stephen P. Robinson (National Physical Laboratory, Hampton Road, TW11 0LW Teddington, UK, Stephen.Robinson@npl.co.uk), John Smith (DSTL, Rm 14, Bldg 352, Porton Down, SP4 0JQ Salisbury, UK, jdsmith@dstl.gov.uk), Victor F. Humphrey (Institute of Sound and Vibration, Univ. of Southampton, University Road, Highfield, SO17 1BJ Southampton, UK, vh@isvr.soton.ac.uk)

The underwater acoustic properties of materials can be assessed in panel form via a simple single hydrophone measurement technique which is ideally conducted on large test panels in open water. Alternatively measurements may be conducted in a laboratory test vessel which is capable of simulating ocean conditions. However, physical limitations imposed by the constraints of the vessel, including a reduction in test panel dimensions, require modification of the traditional technique to allow measurements to be made down to low kilohertz frequencies. This can be achieved by, in addition to a directional source, the use of a directional receiver in the form of a planar multi-hydrophone array. The technique is described and illustrated with measurements of reflection loss and insertion loss between 1 kHz and 7 kHz performed on a test panel in a pressure vessel through changing hydrostatic pressure from ambient to 2.8 MPa. The efficacy of this technique is considered with simulated multi-hydrophone array measurements on a scaled panel in an open tank.

**8:40**

**4aUWb3. Panel transmission measurements: The influence of the non plane wave nature of the incident field.** Victor F. Humphrey (Institute of Sound and Vibration, Univ. of Southampton, University Road, Highfield, SO17 1BJ Southampton, UK, vh@isvr.soton.ac.uk), John Smith (DSTL, Rm 14, Bldg 352, Porton Down, SP4 0JQ Salisbury, UK, jdsmith@dstl.gov.uk)

For reasons of cost and practicality, laboratory measurements of the acoustic transmission and reflection properties of materials for use in underwater applications are typically performed on samples of limited dimensions - and with the source and receiver separated by relatively short distances - resulting in a non planar measurement field. The influence of this on the resulting measurements is investigated in this paper. In particular, for low frequency measurements the influence of the evanescent wave contributions can become significant. In this paper two alternative approaches are used to evaluate the transmission properties. The first method decomposes the incident spherical wave into its plane wave components and integrates the resulting transmitted waves numerically to evaluate the transmitted field. The second approach uses an asymptotic expansion of the field in terms of wave front curvature: bounds are then



placed on the error in this expansion at low frequency using thin plate theory. Results are compared and contrasted for measurements in the frequency range 1 to 60 kHz for panels of simple elastic materials (steel and Perspex (polymethylmethacrylate)). In addition the nature and significance of the modes of the panel for evanescent waves are considered. The consequences for laboratory measurements are also outlined.

9:00

**4aUWb4. Comparisons of the dynamic moduli of various polymers.** John Smith (DSTL, Rm 14, Bldg 352, Porton Down, SP4 0JQ Salisbury, UK, jdsmith@dstl.gov.uk)

To fully characterise the elastic properties of a homogeneous, isotropic material, two independent elastic constants are needed, usually the bulk and shear moduli. For the types of polymers used in sonar applications, these moduli are typically both frequency and temperature dependent. Of particular interest is the position (in frequency and temperature) of peak loss tangent, which is related to the glass transition temperature through the free volume concept. Although there exists in the literature a lot of data on the dynamic shear (and Young's) modulus for various polymers, data on dynamic bulk modulus is scarcer, due to the difficulty of the measurement. This paper reviews the current literature on the dynamic moduli of various polymers, with particular regard to the relationship between the peak loss tangents of the bulk and shear moduli. Simple relaxation models are studied to give insight on the factors affecting the peak of the loss tangent and new measurements on a nitrile-butadiene rubber compound are presented.

9:20

**4aUWb5. Pulse tube measurement of bulk modulus of visco-elastic composite materials: Theory and practice.** Peter R. Brazier-Smith (Thales, Ocean House, Somerset, BA8 0DH Templecombe, UK, peter.brazier-smith@uk.thalesgroup.com), Allan R. Clark (Thales, Ocean House, Somerset, BA8 0DH Templecombe, UK, allan.clark@uk.thalesgroup.com)

A method for the determination of the bulk moduli and loss factors of micro-voided composite materials is presented. The method requires that the reflection and transmission coefficients of a tile of uniform thickness are determined in both amplitude and phase as functions of frequency. Reduction to the bulk modulus and loss factor then proceeds by using the analytic properties of a function of a complex variable derived from the reflection and transmission coefficients. A pulse tube is used for the determination of the complex reflection and transmission coefficients. Although other measurement techniques are available, the pulse tube has proved to be versatile in covering a large range of the frequency-temperature master curve for typical composite materials used in underwater acoustics. It achieves this versatility by using an anti-freeze/water mixture as the medium following which measurements can be made over a range of different temperatures.

### Contributed Papers

9:40

**4aUWb6. Determining dynamic viscoelastic properties without time-temperature shifting.** Walter M. Madigosky (Catholic University of America, Department of Physics, Washington, DC, DC 20064, USA, wmadigosky@yahoo.com), Gilbert F. Lee (NSWCCD, 9500 McArthur Boulevard, Bethesda, MD 20817, USA, gilbert.f.lee@navy.mil), Jan M. Niemiec (NSWCCD, 11798 Fox Rest Court, New Market, MD 21774, USA, jan301@comcast.net)

A novel unbiased procedure to analyze dynamic mechanical data of rheologically simple viscoelastic polymers by modeling the data with the Havriliak and Negami (HN) equation is described. The real and imaginary parts of the HN equation are used to solve for the global frequency-time dependent parameter ( $\omega\tau$ ) for all the data thereby uniquely determining the time-temperature shift function. Displaying the experimental data in the form of a wicket or Argand diagram provides initial estimates for the HN parameters. An unbiased error analysis is performed to minimize the difference between experimental and calculated complex viscoelastic values. Finally the characteristic relaxation time,  $\tau_b > 0$ , is determined by minimizing the error between the calculated data and experimental data at the reference temperature  $\langle T \rangle < 0 \rangle$ . Using this procedure, the complete master curve is generated without the need for overlapping frequency data and the procedural error and operator bias associated with time-temperature shifting is eliminated. The technique can also generate a complete frequency spectrum from isochronal temperature scans such as those obtained from a torsion-pendulum, rheovibron or a Dynamic Mechanical Analyzer (DMA) apparatus, which is not otherwise possible.

10:00

**4aUWb7. A new integral equation method for elastic composites.** Natasha J. Willoughby (The University of Manchester, School of Mathematics, Oxford Road, M13 9PL Manchester, UK, Natasha.J.Willoughby@student.manchester.ac.uk), William J. Parnell (The University of Manchester, School of Mathematics, Oxford Road, M13 9PL Manchester, UK, William.Parnell@manchester.ac.uk), I David Abrahams (The

University of Manchester, School of Mathematics, Oxford Road, M13 9PL Manchester, UK, David.Abrahams@manchester.ac.uk)

A new integral equation approach to elastodynamic homogenization has been proposed [1] to determine the effective properties of periodic fibre reinforced composite materials in the case of SH wave propagation. When all the fibres are aligned in the same direction and considered infinite in extent, the microstructure is two-dimensional. The integral equation method is based on Navier's equations of elasticity in integral equation form, and uses the notion of separation of scales and averaging to find explicit expressions for the effective properties in a convenient form. Here we discuss an extension to the in-plane problem: the response of periodic fibre reinforced composites to time harmonic (low frequency) P/SV wave propagation. For simplicity, we assume all fibres are identical and of circular cross-section, and that both host and fibre phases are isotropic; additionally we restrict attention to lattice geometries which result in, at most, orthotropic elastic symmetry. The governing equations are presented and the effective quasi-static material properties are determined via the integral equation methodology. Results are presented for the effective material properties and compared with existing methods. [1] W.J. Parnell and I.D. Abrahams. "A new integral equation approach to elastodynamic homogenization". Submitted to Proc. Roy. Soc. A, 2007.

10:20-10:40 Break

10:40

**4aUWb8. Velocity dispersion analysis of acoustic scattering wave from elastic shells.** Cao Zhengliang (State Key Lab Ocean Acoustics, Hangzhou Appl Acous Res Inst, Rm 1510 No 96 HuaXing Rd, 310012 Hangzhou, China, caozhengliang@yahoo.com), Du Shuanping (State Key Lab Ocean Acoustics, Hangzhou Appl Acous Res Inst, Rm 1510 No 96 HuaXing Rd, 310012 Hangzhou, China, skldusp@tom.com), Zhou Shihong (State Key Lab Ocean Acoustics, Hangzhou Appl Acous Res Inst, Rm 1510 No 96 HuaXing Rd, 310012 Hangzhou, China, shih\_zhou@yahoo.com.cn), Cong

Weihua(State Key Lab Ocean Acoustics, Hangzhou Appl Acous Res Inst, Rm 1510 No 96 HuaXing Rd, 310012 Hangzhou, China, cwhfrj@hzcnc.com)

In recent years the analysis of velocity dispersion of circumferential waves (or surface waves), excited by an incident plane wave to an elastic cylindrical or spherical shell, has theoretically provided valuable insight into the underlying mechanisms of scattering. In the present study, an approach of velocity dispersion analysis on acoustic scattering wave is advanced for the data induced by a spectral bandwidth signal. The method, based on a signal processing technique in time-frequency domain, is examined with simulating waveforms scattered from an elastic spherical and cylindrical shell respectively by linear frequency-modulated signal. In addition, the forward scattering waves of an elastic cylindrical shell filled with fluid is obtained by a line horizontal array in laboratory waveguide, and the results of velocity dispersion is compared with that of theoretical calculation. [Work supported by the National Natural Science Foundation of China (Grant No. 10704068)]

11:00

**4aUWb9. Experimental study of piezoelectret foams as underwater sensors.** Michael Haberman (The University of Texas, Applied Research Laboratories, 10000 Burnet Road, Austin, TX 78758, USA,

haberman@arlut.utexas.edu), Steven Embleton (The University of Texas, Applied Research Laboratories, 10000 Burnet Road, Austin, TX 78758, USA, steve.embleton@arlut.utexas.edu)

This work explores the use of piezoelectret foams (PF) as acoustic sensors for underwater applications. PF material (Emfit Ltd.) provides an intriguing alternative to piezoelectric ceramics and piezopolymers due to their low density, minimal thickness, and potential for easily creating sensors of complex geometries. The foams consist of a continuous polymer containing electrically polarized elliptical voids. Typical density, thickness, and low frequency receive voltage sensitivity (RVS) values of these foams are 300 kg/m<sup>3</sup>, 80  $\mu$ m, and -175 dB re 1V/ $\mu$ Pa, respectively. This work presents experimentally obtained RVS of piezoelectret sensors with rigid and pressure release backing. The results show PF sensor RVS is comparable to conventional sonar transducers at low ambient pressures with RVS degradation as ambient pressure increased. The results are compared with theoretical predictions using microelectromechanical mean field theory and equivalent circuit models. Theoretical predictions explain the observed sensitivity degradation due to void closure caused by the applied pressure. To overcome the performance degradation, two pressure tolerant sensor design concepts are proposed and tested. The designs aim to employ PF elements for high ambient pressure applications while leveraging foam density and thickness to create lightweight, low profile sensors.

THURSDAY MORNING, 3 JULY 2008

ROOM 342A, 8:00 TO 10:40 A.M.

## Session 4aUWc

### Underwater Acoustics and ECUA: Geoacoustic Sediment Modeling III

Nicholas P. Chotiros, Cochair

*Applied Research Laboratories, University of Texas, PO Box 8029, Austin, TX 78713-8029, USA*

Jean-Pierre Sessarego, Cochair

*Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, Marseille, 13009, France*

### Contributed Papers

8:00

**4aUWc1. Measuring grain roughness for the purpose of high-frequency acoustic modeling.** Kevin Briggs (Naval Research Laboratory, Marine Geosciences Division, Stennis Space Center, MS 39529-5004, USA, kbriggs@nrlssc.navy.mil), Allen Reed (Naval Research Laboratory, Marine Geosciences Division, Stennis Space Center, MS 39529-5004, USA, areed@nrlssc.navy.mil), Richard Ray (Naval Research Laboratory, Oceanography Division, Stennis Space Center, MS 39529-5004, USA, Ricky.Ray@nrlssc.navy.mil), Michael Richardson (Naval Research Laboratory, Marine Geosciences Division, Stennis Space Center, MS 39529-5004, USA, Mike.Richardson@nrlssc.navy.mil)

Grain roughness and packing may be important sediment properties for newer acoustic models. We present Scanning Electron Stereomicroscopic imagery of natural sand grains of varying shape and roughness for evaluation of potential grain interactions of individual grains. The grain shape and presence of microasperities will determine the probability of the contact with surrounding grains being a point or a broader area. Moreover, the use of a micro-roughness power spectrum in characterizing the sand grain roughness may be appropriate. Grain contact information is an essential starting point for developing and evaluating acoustic models that address acoustic losses at high frequencies. This information provides the basis to understand contact mechanics, such as grain slip and frame dilation, during insonification. Because media frame stiffness depends, at the grain scale, on the number and type of grain-to-grain contacts, ultimately we would like to characterize sand sediments with imagery identifying such grain contact

information. The natural shapes of grains, their variations, and their packing density present a variety of non-ideal (point) contacts. X-ray micro-focus Computed Tomography shows great promise for documenting of the number and size (area) of grain contacts within grain aggregations, and a demonstration of these data will also be presented.

8:20

**4aUWc2. An in situ sediment sound speed measurement platform: Design, operation and experimental results.** Jie Yang (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, USA, jieyang@apl.washington.edu), Dajun Tang (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, djtang@apl.washington.edu), Kevin L. Williams (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, USA, williams@apl.washington.edu)

A unique Sediment Acoustic-speed Measurement System (SAMS) was developed to directly measure sediment sound speed. The system consists of ten fixed sources and one receiver. In a typical deployment, the SAMS is deployed from a ship that is dynamically positioned. The sources are arranged just above the sea bottom and the receiver is drilled into the sediment with controlled steps by a vibro-core. The maximal sediment penetration depth is 3 meters. At each receiver depth, the 10 sources transmit to the receiver at different angles in the frequency range of 2-35 kHz, providing 10 estimates of sound speed through time-of-flight measurements from the known source-to-receiver geometry. SAMS was deployed three times during

4a THU. AM

the recent Shallow Water Experiment 2006 (SW06) on the New Jersey shelf at 80 m water depth. Preliminary results of sediment sound speed are  $1618 \pm 11$ ,  $1598 \pm 10$ , and  $1600 \pm 20$  m/s at three separate deployment locations. Little dispersion in sediment sound speed was observed. (Work supported by ONR)

8:40

**4aUWc3. Estimating sediment speed and attenuation with sub-bottom reflections.** Kunde Yang (Institute of Acoustic Engineering, Northwestern Polytechnical University, 710072 Xi'an, China, ykdzym@nwpu.edu.cn), Ross Chapman (University of Victoria, 3800 Finnerty Rd, Vcitoria, BC V8W 3P6, Canada, chapman@uvic.ca), Yuanliang Ma (Institute of Acoustic Engineering, Northwestern Polytechnical University, 710072 Xi'an, China, yhma@nwpu.edu.cn)

An inversion method based on sub-bottom reflection is investigated using LFM data collected by a Vertical Linear Array from the SW06 experiment. The distance between the LFM source and the array is about 230m. After extracting the impulse response with matched filter from the received signal, the sub-bottom reflections were found to be strong. The chirp sonar survey nearby the experiment site showed that there is a prominent shallow sub-bottom R reflector with about 20 m depth (based on 1500 m/s). The relative arrival time and amplitude of the sub-bottom reflection signals were used to estimate the sound speed and the attenuation of the sediment. Because the direct arrivals in the water were influenced strongly by the internal wave, the first bottom reflection was applied as the reference path to calculate the relative arrival time and absorption loss in sediment. The sediment attenuation was estimated by assuming that it had linear frequency dependence. The estimated values of the sediment sound speed and attenuation were compared with matched field geoaoustic inversion results published by other research group.

9:00

**4aUWc4. Shear wave speed increases with depth to the one-sixth power in sandy/silty marine sediments.** Allan D. Pierce (College of Engineering, Boston University, 110 Cummings St, Boston, MA 02215, USA, adp@bu.edu), William M. Carey (College of Engineering, Boston University, 110 Cummings St, Boston, MA 02215, USA, wcarey@bu.edu)

Gravity holds the sediment's particles in loose contact; the strength varies with depth. The distribution in shapes, sizes, and orientations is presumed independent of depth. Each particle is subject to several contact forces, and also to a buoyancy force exerted by the surrounding water. An externally imposed shear stress results in distortions in the individual grains, the nature and magnitudes of which depend on the contact areas between the grains, which in turn depend on depth. A derivation making use of fundamental mechanics, the theory of elasticity, and Hertz's theory of contact yields shear modulus  $G$  as a dimensionless quantity times  $g^{1/3}(\rho_s - \rho_w)^{1/3} E^{2/3} d^{1/3}$ , where  $d$  is depth into the sediment and  $E$  is the elastic modulus of the solid material in the grains. The dimensionless quantity depends on Poisson's ratio and porosity. The shear speed  $(G/\rho_w)^{1/2}$  consequently varies with depth as  $d^{1/6}$ . The prediction is consistent with data reported in the past by Stoll, Yamamoto, and Hamilton; the discrepancy of the theoretical prediction of 0.167 with experimentally derived exponents of the order of 0.25, although not viewed as significant, is discussed, and it is suggested that such may be caused by the variation of porosity with depth.

9:20

**4aUWc5. Measuring attenuation and velocity within unconsolidated lacustrine sediments, using high-resolution seismic data.** Luke J. Pinson (National Oceanography Centre, University of Southampton Waterfront Campus, European Way, SO14 3ZH Southampton, UK, ljwp101@soton.ac.uk), Timothy J. Henstock (National Oceanography Centre, University of Southampton Waterfront Campus, European Way, SO14 3ZH Southampton, UK, then@noc.soton.ac.uk), Justin K. Dix (National Oceanography Centre, University of Southampton Waterfront Campus, European Way, SO14 3ZH Southampton, UK, jkd@noc.soton.ac.uk), Jonathan M. Bull (National Oceanography Centre, University of Southampton Waterfront Campus, European Way, SO14 3ZH Southampton, UK, jmb1@noc.soton.ac.uk), Angus I. Best (National Oceanography Centre, University of

Southampton Waterfront Campus, European Way, SO14 3ZH Southampton, UK, aib@noc.soton.ac.uk)

We present estimates of in-situ compressional-wave attenuation and velocity within the uppermost 30 m of unconsolidated lacustrine sediments within Lake Windermere (U.K.), using high-resolution seismic data acquired with Boomer and Chirp sources. The wide frequency bandwidth of the Chirp source allows attenuation to be examined over a frequency range of approximately 2-9 kHz, and by using a spectral ratio technique incorporating robust re-weighted least squares regression, the apparent quality factor of sediments can be accurately and precisely determined. A 60 m multi-channel streamer used with the Boomer source allows interval velocities between target reflectors to be obtained. Models relating quality factor to mean grain-size distinguish between coarse grain-dominated and clay-dominated sediments. The interval velocities improve the classification of the sediment sequences to clays, laminated silts & sands, and coarse sand and gravel deposits. The results are evaluated against geological and core data.

9:40

**4aUWc6. Environmental effects on frequency behavior of modal attenuation coefficients for sandy bottoms.** Wendy Saintval (University of Miami, RSMAS, 4600 Rickenbacker Causeway, Miami, FL 33149, USA, wsaintval@hotmail.com), William M. Carey (College of Engineering, Boston University, 110 Cummings St, Boston, MA 02215, USA, wcarey@bu.edu), Allan D. Pierce (College of Engineering, Boston University, 110 Cummings St, Boston, MA 02215, USA, adp@bu.edu), James F. Lynch (Woods Hole Oceanographic Institution, 98 Water Street, Bigelow 203A, MS-11, Woods Hole, MA 02543, USA, jlynch@whoi.edu), William L. Siegmund (Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180, USA, siegmw@rpi.edu)

The modal attenuation coefficients (MACs) can be determined using a recent simplification of Biot theory [A.D. Pierce *et al.*, J. Acoust. Soc. Am. **114**, 2345 (2003)]. Numerical calculations use sandy bottom sediments and isospeed, linear, and piecewise linear water profiles, which are simplifications that preserve key features of those obtained in experiments off the New Jersey Shelf. The calculations indicate the importance of downward refracting profiles and the strength of near-interface gradients for increasing energy loss. Principal characteristics of the MACs that are observed from the calculations include: increases with interface gradient, reordering of least attenuated modes, and variations of the frequency power-law exponents of the MACs from  $f^1$  to  $f^4$  at frequencies up to 2 kHz. Evidence of the behavior observed in the calculations is in good agreement with previous analysis of results in Gulf of Mexico experiments [F. Ingenito, J. Acoust. Soc. Am. **53**, 858-863 (1973)], for profiles that were classified as weakly downward refracting or nearly isospeed. [Work partially supported by ONR.]

10:00

**4aUWc7. What is the spatial volume involved for wave reflection from flat and curved interfaces?** Paul Cristini (CNRS-UMR5212 Modélisation et Imagerie en Géosciences, UPPA BP115, 64013 Pau, France, paul.cristini@univ-pau.fr), Nathalie Favretto-Cristini (CNRS-UMR5212 Modélisation et Imagerie en Géosciences, UPPA BP115, 64013 Pau, France, nathalie.favretto@univ-pau.fr), Eric De Bazelaire (11, Route du Bourg, 64230 Beyrie-en-Béarn, France, edebaz@wanadoo.fr)

The spatial region in the vicinity of the interface which actually affects the interface response, and hence the reflected wavefield, is of particular interest for the characterization of reflectors. This region represents a volume of integration of properties above and beyond the interface whose maximum lateral extent corresponds to the lateral extent of the Interface Fresnel Zone (IFZ), and whose maximum vertical extent is equal to a thickness we evaluate approximately for subcritical incidence angles and for the case of a plane homogeneous interface. The maximum vertical extent may be greater than the seismic wavelengths for subcritical incidence angles close to the critical angle and for strong impedance contrast at the interface. The whole part of reflector which actually affects the reflected wavefield is then larger than described by previous estimates which considered only the spatial region beyond the interface. In addition to the case of a flat interface, we also discuss the change in the characteristics of this part of the reflector as a function of the interface curvature.

10:20

**4aUWc8. Laboratory sound speed measurements on high water content sediment samples.** Vanessa A. Martin (Laboratoire GeM (et EDF), UMR CNRS 6183 - Institut de Recherche en Génie Civil et Mécanique, IUT de St Nazaire - département Génie Civil - 58 rue Michel Ange, 44600 Saint Nazaire, France, vanessa.martin@univ-nantes.fr), Alain Alexis (Laboratoire GeM (et EDF), UMR CNRS 6183 - Institut de Recherche en Génie Civil et Mécanique, IUT de St Nazaire - département Génie Civil - 58 rue Michel Ange, 44600 Saint Nazaire, France, alain.alexis@univ-nantes.fr), Vincent Martin (Institut Jean Le Rond d'Alembert, UMR CNRS 7190, UPMC, 2 Place de la Gare de Ceinture, 78210 Saint-Cyr l'Ecole, France, vmartin@ccr.jussieu.fr)

Laboratory measurements of sound speed in fluid viscous materials are known to be difficult, especially for frequencies of a few kHz. An

experimental set up which allows such measurements is developed. Tests are run on sediment samples of various lengths (5cm - 20cm), all water-saturated but with different water contents (or densities). When sound speed only depends on the water content over a narrow frequency bandwidth, its estimation originates from time-of-flight measurements on samples of different lengths. It will appear that the water content does play a significant role on the speed of sound. When sound speed depends on frequency (dispersive waves) due to the sediment viscoelastic behaviour, the dependency can be taken into account. An analysis according to the sample length will be given to characterize the sound dispersion for different water contents. It will be shown that both the above studies on the experimental campaign yield sound speed estimations against sample lengths under various hypotheses. In these conditions, the estimation leads to information about saturated sediment behaviour.

THURSDAY MORNING, 3 JULY 2008

ROOM 362/363, 8:00 A.M. TO 1:00 P.M.

### Session 4aUWd

#### Underwater Acoustics and ECUA: Propagation and Reverberation

David L. Bradley, Cochair

*Pennsylvania State University, Post Office Box 30, State College, PA 16804-0030, USA*

Kevin D. Lepage, Cochair

*Naval Research Laboratory, 4555 Overlook Ave SW, Washington, DC 20375, USA*

#### Contributed Papers

8:00

**4aUWd1. Sound focusing and scanning in shallow water with background internal wave field.** Andrey A. Lunkov (Moscow State Technical University n.a. N.E. Bauman, 2nd Baumanskaya 5, 107005 Moscow, Russian Federation, landr2004@mail.ru), Sergey A. Pereselkov (Voronezh State University, 1 Universitetskaya Sq., 394006 Voronezh, Russian Federation, pereselkov@yandex.ru)

Sound field focusing and scanning with focal spot are investigated for shallow water and long distances (up to 30km). It is studied the availability of horizontal and vertical scanning for controlling long-range low-frequency bottom reverberation as well. These researches are carried out by using numerical modeling for typical acoustic waveguides including random ones with background internal waves. The acoustic field is focused with vertical array by phase conjugation of sound wave from probe source placed at preset point. Horizontal scanning with the focal spot is performed by radiation frequency tuning. It is demonstrated that the feasibility for scanning and hence for controlling bottom reverberation takes place for the regular waveguide under winter conditions when sound speed depends on depth only slightly and background internal waves are nonexistent practically. As an example for these conditions we can control the bottom reverberation on interval ~ 5km in the neighborhood of preset point. In summer for near bottom sound waveguide the range of distances for which it is practicable is much narrower even without internal waves. For the waveguide with intense background internal waves the control of long-range bottom reverberation becomes impossible. The work was supported by RFBR Project 05-02-16842.

8:20

**4aUWd2. An optimized source transmission scheme based on pressure sensitivity kernels.** Kaustubha Raghukumar (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, kaustubha@ucsd.edu), Bruce Cornuelle (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA

92093-0238, USA, bdc@ucsd.edu), William Hodgkiss (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, wsh@mpl.ucsd.edu), William A. Kuperman (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, wkuperman@ucsd.edu)

A first-order Born approximation is used to obtain the pressure sensitivity of the received signal to small changes in medium sound speed. The pressure perturbation to the received signal caused by medium sound speed changes is expressed as a linear combination of single-frequency sensitivity kernels weighted by the source signal in the frequency domain. This formulation is used to optimize the pressure sensitivity kernel to give a new source transmission that can produce a focal spot and at the same time, to have less sensitivity to sound speed fluctuations than time-reversal. The formulation allows for a trade-off between quality of focal spot and sensitivity to environmental fluctuations. The optimized new source transmission uses knowledge of the medium statistics and is related to the regularized inverse filter.

8:40

**4aUWd3. Passive phase conjugation processing to forward scattering waves by target in shallow water.** Yoshiaki Tsurugaya (NEC @Corp., 1-10 @Nissin-cho, Fuchu, 183-8501 Tokyo, Japan, y-tsurugaya@bp.jp.nec.com), Toshiaki Kikuchi (National Defence Academy, 39-21 Uhoyabe 4-chome, Yokosuka, 238-0024 Kanagawa, Japan, ADS01881@nifty.com), Koichi Mizutani (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, mizutani@esys.tsukuba.ac.jp)

This paper describes the detection of an underwater target using a passive phase conjugation processing. It is assumed that a sound source and an array are disposed in shallow water, and a target exists between them. The traveling wave from the sound source and the scattered wave from the target are received by the array. However, they are received almost simultaneously. In addition, because the level of the traveling wave is considerably larger than that of the scattered wave, the detection of the scattered wave is

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difficult. Then, the traveling wave components are removed from the signals received in the array. And, the passive phase conjugation processing is given to the signals. The signal to which the passive phase conjugation is processed is similar with the signal radiated from the target. The signal radiated from the target relates to the sound wave that enters at the position of the target from the sound source for the small target. The sound wave that enters at the position of the target is uniquely decided depending on the condition of the propagation environment and the sound source. Therefore, the result that the passive phase conjugation is processed contains the position information of the target.

9:00

**4aUWd4. Efficient time reversal by Lanczos iterations.** Assad A. Oberai (Rensselaer Polytechnic Institute, Mechanical, Aerospace and Nuclear Engineering, 5048 JEC, 110 8th Street, Troy, NY 12180, USA, oberai@rpi.edu), Gonzalo R. Feijoo (Woods Hole Oceanographic Institution, Woods Hole, MA 02543, USA, gfeijoo@whoi.edu), Paul E. Barbone (Boston University, 110 Cummington St, Boston, MA 02215, USA, barbone@bu.edu)

Time reversal methods have seen many applications in underwater acoustics, medical ultrasound, nondestructive evaluation, and several non-acoustics applications. Techniques to target and/or locate individual scatterers are based upon using iterative time-reversal. From a mathematical perspective, iterative time reversal is akin to the power method of extracting eigenvectors of the scattering operator. As such, standard iterative time-reversal inherits the limitations of the power method. For example, iterative time-reversal converges slowly in the presence of multiple scatterers of similar strength and has difficulty identifying weaker scatterers. On the other hand, Lanczos iterations surmount these difficulties, and tend to converge faster than the power method, even when the latter works well. In this contribution we show how Lanczos iterations can be adapted to time-reversal iteration. This allows the strongest eigenvectors of the scattering operator to be measured with greater accuracy and in many fewer iterations than required by standard time-reversal. We describe how this algorithm may be implemented in a practical situation and build a new time-reversal method around it. We apply this method to some numerical examples to demonstrate its effectiveness and compare its performance with the more traditional time-reversal iterations.

9:20

**4aUWd5. Specific features of surface reverberation in shallow water with focused sound field.** Valeriy G. Petnikov (A.M. Prokhorov General Physics Institute, 38 Vavilov str., 119991 Moscow, Russian Federation, petniko@kapella.gpi.ru), Sergey A. Pereselkov (Voronezh State University, 1 Universitetskaya Sq., 394006 Voronezh, Russian Federation, pereselkov@yandex.ru)

The specific features of long-range low-frequency (230 Hz) surface reverberation using vertical radiating array are studied in the framework of numerical experiment. We assume that the array focuses a sound field at different distances from ruffled surface in shallow water. Focusing is carried out by using a time-reversal mirror at a distance 10 - 30 km from radiating array. We consider sound focusing and surface reverberation in the presence of developed wind waves for acoustic waveguide common to shallow water for different seasons. The feasibility of surface reverberation suppression is estimated by sound focusing in the central region of the waveguide including the region located below thermocline. The results of numerical modeling are compared with the similar results obtained before for bottom reverberation. The work was supported by RFBR Project 05-02-16842.

9:40

**4aUWd6. Active sonar discrimination of water-column objects from bottom clutter using waveguide invariant modeling of distributed backscatter.** Jeffrey Krolik (Duke University, ECE Dept. Hudson Hall 130, PO Box 90291, Durham, NC 27708, USA, jk@ee.duke.edu), Ryan Goldhahn (Duke University, ECE Dept. Hudson Hall 130, PO Box 90291, Durham, NC 27708, USA, rag15@ee.duke.edu), Granger Hickman (Duke University, ECE Dept. Hudson Hall 130, PO Box 90291, Durham, NC 27708, USA, gwh@ee.duke.edu)

The ability to differentiate water-column returns from clutter discretizes due to backscatter from bottom features is critical to improving the detection performance in active sonar. Previous approaches include the use of statistical feature-based classifiers and physics-based depth estimation. Such methods are typically challenged by the convolutive effect of both uncertain multipath propagation and complex scattering characteristics in sonar returns. In previous work, we used waveguide invariant theory to model frequency-selective fading observed in distributed backscatter from the ocean bottom. In particular, the short-time energy spectrum of backscatter from the sea bottom at a given range can be well estimated along striations in time-frequency energy predicted by waveguide invariant theory. In this paper, we present a statistic which tests the observed frequency-selective fading of the short-time energy spectrum for a range cell under test against the spectrum predicted by waveguide invariant theory for a bottom feature using neighboring ranges and frequency bins. Results are presented for discrimination of a water column target versus bottom clutter using data simulated via a normal mode propagation model with environmental parameters from the Malta Plateau. Receiver operating characteristic analysis is performed showing good discrimination of equal energy targets from clutter. [Work supported by ONR].

10:00

**4aUWd7. Using a dispersive source signal to generate a dispersive field in a nondispersive medium.** Shane C. Walker (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, shane@mpl.ucsd.edu), William A. Kuperman (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, wkuperman@ucsd.edu)

In typical treatments, dispersion is considered to be a property of the medium. For example, a free space, static, homogeneous medium is dispersionless while gravity waves, particularly in the case of surface waves in a deep body of water, exhibit dispersion. Here it is shown that dispersion in the field can also result from dispersion in the source signal. As a demonstration, a dispersive source signal is shown to introduce dispersion in a dispersionless medium. Consequently, through proper design of the source signal, it is possible to tailor the resulting field dispersion to suit a variety of applications. Here potential applications to imaging in complex media are discussed. Simulation and experimental results are presented.

10:20-10:40 Break

10:40

**4aUWd8. Equivalence of the waveguide invariant and two path ray theory methods for range prediction based on Lloyd's mirror patterns.** Daphne Kapolka (Naval Postgraduate School, 13693 Tierra Spur, Salinas, CA 93908, USA, dkapolka@nps.edu)

For shallow, range-independent environments where the sound is dominated by low-order modes, the constant which characterizes the modal interference pattern, the waveguide invariant, is approximately equal to one. The speed of a contact which maintains a constant course and speed as it passes through its closest point of approach (CPA) can be determined from the asymptotic behavior of its tonal frequencies from the Doppler shift. This information can be used along with the change in broadband striation frequencies in a Lloyd's mirror pattern over time to extract the range of the contact as it transits through CPA. If instead of using normal mode theory, the Lloyd's mirror pattern is derived as the coherent interference between a straight-line direct and surface-reflected path, a relationship between the striation frequencies and time of a crossing contact can also be derived. This relationship can be shown to be identical to the result obtained from the nor-

mal mode approach when the value of the waveguide invariant is equal to one. Thus, the same equations can be used to extract range information from a Lloyd's mirror pattern regardless of contact range as long as the environmental conditions support a waveguide invariant close to one.

11:00

**4aUWd9. Analytical Time Domain Analysis of an Acoustic Waveguide.** Hüseyin Ö. Sertlek (The Scientific and Technological Research Council of Turkey, Marmara Research Center, Information Technologies Institute, Gebze, 41400 Kocaeli, Turkey, ozkan.sertlek@bte.mam.gov.tr), Serkan Aksoy (Gebze Institute of Technology, Electronics Engineering Department, Çayırova, Gebze, 41400 Kocaeli, Turkey, saksoy@gyte.edu.tr)

A new analytical time domain Normal Mode solution for one layer acoustic channel in two dimensional Cartesian coordinates is presented in this paper. The method is based on the separation of variables technique for the time domain wave equation. Dirichlet boundary condition is applied on the upper and lower boundaries of the channel. Klein-Gordon equation is obtained and solved by using the Green function. The fundamental advantage of this new technique is not necessary to use the Fourier Transformation to obtain time domain response of arbitrary acoustical source signal in the waveguide. In order to validate the obtained results, first of all, the Fast Fourier Transform is applied to the time domain analytical data, then, the comparisons are given between KRAKEN program and a new analytical time domain Normal Mode solution in the sense of Transmission Loss relating to ranges. The excellent agreement is obtained between the results.

11:20

**4aUWd10. Application of the Mode Matching Technique to Fluid - Solid Layered Acoustic Problems.** Hüseyin Ö. Sertlek (The Scientific and Technological Research Council of Turkey, Marmara Research Center, Information Technologies Institute, Gebze, 41400 Kocaeli, Turkey, ozkan.sertlek@bte.mam.gov.tr), Serkan Aksoy (Gebze Institute of Technology, Electronics Engineering Department, Çayırova, Gebze, 41400 Kocaeli, Turkey, saksoy@gyte.edu.tr), Deniz Bolukbas (The Scientific and Technological Research Council of Turkey, Marmara Research Center, Information Technologies Institute, Gebze, 41400 Kocaeli, Turkey, deniz.bolukbas@bte.mam.gov.tr), Sevgi Akgün (The Scientific and Technological Research Council of Turkey, Marmara Research Center, Information Technologies Institute, Gebze, 41400 Kocaeli, Turkey, sevgi.akgun@bte.mam.gov.tr)

An application of Mode Matching technique for analyzing interactions between range dependent fluid-solid layers in two dimensional cylindrical coordinates is presented by using Normal Mode method. The transmission loss for fluid-solid medium with semi-infinite bottom layer is calculated for range-dependent bottom profiles. The bottom solid half space is divided independent vertical regions due to different bottom parameters for simulation of the real acoustic environment. The transition between the independent vertical regions is ensured by using Mode Matching technique with Normal Mode method. The attenuation of acoustic wave modes in a range dependent two layered fluid-solid medium due to absorption of the lower semi-infinite solid layer is calculated. In order to validate the obtained results in the sense of Transmission Losses, Parabolic Equation program (RAM) is used for different transition cases. The good agreement is found between the results which can be used for realistic Sonar simulators.

11:40

**4aUWd11. The comparative analysis of two solutions of Pekeris boundary problem.** Nadezhda Zlobina (Institute of Marine Technology Problems, Sukhanov str., 5a, 690950 Vladivostok, Russian Federation, zlobina@marine.febras.ru), Boris Kasatkin (Institute of Marine Technology Problems, Sukhanov str., 5a, 690950 Vladivostok, Russian Federation, bigcezar@mail.ru)

The solution of the reduced Pekeris boundary problem, satisfying to the generalized radiation condition on an impedance interface is obtained. Mode part of the solution is presented by series expansion on the total system of regular normal waves, of generalized normal waves and of leakage normal

waves. The principle of the continuous continuation of the obtained solution in physical half-space with validity of boundedness condition is formulated. Expansion of the solution on physical half-space in a class of generalized functions is constructed. The obtained generalized solution includes canonical series expansion on the set of eigenfunctions of both conjugate operators corresponding an initial not self-conjugate boundary problem, the associated eigenfunctions of a waveguide with two soft boundaries, the associated eigenfunctions of a waveguide with upper soft boundary and lower rigid boundary and a set of spherical components. Results of the comparative analysis of the classical solution and generalized solution are given.

12:00

**4aUWd12. Parabolic Wavelet Technique to Numerical Modelling of Shallow Water Acoustic.** Mostafa Bakhoday Paskyabi (Research Engineering Institute, 16th Kilometers of Old Road Of Tehran-Karaj, Control Engineering Group, Agricultural Engineering Institute, 13445-754 Tehran, Iran, bakhoda@guilan.ac.ir), Seyed Taha Mortazavi (Research Engineering Institute, 16th Kilometers of Old Road Of Tehran-Karaj, Control Engineering Group, Agricultural Engineering Institute, 13445-754 Tehran, Iran, t\_mortazavi@yahoo.com)

In this paper, parabolic equation (PE) model is applied for environment of slowly varying with range and azimuth in a region of Persian Gulf. This method is currently the most used to study propagation in non-stratified media. Due to required accuracy for a gridding step comparable with the wavelength, the computation load is very heavy. To overcome this disadvantage, we develop the weak form of PE formulation by the use of Deslaurier-Daubuc interpolating wavelet (DDW) basis functions. The formulation is similar to combination of conventional Finite Element Method (FEM) and split step technique except that, here DDW basis is used for approximating in depth to reduce the cost of computations and to increase the accuracy of method by the use of multi-resolution structure of wavelets. The localized and circulant representations of depth differential operators based on DDW connection coefficients allow efficient imposition of boundary values and circumvent some disadvantages of the traditional PE. Furthermore, details of bottom boundary imposing and some disadvantages of approach are presented. Numerical experiments are performed to study two layers water-bottom environment with none-smooth sediment interface to some canonical standard test problems in comparison with solutions obtained by conventional PE method.

12:20

**4aUWd13. Studying converted waves in shallow marine environment.** Nihed El Allouche (Delft University of Technology, P.O. Box Postbus 5048, 2600 GA Delft, Netherlands, n.elallouche@tudelft.nl), Guy G. Drijkoningen (Delft University of Technology, P.O. Box Postbus 5048, 2600 GA Delft, Netherlands, g.g.drijkoningen@tudelft.nl), Willem Versteeg (Ghent University, Krijgslaan 281, S8, 9000 Ghent, Belgium, willem.versteeg@ugent.be), Dick G. Simons (Delft University of Technology, P.O. Box Postbus 5048, 2600 GA Delft, Netherlands, d.g.simons@tudelft.nl)

For many years, waves converted from compressional to shear mode have been successfully applied in hydrocarbon exploration to image and characterize the subsurface. Since shear-waves propagate with a velocity that is dependent on the shear modulus and are thus directly related to the shear strength of the sediment, they are very useful for geotechnical purposes. Generally, P-wave reflection amplitudes contain S-wave information but Riedel et al. (2001) showed that these reflections are not very sensitive to this. An alternative approach is to obtain S-wave information directly from converted waves. However, it is not clear whether these waves can be applied for geotechnical aims. The main focus will be on understanding the dependence of mode conversion on the seismic properties. In this study, we investigate the possibility of acquiring converted waves in marine unconsolidated sediments. From our numerical experiments we found that the conversion is maximal at two angles where the smallest angle appears to be more favorable in the environment of interest. Furthermore, we show how and where converted waves can be observed on a seismogram and the optimum field configuration to acquire them.

12:40

**4aUWd14. Conservative initial-boundary value problems for the Wide-Angle PE in waveguides with variable bottoms.** Dimitra Antonopoulou (Department of Applied Mathematics, University of Crete, 71409 Heraklion, Greece, danton@tem.uoc.gr), Vassilios Dougalis (Mathematics Department, University of Athens, 15784 Zografou, Greece, doug@math.uoa.gr), 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), George Zouraris (Department of Applied Mathematics, University of Crete, 71409 Heraklion, Greece, zouraris@math.uoc.gr)

We consider the third-order, Claerbout-type Wide-Angle Parabolic Equation (PE) in the context of Underwater Acoustics in a cylindrically symmetric medium consisting of water over a soft bottom  $B$  of range-dependent topography. There are strong indications, that the initial-boundary value

problem for this equation with just a homogeneous Dirichlet boundary condition on  $B$ , may not be well-posed, for example when  $B$  is downsloping. In previous work we proposed an additional bottom boundary condition that, together with the zero field condition on  $B$ , yields a well-posed problem. In the present paper we continue our investigation of additional bottom boundary conditions that yield well-posed, physically correct problems. Motivated by the fact that the solution of the wide-angle PE in a domain with horizontal layers conserves its  $L_2$  norm in the absence of attenuation, we seek additional boundary conditions on a variable-topography bottom, that yield  $L_2$ -conservative solutions of the problem. We identify a family of such boundary conditions after a range-dependent change of the depth variable that makes  $B$  horizontal. We discretize the continuous problems by second-order accurate Crank-Nicolson type finite difference schemes, and show, by means of numerical experiments, that the new models yield accurate simulations of the acoustic field in standard, wedge-type domains with upsloping and downsloping bottoms.

THURSDAY MORNING, 3 JULY 2008

ROOM 341, 11:40 A.M. TO 7:40 P.M.

### Session 4aUWe

#### Underwater Acoustics and ECUA: Tank Experiments I

Raymond J. Soukup, Cochair

*U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA*

Jean-Pierre Sessarego, Cochair

*Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, Marseille, 13009, France*

#### Invited Papers

11:40

**4aUWe1. Tank experiment: The ultrasonic approach to ocean physics.** 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)

Acoustic/elastic waves ranging from a few kHz to a few MHz are nowadays easy to emit/receive through the use of "key in hands" multi-channel systems. These systems provide the instantaneous amplitude and phase of the deterministic wave propagating in the medium with a dynamic larger than 90 dB in some cases (16-bit amplitude sampling). The use of a large number of channels (at least 64) is mandatory to simultaneously investigate the spatial and temporal aspect of wave propagation in complex media. The advantage of a laboratory-scaled ocean model relies in the ease with which it can be build, modified, and/or controlled over time. Finally, the trend in ocean wave physics is now to study the dynamics of such medium (internal waves, turbulent flow, sensor motion) that requires the use of a real-time acquisition system. Examples of the types of analog experiments that could be carried out with a laboratory-scaled system are the study of nonuniform doppler shifted fields in reverberant environments, ocean acoustic tomography and similar inverse problems, coherent communications in complex moving environments, etc. After the description of the equipment, we give more specific examples of potential experiments to illustrate the versatility of the laboratory set-up.

12:00

**4aUWe2. Tank experiments of sound propagation over a tilted bottom: Comparison with a 3-D PE model.** Alexios Korakas (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, alexis.korakas@ec-lyon.fr), 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), Jean-Pierre Sessarego (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, sessarego@lma.cnrs-mrs.fr), Didier Ferrand (Laboratoire de Mécanique et d'Acoustique (UPR CNRS 7051), 31, chemin Joseph Aiguier, 13402 Marseille Cedex 20, France, ferrand@lma.cnrs-mrs.fr)

We present results of tank experiments of long-range acoustic propagation over a wedge-shaped oceanic bottom. Previous work investigated the propagation of broadband pulses in a range-independent configuration showing good agreement between experimental results and model predictions [Korakas et al., Proceedings of UAM 2007 (Heraklion, Crete, Greece, 25-29 June 2007)]. As a follow-up, preliminary experiments were carried out considering a tilted bottom. Non-negligible 3-D effects were observed and proved to compare favorably with numerical predictions obtained running a parabolic equation based code [Sturm et al., Proceedings of UAM 2007]. In the

present work we continue the investigation of the wedge-like environment. Additional series of experimental measurements are performed. Received signals are recorded on a very fine spatial grid. Then, the measured data are processed using matched field processing. In our inversion algorithm, the replica are provided by a fully three-dimensional parabolic equation code. The technical aspects of the inversion procedure used (e.g., CPU time) are discussed.

12:20

**4aUWe3. Laboratory investigations of the detection and characterization of buried targets by iterative, single-channel time reversal.** Zachary J. Waters (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, zjwaters@bu.edu), Benjamin R. Dzikowicz (Naval Surface Warfare Center, Panama City Division, Code HS-11, 100 Vernon Ave., Panama City, FL 32407, USA, benjamin.dzikowicz@navy.mil), R. Glynn Holt (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, rgholt@bu.edu), Ronald A. Roy (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, ronroy@bu.edu)

Due to the dynamic nature of the shallow water environment, targets are often buried beneath the seafloor, hindering their detection and identification by acoustic methods. Using iterative time reversal with a single channel transducer [Waters et al., J. Acoust. Soc. Am. 122, 3023 (2007)], the monostatic return from a buried resonant target is enhanced, yielding convergence to a narrowband waveform characteristic of the dominant mode in the target's scattering response. Scaled laboratory experiments are performed with a broadband ( $Q^2$ ) transducer operating in the 0.5-2 MHz frequency range. Solid and evacuated metallic spheres used as targets are buried beneath a layer of simulated sediment. Images generated by scanning the transducer laterally in two dimensions over an area of sediment containing multiple targets show enhancement of different modes in a single target's scattering response and test the selectivity between targets of differing type. Experiments with the transducer positioned at normal and non-normal incidence, quantify the enhancement in the signal-to-noise ratio of target returns as a function of window size and position. [Work supported by The Office of Naval Research and the Center for Subsurface Sensing and Imaging Systems (NSF ERC Award No. EEC-9986821).]

12:40-2:00 Lunch Break

### *Invited Papers*

2:00

**4aUWe4. Detection and classification of a cylindrical target partially or completely buried in thin sand/water mixture by time-frequency representation.** Gerard Maze (LAUE, Université du Havre, Place Robert Schuman, F-76610 Le Havre, France, gerard.maze@univ-lehavre.fr), Dominique Decultot (LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, dominique.decultot@univ-lehavre.fr), Katia Cacheleux (LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, katia.cacheleux@univ-lehavre.fr), Romain Liétard (LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, romain.lietard@univ-lehavre.fr)

Numerous papers show that it is possible to characterize an air-filled cylindrical shell immersed in water from a time-frequency representation. The time scattered signal is constituted by echoes related to the reradiation of waves circumnavigating around the cylindrical target. For particular frequencies, these echoes make resonances which characterize this target. In first part, the scattered signal is calculated, the cylindrical shell is in water or in a medium with characteristics similar to the mixture sand/water used for the experimental measurements. In second part, the cylindrical shell is partially or completely buried in a thin sand/water mixture. It is insonified in normal or oblique incidence relatively to the interface water/mixture. The impulse time scattered signal is plotted and it is treated with a time-frequency representation. The comparison between the theoretical and experimental time-frequency representations shows us that it is possible to detect and classify the object buried in thin sand/water mixture.

2:20

**4aUWe5. Acoustic radiation of low frequency flexural vibration modes in a submerged plate.** Dominique Decultot (LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, dominique.decultot@univ-lehavre.fr), Romain Liétard (LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, romain.lietard@univ-lehavre.fr), Farid Chati (LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, farid.chati@univ-lehavre.fr), Gerard Maze (LAUE, Université du Havre, Place Robert Schuman, F-76610 Le Havre, France, gerard.maze@univ-lehavre.fr), Aleksander Klauson (Tallinn University of Technology, Dept. of Mechanics, Ehitajate tee 5, 19086 Tallinn, Estonia, aklauson@staff.ttu.ee)

In some submarine structures, the acoustic radiation of vibrations at the neighbourhood of sonar equipment limits their use. So the aim of this study is to understand the process of acoustic radiation of a submerged plate subjected to a vibration. For the low frequency domain, only two types of wave can propagate: the first antisymmetric Lamb wave ( $A_0$ ) and the first symmetric Lamb wave ( $S_0$ ). When the plate is immersed in water the  $A_0$  wave is modified and the new wave is named A wave. In this work, experimental and numerical analysis of vibration modes in a plate are carried out. The studied rectangular plate of thickness 10 mm is made of steel. Its length and width are respectively 1.0 m and 0.5 m. The plate is partially immersed in water (90%). Flexural vibrations are generated by a shaker normally connected to the emerging plate part. The applied signal is one sinusoidal period at a frequency which is under the critical frequency. Relations between the admittance of the plate and the radiated pressure in water are highlighted.



**4aUWe6. Inverse problems in sound radiation of complex structures from measurements in a large acoustic tank.** Earl G. Williams (Naval Research Laboratory, 4555 Overlook Ave, Washington, DC 20375, USA, earl.williams@nrl.navy.mil), Brian H. Houston (Naval Research Laboratory, 4555 Overlook Ave, Washington, DC 20375, USA, houston@code7136.nrl.navy.mil), Nicolas Valdivia (Naval Research Laboratory, 4555 Overlook Ave, Washington, DC 20375, USA, valdivia@pa.nrl.navy.mil), Peter C. Herdic (SFA Inc., Suite 405, Crofton, MD 21114, USA, herdic@code7136.nrl.navy.mil)

The Laboratory for Structural Acoustics (LSA) at NRL consists of an indoor cylindrical tank (17 m dia. by 15 m deep) filled with ~ 1 million gallons of deionized water. Key features include: 1) vibration isolation, 2) active temperature control, and 3) anechoic materials. This unique laboratory is instrumented with sophisticated mechanical, electronic and optical systems, that include large work-space in-water robotic scanners to generate nearfield acoustical holography (NAH) databases. We discuss such a database consisting of the underwater near-field pressure measured on a two-dimensional surface conformal to an internally driven complex structure floating at the air-water interface. Various inverse approaches are discussed to image the normal velocity and intensity of the structure at its interface to the fluid, as well as the total power radiated, revealing mechanisms of radiation related to the internal structure. These inverse approaches consist of the equivalent source method compared with the well established Fourier acoustics methods of NAH. This work was supported by the US Office of Naval Research.

**4aUWe7. Broadband elastic scattering by fiberglass spherical shells and plates measured in a water tank: Acoustic inversion and wave analysis.** Alessandra Tesei (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, tesei@nurc.nato.int), Paul D. Fox (National Physical Laboratory, Hampton Road, TW11 0LW Teddington, UK, pf@isvr.soton.ac.uk), Gary B. Robb (National Oceanography Centre, University of Southampton Waterfront Campus, European Way, SO14 3ZH Southampton, UK, gbor199@noc.soton.ac.uk)

Spherical shells and plane plates made of different types of fiberglass (either random or textured) were measured in the backscatter direction, suspended in a water tank in a broadband frequency range between 30 and 350 kHz. The range of  $ka$  for the spheres was approximately 8 to 90, with the  $fd$  range for the plates approximately 0.15 to 1.75 MHz\*m. The aim of the study was to investigate the effects of the fiber type on the object signature, as the frequency and the type of fiber layers vary. Inversion of the material parameters was conducted on the basis of the object's temporal echo. In particular, the estimate of material loss is crucial to determine at what frequency elasticity becomes irrelevant to the object's global response. The spherical shells were measured either void or filled with different materials (liquid and solid) in order to evaluate the contribution of the shell-borne elastic waves with respect to sound scattered from the interior of the object. Elastic wave analysis and analytical modeling tools were used to support the physical interpretation of the measured responses from the different objects.

**4aUWe8. Modal analysis of resonances of an elliptic elastic cylinder immersed in water.** Fernand Leon (LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, fernand.leon@univ-lehavre.fr), Farid Chati (LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, farid.chati@univ-lehavre.fr), 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)

Considerable work has been done on the scattering by cylindrical objects having a circular cross section. The modal formalism based on the theory of elasticity has been developed for studying the acoustic scattering by these elastic cylinders immersed in water. In particular, it has been demonstrated, in normal incidence, that the eigenfrequencies of a circular cylinder can be determined from a resonance spectrum. These eigenfrequencies correspond to circumferential waves that form a resonance when a phase matching along a closed path is obtained. Each eigenfrequency is characterized by a given mode  $n$ , i.e., the number of wavelengths spanning the circumference. Comparatively, little attention has been devoted to the more general case of the noncircular cylindrical cylinders such as the elliptical elastic cylinders. For these objects, we have been developed a modal formalism based on the theory of elasticity. From the results obtained theoretically and experimentally, we show how to obtain a resonance spectrum, independently of the azimuth incident angle and of the radii ratio (minor radius and major radius) so that the eigenfrequencies can be determined. We present also a modal analysis of resonances as function of the azimuth incident angle and of the radii ratio.

### 3:40-5:20 Posters

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

5:20

**4aUWe9. Estimation of some geoacoustic parameters of a tank experiment by match field processing.** Panagiotis Papadakis (IACM-FORTH, N. Plastira 100, 70 013 Vassilika Vouton, Greece, panos@iacm.forth.gr), Michael Taroudakis (University of Crete & FORTH/IACM, Vassilika Vouton, P.O.Box 1385, 711 10 Heraklion, Greece, taroud@iacm.forth.gr), Jean-Pierre Sessarego (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, sessarego@lma.cnrs-mrs.fr), Patrick Sanchez (Laboratoire de Mécanique et d'Acoustique (UPR CNRS 7051), 31, chemin Joseph Aiguier, 13402 Marseille Cedex 20, France, patrick.sanchez@eoures.com)

The work deals with an inversion procedure for the estimation of geometrical and environmental parameters of a simulated shallow water environment. Two of the most significant parameters in a shallow water environment affecting the acoustic field in long range acoustic propagation are the sound speed of the sediment and the depth of the water column. The accurate estimation of these parameters is perhaps the most important task in scaled laboratory experiments. Therefore, the calibration phase of such an experiment involves the estimation of these parameters, preferably using acoustical techniques, regardless the final scope of the experiment. The present work describes an inversion procedure based on standard match field processing. The acoustic signal used was a short gaussian shaped pulse and it was recorded at a certain distance from the source and for several depths in the water column. The spectrum of the recorded signal was obtained using FFT and the complex acoustic field at the central frequency was calculated. The estimated values of the water depth and the sound speed in the sand which was used to simulate the sea-bed, were found in accordance with the observed modal structure of the acoustic field.

5:40

**4aUWe10. High frequency propagation in and scattering from water-saturated granular sediments: Laboratory study.** Anatoliy N. Ivakin (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, USA, ivakin@apl.washington.edu), Jean-Pierre Sessarego (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, sessarego@lma.cnrs-mrs.fr)

Acoustic properties of water-saturated granular sediments at frequencies from 150 kHz to 8 MHz were studied in controlled laboratory conditions using broadband transducers. Two samples of medium sand sediments, with the same mean grain size, but with different width of the size distribution, were taken for the study, degassed, and their surface was flattened. Another sample of sediments was composed of glass beads of the same grain size. The main difference of glass beads from sand grains was their shape. Backscattering strength at normal and oblique incidence and reflection coefficient at normal incidence were measured for the three samples. The reflection experiments were made for different thicknesses of the samples, so that reflections from both first and second interfaces of the sediment layer were measured. This allowed also estimating sound speed and attenuation in the sediments. The results obtained for the three chosen types of the sediments were compared to demonstrate effects of the grain size distribution width and the grain shape on acoustic properties of the sediment. [Work supported by ONR and CNRS].

6:00

**4aUWe11. A time domain model of scattering from small discrete volume particles: Tank validation.** Gaetano Canepa (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, canepa@nurc.nato.int), Jean-Pierre Sessarego (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, sessarego@lma.cnrs-mrs.fr), Alessandra Tesei (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, tesei@nurc.nato.int), Régine Guillermin (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, guillermin@lma.cnrs-mrs.fr), Raymond J. Soukup (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, soukup@abyss.nrl.navy.mil)

A model of the time domain scattering from small discrete volume particles is presented here with an experimental validation. The model was implemented on the backbone of the BORIS-3D model which originally included only surface scattering and volume scattering from small perturbation of the volume density and sound speed. The proposed model adds discrete volume scattering and simulates both monostatic and bistatic configurations. The experimental validation was performed in a tank of the CMRS/LMA Laboratory (Laboratoire de Mécanique et d'Acoustique) using an in-house produced silicon plate (with a flat upper interface) in which 10% of the volume embeds spherical glass beads 1 mm in diameter. The characteristics of this plate was well known; in particular, the amplitude of the volume backscattering (as a function of the scattering angle) has already been presented. In this work, we are particularly interested in the time domain evolution of the scattered echo. The results of this study show a very good agreement between simulated and experimental data in both amplitude and time evolution shape.

6:20

**4aUWe12. Backscattering From Scale Models of Elastic Ocean Bottoms with Power-law Roughness.** Raymond J. Soukup (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, soukup@abyss.nrl.navy.mil), Edward L. Kunz (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, kunz@abyss.nrl.navy.mil), Gaetano Canepa (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, canepa@nurc.nato.int), Harry J. Simpson (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, harry.simpson@nrl.navy.mil), 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)

Rough surface analogs to ocean bottom interfaces with two contrasting power-law roughness spectra were fabricated from slabs of PVC. A tank experiment with a nearly monostatic geometry and transmitted signals in the band 100-300 kHz was performed, with the source and receiver positioned to produce acoustic interactions on multiple locations on the surfaces. Models of scattering strength and

the received time series, using deterministic and stochastic representations of the surface, are used to verify the predicted dependence of the scattering, with emphasis on the sub-critical angle ( $< 50$  degrees grazing) region, where the predicted difference in scattering strength due to the contrast in roughness between the two fabricated surfaces is on the order of 10 dB. The numerical models employed were perturbation theory, a second-order small-slope calculation using the power-law roughness parameters, and a fourth-order small-slope calculation using the actual grid of heights as an input. The experimental effort also focused on the near-critical angle region where the predictions of the numerical models differed markedly. [Work supported by ONR and NURC.]

6:40

**4aUWe13. Shallow-water tank experiments and model comparisons over range-dependent elastic bottoms.** Jon M. Collis (Boston University, Department of Aerospace and Mechanical Engineering, 110 Cummington Street, Boston, MA 02215, USA, jcollis@bu.edu), 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), Harry J. Simpson (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, harry.simpson@nrl.navy.mil), Raymond J. Soukup (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, soukup@abyss.nrl.navy.mil), William L. Siegmund (Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180, USA, siegmw@rpi.edu)

A series of tank experiments has been conducted in order to obtain high quality data for acoustic propagation in shallow-water environments with elastic bottoms. Such problems can now be solved accurately with the parabolic equation method, which is being used for comparisons with measured transmission loss. Results from the initial experiment with a flat or sloped slab of PVC demonstrated both benchmark quality agreement between computed solutions and data, and the necessity of accounting for elasticity in the bottom [J. M. Collis, *et al.*, JASA **122**]. This paper will present results of a second experiment, conducted in the same manner as the first, but modified to allow for variable bottom slopes. Time series were collected at 100-300 kHz on horizontal arrays for two source positions. Parabolic equation solutions for treating variable slopes, using coordinate transformation methods of mapping and axis rotations, will be benchmarked against the new data. [Work supported by the Office of Naval Research.]

7:00

**4aUWe14. The Broadband In-Water Structural Acoustics of Unexploded Ordinance.** Harry J. Simpson (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, harry.simpson@nrl.navy.mil), Brian H. Houston (Naval Research Laboratory, 4555 Overlook Ave, Washington, DC 20375, USA, houston@code7136.nrl.navy.mil), Michael L. Saniga (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, saniga@code7136.nrl.navy.mil), Joseph A. Bucaro (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, joseph.bucaro@nrl.navy.mil)

The Laboratory for Structural Acoustics (LSA) at the Naval Research Laboratory consists of a 1 million gallon, deionized water, indoor cylindrical tank (17m diameter by 15m deep). The key features include: 1) vibration and temperature isolation, 2) feedback controlled heating and adiabatic materials (temperature variability  $< 0.01^\circ\text{C}$ ), and 3) reverberation reducing anechoic materials. This laboratory has computer controlled robotic scanners and manipulators used for precision freefield measurements including nearfield acoustic holography and compact range scattering. The precision robotics, environmental control, and painstaking measures to insure homogeneity and stability result in a high fidelity, versatile, and unique underwater acoustic measurement laboratory. The LSA also contains an indoor rectangular tank (10m by 8m) laboratory, with a 3m deep sand bottom and 4m of water column. In a similar fashion to the freefield laboratory, this laboratory is used to study target scattering in a marine bottom environment. We discuss such databases focused on the challenging problem of unexploded ordinance (UXO) in water where we use the structural acoustic response measured in a series of laboratory experiments to detect and identify several common UXOs. The 360 degree broadband (1-140 kHz) compact range monostatic and bistatic measurements taken in both laboratories will be discussed.

7:20

**4aUWe15. Benchmarking of computational scattering models using underwater acoustic data from a corrugated wax slab.** 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), Gaetano Canepa (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, canepa@nurc.nato.int), Raymond J. Soukup (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, soukup@abyss.nrl.navy.mil), Edward L. Kunz (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, kunz@abyss.nrl.navy.mil), Jean-Pierre Sessarego (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, sessarego@lma.cnrs-mrs.fr), Kevin Rudd (U.S. Naval Res. Lab., Acoust. Div., Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375, USA, kevin.rudd@nrl.navy.mil)

Measured time series for underwater acoustic scattering from a 30 cm x 30 cm x 5 cm wax slab with a two-dimensional corrugated (rippled) surface are compared with simulation results. The experimental geometry and directionality of the sensors allowed for ensonification of the rippled surface and the appearance of shadowing effects at low grazing angles. The acoustic source transmitted impulses at 200-800 kHz (wavelengths between 0.75-0.19 cm). The height and spacing of the ripples were 0.3 cm and 3 cm, respectively, and the slab had negligible shear speed and a measured attenuation. We simulate the experiment with the following methods listed in increasing levels of physical accuracy and computational cost: Kirchhoff Approximation (KA), Second-order Small-Slope approximation (SSA2), the Wide-angle On-Surface Radiation Condition method (WOSRC), a Pseudo-differential Impedance Operator method (PIO), a 2-domain Integral Equation method (IE-2DOM), and an Elastodynamic Finite Integration Technique (EFIT). The range of techniques allowed us to examine effects such as reflections off the interior bottom or ends of the slab and the effectiveness of the asymptotic (KA/SSA2) and pseudo-differential (WOSRC/PIO) methods for cases that include shadowing. (Work sponsored by ONR and NURC.)

## Session 4pAAa

## Architectural Acoustics: Archeological Acoustics I

David Lubman, Cochair

*DL Acoustics, 14301 Middletown Lane, Westminster, CA 92683, USA*

Jens Holger Rindel, Cochair

*Odeon A/S, Scion DTU, Diplomvej Building 381, Lyngby, DK-2800, Denmark*

## Invited Papers

1:20

**4pAAa1. Sound resonance in prehistoric times: A study of Paleolithic painted caves and rocks.** Iegor Reznikoff (Université de Paris X, Département de Philosophie, 92001 Nanterre, France, dominiqueleconte@yahoo.fr)

Caves have natural properties of resonance: some parts sound very well, the sound lasts for some seconds or gives several echoes, some other parts have a dull resonance or no resonance at all. It is extremely interesting to compare in a given cave the map of the most resonant locations with the map of the locations of the paintings: are there correlations between resonance and paintings? We have studied many Paleolithic caves in France in which the answer was remarkably positive; stated shortly: the more resonant the location, the more paintings or signs are situated in this location. Here are presented some studies and results in the caves of Isturitz and Oxcelhaya in Pays Basque and in some other caves. Some considerations are given about the resonance - pictures relationship in open spaces with prehistoric painted rocks. Bibliography I. Reznikoff: Prehistoric Paintings, Sound and Rocks in *Studien zur Musikarchäologie III: 2nd International Symposium on Music Archaeology*, Sept. 2000, ed. E. Hickmann, Berlin, Rahden, 2002, 39-56. The Evidence of the Use of Sound Resonance from Palaeolithic to Medieval Times, *Archaeoacoustics*, C. Scarre & G. Lawson ed., University of Cambridge, Cambridge, 2006, 77-84. On Primitive Elements of Musical Meaning, *www.musicandmeaning.net*, JMM 3 (Invited papers), 2005.

1:40

**4pAAa2. Architectural and acoustic restoration of the 'Benevento' Roman Theatre.** 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), Leda De Gregorio (Built Environment Control Laboratory Ri.A.S., Second University of Naples, Abazia di S. Lorenzo, 81031 Aversa, Italy, ledadegreg@libero.it), Umberto Palmieri (Built Environment Control Laboratory Ri.A.S., Second University of Naples, Abazia di S. Lorenzo, 81031 Aversa, Italy, uzy@libero.it)

Acoustics as well as the extraordinary architecture are substantial part of the heritage of ancient Greek and Roman theatres. These archaeological areas are more often used for classical and modern performing activities but to emphasise the original acoustics, restoration of the original shape, reconfiguration of the space and introduction of new technologies and materials may be needed. These activities must happen taking into account the "International Charter for The Conservation and Restoration of Monuments and Sites" as main tools defining conservation, use and maintenance guidelines in matter of discipline for ancient performing places. This paper presents some proposals for the stage reconstruction of the Benevento Roman Theatre and examine the associated effects on the theatre acoustics. Design proposals, minded to preserve the place authenticity, improving the acoustics and keeping intact the surrounding atmosphere for spectators, are the result of archaeological data, contemporary performances requirements, state of the art in matter of stage setting design. Acoustic field prediction is based on numerical models and auralization aiming to reproduce the real listening conditions.

## Contributed Paper

2:00

**4pAAa3. Acoustic characterization of the ancient theatre at Syracuse.** Marco Gullo (DREAM, Univ. of Palermo, Viale delle Scienze, Edificio 9, 90128 Palermo, Italy, mgullo@dream.unipa.it), Armando La Pica (DREAM, Univ. of Palermo, Viale delle Scienze, Edificio 9, 90128 Palermo, Italy, lapica@unipa.it), Giuseppe Rodono' (DREAM, Univ. of Palermo, Viale delle Scienze, Edificio 9, 90128 Palermo, Italy, rodono@unipa.it), Vincenzo Vinci (DREAM, Univ. of Palermo, Viale delle Scienze, Edificio 9, 90128 Palermo, Italy, vincenzo\_vinci@hotmail.it)

The ancient theatre at Syracuse, the largest open air theatre in Sicily, experienced many structural changes along its history. Today its very

renowned drama festival attracts visitors from all the world that fill the entire cavea: customized seats accommodation and a prominent stage play an important role in the theatre acoustics. Because no data were available in literature on its acoustical performance a measurement session was carried out by our team in the framework of a wide research project on ancient theatres acoustics in the modern use. From data recorded "room criteria" parameters have been evaluated together with spectral analysis in order to gain a deeper information on the acoustic field. Main results are listed and commented and a comparison among data collected on field during the team experience in the past years is reported.



## Invited Paper

2:20

**4pAAa4. On the use of ancient theatres for modern performances: a scale model approach.** Andrea Farnetani (Engineering Dept. - Univ. of Ferrara, Via Saragat, 1, 44100 Ferrara, Italy, andrea.farnetani@unife.it), Nicola Prodi (Engineering Dept. - Univ. of Ferrara, Via Saragat, 1, 44100 Ferrara, Italy, nicola.prodi@unife.it), Roberto Pompoli (Engineering Dept. - Univ. of Ferrara, Via Saragat, 1, 44100 Ferrara, Italy, roberto.pompoli@unife.it)

Ancient theatres are widely used today for modern performances including drama, music and ballets. Despite the state of conservation of the stage, the scenery is seldom designed with little care about its acoustical efficiency. Moreover, depending on the specific venue, a sound system can be employed in the performance. To clarify the acoustical impact of all these elements in ancient theatres, different stage settings and a sound system were investigated by means of scale model measurements. The scale model is a 1:20 scale reproduction of the ancient theatre of Siracusa (Italy). It is conceived as modular structure so that different configurations of the cavea and of the stage can be reproduced. To investigate the stage-set effects different groups of reflecting panels were arranged on the platform and an orchestra shell was tested too. Then, to simulate the sound system, two directional high frequency sources were assembled and optimized. The interplay of stage, sound system and theatre architecture was outlined by a comprehensive set of acoustical measurements.

## Contributed Paper

2:40

**4pAAa5. "Acoustics vases in ancient theatres: disposition, analysis from the ancient tetracordal musical system".** Arturo Barba Sevillano

(Grup d'Acustica Arquitectònica, Ambiental i Industrial, E.T.S.I.I, Univ. Politècnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, arbarse@doctor.upv.es), Radu Lacatis (Grup d'Acustica Arquitectònica, Ambiental i Industrial, E.T.S.I.I, Univ. Politècnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, rala1@doctor.upv.es), Alicia Giménez (Grup d'Acustica Arquitectònica, Ambiental i Industrial, E.T.S.I.I, Univ. Politècnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, agimenez@fis.upv.es), José Romero (Grup d'Acustica Arquitectònica, Ambiental i Industrial, E.T.S.I.I, Univ. Politècnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, romerof@fis.upv.es)

The acoustics of the ancient Greek and Roman theatres has always been rated as excellent by experts, without discussion. Beyond the purely architectural aspects, in this kind of outdoor theatres some mechanisms were used in order to improve the acoustics. In this paper we have studied the texts about "theatre's vases" of the famous book "On Architecture" by Vitruvius. Different interpretations and illustrations of these vases, that several translators carried out in the sixteenth to the eighteenth centuries, have been researched. From the wide bibliography consulted in this regard we have developed a plane with the disposition of the bronze vases in the theatres. In this plane we have specified the frequency of each one of them, and explained their disposition from the tetracordal musical system acquired from the Greek culture. Finally, an analysis of the disposition of the vases has been made. We have studied and looked for the musical intervals and harmonic relations among adjacent vases.

## Invited Papers

3:00

**4pAAa6. Musical granite pillars in ancient Hindu temples.** Paul Calamia (Rensselaer Polytechnic Institute, Greene Bldg., 110 8th St., Troy, NY 12180, USA, calamp@rpi.edu), Jonas Braasch (Rensselaer Polytechnic Institute, Greene Bldg., 110 8th St., Troy, NY 12180, USA, braasj@rpi.edu)

In some ancient Hindu temples built in India between the 7th century and the 16th century can be found a fascinating architectural element: intricately designed musical pillars of solid granite. These pillars, clusters of which were often carved from a single piece of stone, were tuned by means of their length, width, and tension (induced by a load from above) and were played to accompany devotional readings and dance performances. Various tunings and sound characteristics were employed based on the intended purpose. In this paper, examples of extant pillars will be presented and discussed within the context of their acoustical and architectural significance.

3:20

**4pAAa7. Convolution-scattering model for staircase echoes at the temple of Kukulkan.** David Lubman (DL Acoustics, 14301 Middletown Lane, Westminster, CA 92683, USA, dlubman@dlacoustics.com)

Chirped echoes from staircases at the temple of Kukulkan at Chichen Itza have stimulated much interest since first reported at scientific meetings by the author in 1998. Among them are Declercq et al (J. Acoust. Soc. Am. 116, (6) 2004). They correctly observed that the echo depends strongly on the "type" of incident sound, but offered no explanation. Those authors overlooked an earlier explanation given at the First Pan-American/Iberian Meeting on Acoustics at Cancun, Mexico (Lubman, J. Acoust. Soc. Am. 112, (5) (2002). It included a mathematical model of the chirped echo effect and a full simulation of the chirped echo, including auralization. This model explicitly shows echo dependence on the time-frequency structure of the impinging sound. With it, ethnomusicologists and others can conveniently simulate the staircase echo for any recorded sound stimulus, including handclaps, voices, and ethnic sound instruments. The clap-echo system is modeled as a time-invariant linear system. The echo time series is found by convolving the stimulus (e.g., handclap) with the staircase impulse response. This earlier model is far more computationally efficient than the Declercq model, and achieves greater accuracy by avoiding the false and needless assumptions of plane wave propagation and infinite corrugated periodic surfaces.

*Invited Paper*

5:00

**4pAAa8. On the Acoustics of the Underground Galleries of Ancient Chavín de Huántar, Peru.** Jonathan S. Abel (Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180, USA, abel@ccrma.stanford.edu), John W. Rick (Stanford University, Archaeology Center, P.O. Box 20446, Stanford, CA 94309, USA, johnrick@stanford.edu), Patty P. Huang (Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180, USA, pph@ccrma.stanford.edu), Miriam A. Kolar (Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180, USA, kolar@ccrma.stanford.edu), Julius O. Smith (Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180, USA, jos@ccrma.stanford.edu), John M. Chowning (Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180, USA, jc@ccrma.stanford.edu)

Chavín de Huántar is a monumental World Heritage archaeological site in the Peruvian highlands predating Inca society by over 2000 years. The importance of site acoustics is suggested by distinctive architectural features, notably an extensive network of underground galleries used in part for ritual purposes. The labyrinthine galleries are arranged in a series of small rectangular alcoves off narrow corridors. Here we focus on measuring and modeling the acoustics at Chavín to understand the implications of auditory experiences within the galleries as related to the site's role in developing religious authority. Acoustic measurements and models of a site can be used to archive site acoustics, estimate the acoustics of inaccessible or alternative site architectures, and reconstruct original site acoustics; they may also be used to corroborate aspects of rituals suggested by other archaeological data. Preliminary acoustic measurements at Chavín show a short reverberation time, dense and energetic early reflections, and a large lateral energy fraction. The short reverberation time would enable rhythmically articulated group performance using Strombus shell trumpets found onsite. The early reflections would provide strong acoustic reinforcement and consistent resonances for participants in gallery alcoves. The wide sound-field would envelop participants, contributing to the ritual experience.

*Contributed Paper*

5:20

**4pAAa9. Assessment of Acoustical Characteristics for Historical Baths (Hammams).** Asuman Aydin (Middle East Technical University, Dept of Architecture, METU (ODTU), Inonu Blvd, 06531 Ankara, Turkey, aydinasuman@yahoo.com), Ayse Tavukcuoglu (Middle East Technical University, Dept of Architecture, METU (ODTU), Inonu Blvd, 06531 Ankara, Turkey, aysetavukcuoglu@yahoo.com), Mehmet Caliskan (Middle East Technical University, Dept of Architecture, METU (ODTU), Inonu Blvd, 06531 Ankara, Turkey, caliskan@metu.edu.tr)

Comprehensive studies are necessary to better understand the original acoustical characteristics of historical baths (hammams) in order to discover the historical technologies establishing acoustical properties in these structures and keep their proper functioning for long periods of time. The study was conducted on five historical Turkish hammams, belonging to the Otto-

man period. Their acoustical performances were examined in terms of basic acoustical parameters, such as, reverberation time, early decay time, clarity, lateral fraction, sound transmission index. The 3D computer modelling and acoustics simulation were done for each structure by "ODEON combined 8.5". Their audio performance was evaluated by taking into consideration of the volume, materials use and environmental conditions. The analyses were done for the case of the unoccupied condition while one of the structures, Sengul Hammam in Ankara, was analysed both for non- and fully-occupied conditions. The joint interpretation of the results was done in order to define the acoustical features of the hammam structures. The potentials for improving the acoustical performance of interiors were also discussed. The use of 3D computer modeling and acoustical simulation, adapted for the case of historical hammam structures, provided a good opportunity for the assessment of their acoustical performance on quantitative basis.

*Invited Papers*

5:40

**4pAAa10. History and acoustics of the Asian free-reed mouth organs.** James Cottingham (Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52402, USA, jcotting@coe.edu)

Mouth-blown instruments employing a free reed coupled to a pipe resonator have long been known and used throughout East and Southeast Asia. Details of the origin and development of these instruments are not known, but are closely connected with the history and prehistory of a multitude of ethnic groups. Free reed instruments have been employed in a variety of ways, from simple signaling devices to use in the court music of Japan and China. The pipe resonators vary from the buffalo horn to bamboo pipes of nearly cylindrical cross section. The instruments exemplify a pipe-resonator coupling significantly different from that of the standard wind instruments of European origin. In some cases the reed is at or near one end of an open or closed pipe resonator, but in other examples the reed is mounted in the side of the resonator away from the ends. A summary of recent experimental investigations of these instruments will be presented, along with musical examples.

**4pAAa11. The mridangam: A study of the history and acoustics of an ancient South Indian drum.** Rohan Krishnamurthy (Kalamazoo College, 544 Sunrise Circle, Kalamazoo, MI 49009, USA, rohan@rohanrhythm.com), Ian Hempe (Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52402, USA, ikhempe@coe.edu), James Cottingham (Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52402, USA, jcotting@coe.edu)

The acoustical properties of the South Indian drum, the mridangam, were studied. The barrel-shaped mridangam has been described in ancient Hindu scriptures and depicted in cave paintings and temple sculptures. With a claimed antiquity dating back to the Vedic period, it is the principal percussion instrument in South Indian classical music and dance and possesses unique tonal properties. The mridangam is comprised of three primary parts: The tonal head (valanthalai), the bass head (thoppi), and the central shell (kattai), to which the two heads are traditionally fastened by leather rope. Measurements of modes and mode frequencies were made on traditional drums, as well as on drums where the heads were remounted using a new and user-friendly design. Measurements of drumhead vibration and sound spectra were also made when the drumhead was excited by a skilled player using standard strokes. The frequencies of the first few modes of the tonal head were found, as expected, to be tuned approximately harmonic. Practical performance variables, including effects of altering mounting tension and coupling between the drumheads, were also studied. Results from the study will be followed by a practical demonstration of the instrument.

**4pAAa12. Auditory capacities of human fossils: A new approach to the origin of speech.** Ignacio Martínez ((1) Universidad de Alcalá de Henares; (2) Centro UCM-ISCIH de Evolución y Comportamiento Humanos, Centro UCM-ISCIH de Evolución y Comportamiento Humanos, C/ Sinesio Delgado, N° 4, Pabellón 14, 28029 Madrid, Spain, imartinezm@isciii.es), Rolf Michael Quam ((1) American Museum of Natural History; (2) Centro UCM-ISCIH de Evolución y Comportamiento Humanos, Division of Anthropology, American Museum of Natural History, Central Park West @ 79th St., New York, NY 10024, USA, rquam@amnh.org), Manuel Rosa (Universidad de Alcalá de Henares, Edificio de Ciencias, Campus Universitario, 28871 Alcalá de Henares, Spain, manuel.rosa@uah.es), Pilar Jarabo (Universidad de Alcalá de Henares, Edificio de Ciencias, Campus Universitario, 28871 Alcalá de Henares, Spain, mpilar.jarabo@uah.es), Carlos Lorenzo ((1) Universitat Rovira i Virgili (2) Centro UCM-ISCIH de Evolución y Comportamiento Humanos, Universitat Rovira i Virgili, Facultat de Lletres Institut de Paleoecologia Humana, Area de Prehistoria, i Evolució Social, Imperial Tarraco, 1, 43005 Tarragona, Spain, clorenzo@prehistoria.urv.es), Juan Luis Arsuaga ((1) Universidad Complutense de Madrid; (2) Centro UCM-ISCIH de Evolución y Comportamiento Humanos, Centro UCM-ISCIH de Evolución y Comportamiento Humanos, C/ Sinesio delgado, N° 4, Pabellón 14, 28029 Madrid, Spain, jlarsuag@isciii.es)

The origin and evolution of human language has mainly dealt with the reconstruction of the upper respiratory tract of human fossils. After decades of controversy no clear results have arisen from these studies. We propose a new approach to this issue based on the possibility to reconstruct the sound power transmission, through the external and middle ear, in fossil specimens. The results thus obtained in the more than 500 ky old fossils from the Sima de los Huesos site (Sierra de Atapuerca, Spain) show that this hominins had the same auditory capacities as modern human, suggesting an older origin for speech than any previous study.

**4pAAa13. The marvellous sound world in the 'Phonurgia Nova' of Athanasius Kircher.** Lamberto Tronchin (DIENCA - CIARM, University of Bologna, viale Risorgimento, 2, I-40136 Bologna, Italy, tronchin@ciarm.ing.unibo.it), Ilaria Durvilli (DIENCA - CIARM, University of Bologna, viale Risorgimento, 2, I-40136 Bologna, Italy, ilaria.durvilli@mail.ing.unibo.it), Valerio Tarabusi (DIENCA - CIARM, University of Bologna, viale Risorgimento, 2, I-40136 Bologna, Italy, ciarm02@hotmail.com), Galia Mastromatteo (DIENCA - CIARM, University of Bologna, viale Risorgimento, 2, I-40136 Bologna, Italy, gam@ciarm.ing.unibo.it)

Athanasius Kircher, Jesuit, was born in Geisa, Thüringen, in 1608. He spent a large amount of his life in Rome, where he died in 1680. He was active in many different topics, ranging from geology to philosophy. He was the author of many books at his time, among all the Musurgia Universalis, written in 1650, and the Phonurgia Nova, of 1673. Whilst the Musurgia Universalis gathered a wide attention also in recent years, the Phonurgia Nova, which means "a new method of sound production", was only recently rediscovered. In this paper the original Latin version of the Phonurgia Nova, which comprises two different books, i.e. the "Phonosophia nova" and the "Phonosophia anacampica", is analyzed. The first book deals with the influence of music in the human being, whereas the second book analyses the sound propagation in enclosure. This book reveals some interesting acoustic apparatus for sound production and propagation, as the "tuba stentorophonica" (the loud trumpet), the "statua citofonica" (the talking statue). Some of these phonic apparatus are described, analyzed and commented.

## Session 4pAAb

## Architectural Acoustics: Archeological Acoustics II (Poster Session)

David Lubman, Cochair  
*DL Acoustics*

Jens Holger Rindel, Cochair  
*Odeon A/S*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pAAb1. Acoustics of Unique Baroque Theatre in Český Krumlov.** Jana Dolejsi (Studio D - Akustika s.r.o., Zizkova 12, 371 22 Ceske Budejovice, Czech Republic, jana.dolejsi@akustikad.com), Pavel Slavko (Státní hrad a zámek, 381 01 Cesky Krumlov, Czech Republic, castle@ckrumlov.cz), Ladislav Pouzar (Chvalšinská 231, 381 01 Cesky Krumlov, Czech Republic, ladislav@pouzar.cz), Monika Rychtarikova (Lab. ATF, Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, Monika.Rychtarikova@bwk.kuleuven.be)

The castle theater in the Český Krumlov is a unique historical site in Europe which represents a valuable example of a theater with Baroque scenes from the late 18th century. It was preserved with exceptional completeness and authenticity since it has never burned, and during the last two centuries it was not modernized. The baroque period must be understood as a universal style with many methods of expression which act in harmony, and the individual elements must complement each other and be interconnected to produce an effective unit. It was a challenge to experimentally prove these relationships and connections in as wide context as possible (i.e. relations between the architecture, contemporary scene lighting, perspectives and proportions, roomacoustic properties, contemporary musical interpretations and vocal techniques, placing of the orchestra and other. According to musicians, actors and audience is this theatre known as a hall with excellent acoustic condition. However, no room acoustic measurements were performed in the theatre in the last century. This article report on results from the acoustical measurements and simulation in the main hall of the theatre which help us to understand which acoustical values people find pleasant in the room of this character.

**4pAAb2. Acoustical reconstruction of San Petronio Basilica in Bologna during the Baroque period: the effect of festive decorations.** Francesco Martellotta (DAU - Politecnico di Bari, via Orabona 4, 70125 Bari, Italy, f.martellotta@poliba.it), Ettore Cirillo (DAU - Politecnico di Bari, via Orabona 4, 70125 Bari, Italy, e.cirillo@poliba.it), Sabrina Della Crociata (DAU - Politecnico di Bari, via Orabona 4, 70125 Bari, Italy, sadell1984@libero.it), Emanuele Gasparini (via Centurare 33, 37062 Dosobuono, Italy, emanuele.gasparini@libero.it), Daniela Prezioso (DAMS - Alma Mater Studiorum Univ. di Bologna, via Barberia, 4, 40123 Bologna, Italy, aleinad2002@libero.it)

The Basilica of San Petronio in Bologna (Italy) is a large Gothic church characterized by three naves divided by cluster piers made of brick and flanked by square chapels. It is 130 m long, 60 m wide and 44 m high, developing a volume of 170000 m<sup>3</sup>. The widespread use of smooth plaster and the substantial lack of decoration give rise to a reverberation time (in unoccupied conditions) which varies from about 13 s at 125 Hz to 5 s at 4 kHz, with an average of 10.7 s at mid frequencies. In occupied conditions the expected mid-frequency reverberation time should lower to about 7 s. None-

theless, these acoustic conditions appear scarcely compatible with the characteristics of the Baroque music which was composed for the "Cappella musicale" during the 17th century. However, historical research pointed out how, in that period, rich draping and curtains were often used during the major religious and civil celebrations. The analysis of the acoustic consequences of such temporary installations was performed by means of acoustic simulation based on historical records calibrated on the current configuration of the church. The paper presents the results of such reconstruction.

**4pAAb3. Assessment of strong reflections in ancient theatres: Spatial information from parallel measurement data.** Giuseppe Rodonò (DREAM, Univ. of Palermo, Viale delle Scienze, Edificio 9, 90128 Palermo, Italy, rodon@unipa.it), Marco Gullo (DREAM, Univ. of Palermo, Viale delle Scienze, Edificio 9, 90128 Palermo, Italy, mgullo@dream.unipa.it), Armando La Pica (DREAM, Univ. of Palermo, Viale delle Scienze, Edificio 9, 90128 Palermo, Italy, lapica@unipa.it), Vincenzo Vinci (DREAM, Univ. of Palermo, Viale delle Scienze, Edificio 9, 90128 Palermo, Italy, vincenzo\_vinci@hotmail.it)

The sound field in ancient open-air theatres shows a finite time-response in a transient. The structure of the time-response depends on geometrical characteristics of the theatre and source-receiver position: upon geometry dimension of the stage area its possible to observe nearly strong reflections and quantify the associated delay time. Sampling the theatre space with a single microphone does not allow any directional information on the sound field but the parallel use of more microphones and software post-processing could add spatial information. Dataset consists of four synchronized impulse response measured along a radial direction in the theatre plan for different source positions. A directional receiver system based on a post-processing method has been implemented and applied to measurement data. The obtained experimental results are reported and discussed.

**4pAAb4. Ultrasonic quantitative strength assessment of artificially aged and archaeological wood samples.** Ari Salmi (Univ. of Helsinki / Dept. of Physical Sciences, POB 64 (Gustaf Hållströmin katu 2), 00014 Helsinki, Finland, edward.haeggstrom@helsinki.fi), Kari Steffen (P.O.B 56 University of Helsinki, 00014 Helsinki, Finland, kari.steffen@helsinki.fi), Joona Eskelinen (Univ. of Helsinki / Dept. of Physical Sciences, POB 64 (Gustaf Hållströmin katu 2), 00014 Helsinki, Finland, joona.eskelinen@helsinki.fi), Edward Hægström (Electronics Research Unit, University of Helsinki, P.O.Box 64 (Gustaf Hållströmin katu 2), FIN-00014 Helsinki, Finland, edward.haeggstrom@helsinki.fi)

A large fraction of discovered archaeological artefacts are wooden. Since materials grow brittle and their stiffness decreases over time, it is vital for archaeologists to have methods that assess the strength of the object prior to



moving it. We present preliminary shear and elastic modulus data measured using 100 kHz longitudinal and shear ultrasonic tone burst through-transmission. Artificially aged wooden samples and archaeological samples of known age were used to validate the aging procedure by comparison of the shear and elastic modulus. We also measured the modulus of water-logged samples obtained from the ship wreck *Vrouw Maria* sunken in the Finnish archipelago in 1771, and compared to samples artificially aged to similar age. Scanning electron microscopy (SEM) was used to compare the structure of artificially and naturally aged samples. The results presented may be used to create an artefact model for strength assessment, and to give guidelines of strength vs. age for archaeologists to support their logistics decisions.

**4pAAb5. The Parpalló Cave: A singular archaeological acoustic site.**  
Noé Jiménez González (Universidad Politécnica de Valencia, C/ La trampa

Nº 2, 02520 Chinchilla de Montearagón, Spain, nojigon@epsu.upv.es), Rubén Picó (EPSG - Univ. Politécnica de Valencia, c/ Nazaret-Oliva s/n, 46780 Grau de Gandia, Spain, rpico@fis.upv.es), Javier Redondo (IGIC - Universitat Politècnica de València, Cra. Nazaret-Oliva S/N, E-46730 Gandia, Spain, fredondo@fis.upv.es)

The Parpalló Cave is located in the slopes of Montdúber, in Valencia (Spain). It is one of the most important Palaeolithic sites, not only in Spain, but in the world. It was a privileged location for its inhabitants from the Upper Palaeolithic onwards. It has one of the most spectacular collections of Palaeolithic art mobilier found to date. The Parpalló cave features a considerably large opening which, undoubtedly, influences its acoustic properties. Indeed, the sound pressure field inside the cave is not excessively reverberant and intelligibility is significantly better than in similar enclosures. In this work a study of the acoustic properties of the Parpalló cave is performed by using numerical simulation.

THURSDAY AFTERNOON, 3 JULY 2008

P2-B, LEVEL 2, 3:40 TO 5:20 P.M.

### Session 4pAAc

## Architectural Acoustics and Psychological and Physiological Acoustics: Speech Segregation in Rooms II (Poster Session)

Barbara Shinn-Cunningham, Cochair  
*Boston University Hearing Research Center*

Mathieu Lavandier, Cochair  
*Cardiff University*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pAAc1. Effects of spatial factors of speech, noise, and reverberation sounds on listening difficulty.** Hayato Sato (Environmental Acoustics Laboratory, Graduate School of Engineering, Kobe University, Rokkodai 1-1, Nada, 657-8501 Kobe, Japan, hayato@kobe-u.ac.jp), Tomoyasu Takaoka (Environmental Acoustics Laboratory, Graduate School of Engineering, Kobe University, Rokkodai 1-1, Nada, 657-8501 Kobe, Japan, Tomoyasu.Takaoka@mb2.seikyoku.ne.jp), Masayuki Morimoto (Environmental Acoustics Laboratory, Graduate School of Engineering, Kobe University, Rokkodai 1-1, Nada, 657-8501 Kobe, Japan, mrmmt@kobe-u.ac.jp)

It is well known that the amount of auditory masking in binaural listening is affected by spatial factors of signal and masking sounds such as the degree of interaural cross-correlation and the arriving direction. The amount of auditory masking for speech signal are usually measured using speech intelligibility tests. However, in the sound fields that appears in everyday life, speech intelligibility are usually close to 100%. Therefore, it was unclear how much the effects of spatial factors on speech communication in everyday life are. Listening tests were performed to clarify the effects of spatial factors on listening difficulty ratings [M. Morimoto *et al.*, J. Acoust. Soc. Am. **116**, 1607-1613 (2004)] focussing on the degree of interaural cross-correlations of speech, noise, and reverberation sounds, and on the arriving direction of speech signal. The two spatial factors were controlled by

several loudspeakers which are horizontally installed in an anechoic room. The results of listening tests showed that the effects of the two spatial factors on listening difficulty ratings decreased with improving listening conditions, and the effects can be ignored in the sound fields that often appears in everyday life.

**4pAAc2. Binaural enhancement of speech intelligibility metrics.** Florent Deux (Chalmers University of Technology, Division of Applied Acoustics, Chalmers Room Acoustic Group, SE-41296 Göteborg, Sweden, florentdeux@gmail.com), Mendel Kleiner (Chalmers Rm. Acoust. Group, Dept. of Appl. Acoust., Chalmers Univ. of Technology, 41296 Gothenburg, Sweden, mendel.kleiner@chalmers.se)

Efficient methods of quantifying speech intelligibility are needed for designing and understanding functional architectural spaces. All current measures of speech intelligibility are based on monaural impulse response, which excludes consideration of important binaural aspects of human hearing, including dereverberation and decoloration. Acquisition of binaural data has increasingly become common practice, using dummy heads or in-ear microphones, making the development of binaural intelligibility measures especially timely. The need for such a measure is illustrated by calculations of speech transmission index (STI) with a single energy-based impulse obtained through combination of binaural data channels. Each method of

combination produces significantly different STI values that illustrate the impact of orientation and location on intelligibility calculations. The relationship between such estimation variation and subjective experience must be studied to determine the research direction for a much needed, uniquely binaural speech intelligibility measure. The aim of this research is then to

provide a guide to be used in comparing values obtained by the various methods. The project will then involve measurements of psychoacoustic metrics for speech intelligibility by objective methods as well as determination of speech intelligibility by subjective methods using tests with binaural recording and playback listening.

THURSDAY AFTERNOON, 3 JULY 2008

P2-B, LEVEL 2, 3:40 TO 5:20 P.M.

## Session 4pAAd

### Architectural Acoustics: Architectural Acoustics Potpourri III (Poster Session)

Ingo Witew, Cochair

*Institute of Technical Acoustics, Templergraben 55, Aachen, 52056, Germany*

Byron Harrison, Cochair

*Talaska, 1033 South Boulevard, Oak Park IL 60302, USA*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pAAd1. Analysis of the impact of sound diffusion in the reverberation time of an architectural space - A proposal for the characterization of diffusive surfaces using scale models.** Elisa Garay-Vargas (Program of Graduate Studies in Design, CyAD - Universidad Autónoma Metropolitana, Av. San Pablo 180, Edificio D 101, Col. Reynosa Tamaulipas, Delegación Azcapotzalco, 02200 Mexico, D.F., Mexico, [elisagaray@gmail.com](mailto:elisagaray@gmail.com)) , Fausto E. Rodriguez-Manzo (Departamento de Procesos y Técnicas, CyAD, Universidad Autónoma Metropolitana-Azcapotzalco, Av. San Pablo 180, Edificio H-PB, Col. Reynosa Tamaulipas, Delegación Azcapotzalco, 02200 Mexico, D.F., Mexico, [rfme@correo.azc.uam.mx](mailto:rfme@correo.azc.uam.mx))

The acoustic quality of architectural spaces has been related directly with the reverberation phenomenon, in order to understand more about sound in architecture, this proposal tries to conclude how diffusion acts on the reverberation time. The fact that architects deal with absorption coefficients and change the reverberation time by playing and designing with materials, opens a door to the possibility of new ways of architectural design, knowing how simple or complex and even natural diffusion surfaces will act in space. Several examples of the use of diffusion and reverberation in architectural spaces can be found in buildings like the Jewish Museum from Daniel Libeskind or the Therme Vals from Peter Zumthor, the use of big surfaces of concrete in the first and uneven surfaces in the second, transform each of these places in one with very interesting sound qualities. To create spaces like these we can take diffusion as a design argument but, How will certain diffusive surfaces influence on the reverberation time of an architectural space? To answer this question it is necessary to experiment with scale models measuring the reverberation time, and not the diffusion itself.

**4pAAd2. Re-evaluating Noise Criterion in Digital Audio Recording Environments.** Linda Gedemer (5930 Penfield Ave., Woodland Hills, CA 91367, USA, [gedeml@rpi.edu](mailto:gedeml@rpi.edu))

The absence of audible noise in recording studios is highly desirable. For this reason, the acoustic design of recording facilities usually involves an accurate assessment of the audibility of background noise. However, noise criterion that sets levels for inaudible noise do not take in account the sensitivity of today's digital recording equipment and post production processing. Noise-level criterion and measurement techniques exist in order to create an atmosphere where noise is rated at a level of audibility with

respect to the type of space being measured. In this paper these criterion will be analyzed in terms of critical recording spaces. The criterion will be considered in terms of current building practices and digital audio recording equipment used. A set of questions were sent to working sound engineers in order to receive first hand opinions regarding noise issues in recording studios. It is not the intention of this report to suggest a new noise criterion but rather to better understand existing problems which could then help in the creation of a more relevant noise criterion to be developed in the future.

**4pAAd3. Possible correlation between acoustic and thermal performances of building structures.** Giovanni Semprini (University, DIENCA Dept. Facoltà di Ingegneria, Viale Risorgimento 2, 40136 Bologna, Italy, [giovanni.semprini@mail.ing.unibo.it](mailto:giovanni.semprini@mail.ing.unibo.it)), Alessandro Cocchi (University, DIENCA Dept. Facoltà di Ingegneria, Viale Risorgimento 2, 40136 Bologna, Italy, [alessandro.cocchi@mail.ing.unibo.it](mailto:alessandro.cocchi@mail.ing.unibo.it)), Cosimo Marinosci (DIENCA - Univ. of Bologna, Viale Risorgimento 2, 40136 Bologna, Italy, [cosimo.marinosci@mail.ing.unibo.it](mailto:cosimo.marinosci@mail.ing.unibo.it))

Most European standards required high performance values for sound and thermal insulation in building structures, according to Directive EEC 89/106. Sound transmission and heat transfer in structures have different physical and analytical approach and specific parameters of performance (i.e. sound transmission loss or thermal transmittance) are not directly correlated each others; many kind of structures have also different behaviour depending on mechanical properties of materials, numbers of layers of materials, etc. The aim of this work is to analyse possible correlation between sound transmission performances and thermal properties values in order to evaluate common trends related to physical properties of the various building components, like for example density or surface mass.

**4pAAd4. On the use of a corrugated ceiling for noise reduction in rooms.** Nico F. Declercq (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, [nico.declercq@me.gatech.edu](mailto:nico.declercq@me.gatech.edu)), Katelijn Vanderhaeghe (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS

2958, 2 rue Marconi, 57070 Metz, France, declercq@ieee.org), Katrien Dewijngaert (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, declercq@ieee.org), Patricia Verleysen (Ghent University, Mechanics of Materials and Structures, Sint-Pietersnieuwstraat 41, 9000 Ghent, Belgium, Patricia.Verleysen@ugent.be)

In certain halls the acoustics is dominated by distinct frequencies having disturbingly high intensities caused by yelling children, such as in an indoor swimming pool, or by industrial production processes. The use of damping materials to cover the ceiling, floor and side walls is not always permitted because of esthetic reasons or because of safety grounds or even humidity. The current study shows that in those cases the use of a corrugated surface is possible and is actually more effective in filtering out very distinct frequencies. An example is presented for frequencies at 3 to 4 kHz and numerical results are obtained in support of the use of a corrugated structure rather than classic damping materials such as baffles.

**4pAAAd5. Musicians' Adjustment of Performance to Room Acoustics, Part III: Understanding the Variations in Musical Expressions.** Kosuke Kato (Center for Advanced Science and Innovation, Osaka University, Yamadaoka 2-1, Suita-shi, 565-0871 Osaka, Japan, kato@casi.osaka-u.ac.jp), Kanako Ueno (Institute of Industrial Science, University of Tokyo, Komaba 4-6-1, Meguro-ku, 153-8505 Tokyo, Japan, ueno@iis.u-tokyo.ac.jp), Keiji Kawai (Graduate School of Science and Technology, Kumamoto University, Kurokami 2-39-1, Kumamoto-shi, 860-8555 Kumamoto, Japan, kawai@arch.kumamoto-u.ac.jp)

This paper attempts to investigate the acoustic variations in the musical sound signals produced by professional performers under different room acoustic conditions. The sound signals produced by four professional instrumentalists and an operatic baritone singer under simulated concert hall sound fields were recorded by placing a unidirectional microphone close to either the instruments or the mouth of the opera singer. In order to quantify the extent of the variations in the resulting sound signals due to the adjustment of the musical expressions, an acoustic analysis was conducted. The results indicated that the "note-on ratio," defined as the ratio of tone duration and inter-onset interval, of several staccato tones was decreased in reverberant halls. The higher harmonics of tones of an oboist and a flutist were suppressed in reverberant sound fields, while the vibrato extent of a violinist was considerably varied as was reported in our previous study (Part II). Based on the results of the interviews with the performers during the recordings, it was inferred that the variations in the musical sound signals were produced by the performers to adjust to the room acoustic conditions.

**4pAAAd6. In situ measurement of acoustic parameters with smart devices.** Karl Van Nieuwenhuyse (Vrije Universiteit Brussel, Acoustics and Vibration Research Group, Dept. of Mechanical Engineering, Pleinlaan 2, BE-1050 Brussels, Belgium, karl.van.nieuwenhuyse@skynet.be), Patrick Guillaume (Vrije Universiteit Brussel, Acoustics and Vibration Research Group, Dept. of Mechanical Engineering, Pleinlaan 2, BE-1050 Brussels, Belgium, Patrick.Guillaume@vub.ac.be), Steve Van Landuit (Vrije Universiteit Brussel, Acoustics and Vibration Research Group, Dept. of Mechanical Engineering, Pleinlaan 2, BE-1050 Brussels, Belgium, svanlandui@vub.ac.be), Cedric Vuye (Hogeschool Antwerpen, Dept. of Industrial Sciences, Paardenmarkt 92, BE-2000 Antwerpen, Belgium, c.vuye@ha.be), Gert De Sitter (Vrije Universiteit Brussel, Acoustics and Vibration Research Group, Dept. of Mechanical Engineering, Pleinlaan 2, BE-1050 Brussels, Belgium, gert.de.sitter@vub.ac.be)

Instruments for in situ analysis of the acoustic parameters of a room are, due to their high prime cost, not widely accessible. In an effort to change this, new software was developed to enable these measurements by use of highly portable consumer electronics such as PDA's and Smart phones. The newly developed software calculates the reverberation time using the impulse method and aims at delivering accurate results using the built-in microphone or an external one. This paper will discuss the accuracy of the new software in comparison with professional equipment through several case

studies, even as the different techniques used to overcome the inherent problems forthcoming the equipment used, i.e. microphone quality and automatic gain control.

**4pAAAd7. Efficiency of two-dimensional interpolation algorithms for high-quality dynamic binaural synthesis.** Karim Helwani (Department of Audio Communication, Technical University of Berlin, Sekr. EN-08, Einsteinufer 17c, 10587 Berlin, Germany, karim.helwani@gmx.de), Alexander Lindau (Department of Audio Communication, Technical University of Berlin, Sekr. EN-08, Einsteinufer 17c, 10587 Berlin, Germany, alexander.lindau@tu-berlin.de), Stefan Weinzierl (Department of Audio Communication, Technical University of Berlin, Sekr. EN-08, Einsteinufer 17c, 10587 Berlin, Germany, stefan.weinzierl@tu-berlin.de)

Binaural synthesis of acoustical environments is based on binaural room impulse responses (BRIRs) measured with an angular resolution of typically between 1° and 15°. Considering the size of the resulting BRIR database used for auralization and the long measurement duration for its acquisition, it is reasonable to use interpolation from a lower resolution BRIR grid. Based on a mathematical formulation of the interpolation problem for BRIRs, a set of different solutions for two-dimensional spaces is described and compared with regard to efficiency and real time performance. In order to evaluate the degradation introduced by interpolation, different methods have been implemented and applied to a HRTF and a BRIR database measured in two degrees of freedom with 1°/1° horizontal/vertical resolution. These are bilinear interpolation (i.e. nearest neighbour, inverse distance weighting), spherical spline interpolation, wavelet interpolation, and interpolation based on principal component analysis (PCA). A listening test following an ABX procedure has been performed to evaluate the efficiency of the different interpolation methods according to the detection rates of interpolated versus measured BRIR databases in a dynamically auralized acoustical environment.

**4pAAAd8. Perception of teachers and pupils to the acoustics of classrooms.** Carmen L. Loro (Federal University of Parana, Centro Politécnico - Setor de Tecnologia, Bairro Jardim das Américas, 81531-990 Curitiba, Brazil, loro@arnet.com.br), Paulo Henrique Trombetta T. Zannin (Federal University of Parana, Centro Politécnico - Setor de Tecnologia, Bairro Jardim das Américas, 81531-990 Curitiba, Brazil, paulo.zannin@pesquisador.cnpq.br)

The present survey has evaluated the acoustic characteristics of classrooms built as modular classrooms. The study has focused on the background noise, reverberation time, noise insulation and interviews with teachers and pupils. The acoustic comfort of classrooms in a Brazilian public school has been evaluated through interviews with 62 teachers and 464 pupils. Acoustic measurements have revealed the poor acoustic quality of the classrooms. The walls between the classrooms and the corridor have permanent ventilation openings on glassy bricks. The measured weighted apparent sound reduction index (ISO 140-4; ISO 717-1) for the wall, with a door and glassy brick, was  $R'w = 17$  dB. The low value of the weighted apparent sound reduction index  $R'w = 17$  dB contributes significantly to the noise transmission from one room to the other, contributing to the elevated levels of background noise inside the classrooms. Interviews with teachers and pupils have shown that the main noise sources noted in the classrooms originate inside the school: voices of students and of the teacher of the neighboring classroom.

**4pAAAd9. Comparing the sound absorption of different objects.** Robert Hickling (Sonometrics Inc., 8306 Huntington Road, Huntington Woods, MI 48070-1643, USA, sonometrics@comcast.net), John Kopko (Kolano and Saha Engineers Inc., 3559 Sashabaw Road, Waterford, MI 48329, USA, jkopko@kandse.com), Pranab Saha (Kolano and Saha Engineers Inc., 3559 Sashabaw Road, Waterford, MI 48329, USA, prsaha@kandse.com)

Current methods of measuring sound absorption involve a piece of material. Small pieces are tested in impedance tubes. Larger pieces are tested in a reverberation room, using the reverberation-time decay method, as in ISO Standard 354. However these procedures do not determine the absorption of a complete object. It is proposed to determine the absorption of

an object by measuring the negative sound power of the object in a reverberation room using sound intensity. It has been shown that sound power using sound intensity can be measured accurately in reverberation rooms (R. Hickling, Proceedings NOISE-CON 97, pages 483-488). Negative sound power on its own, however, cannot provide a satisfactory comparison indicator between objects because it depends on the room. It is necessary, there-

fore, to develop a normalization procedure to make the negative sound-power measurement independent of the room. A procedure is described which was tested for the same object (a car seat) in two different reverberation rooms. The results are encouraging. It is planned to conduct further tests using a variety of objects in different rooms to check the validity of the normalization procedure.

THURSDAY AFTERNOON, 3 JULY 2008

P2-B, LEVEL 2, 3:40 TO 5:20 P.M.

### Session 4pAAe

## Architectural Acoustics and Noise: Acoustics and Privacy in Healthcare Facilities II: Emerging Research Around the World II (Poster Session)

William Cavanaugh, Cochair  
*Cavanaugh Tocci Associates, Inc.*

Kerstin Persson Waye, Cochair  
*Dept. of Environ. Medicine, The Sahlgrenska Acad. of Gothenburg Univ.*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Paper

**4pAAe1. The noise control in laboratories and health areas: A keyword in order to practice biosafety.** Marta Ribeiro Valle Macedo (Fundação Oswaldo Cruz, Avenida Brasil 4365 - Manguinhos, 21040-360 Rio de Janeiro, Brazil, [mrbeiro@fiocruz.br](mailto:mrbeiro@fiocruz.br))

The noise control in laboratories and health areas: a keyword in order to practice biosafety Marta Ribeiro Valle Macedo Coordenação de Saúde do trabalhador, Fundação Oswaldo Cruz, Av. Brasil 4365, Manguinhos-Rio de Janeiro, Brasil, CEP.21040-360 The environmental noise control is an increasing preoccupation at the international level in order to observing the effects on the human health. At the outdoor environment or in habitations, in work places and health areas, these effects are verified, as mentioned by World

Health Organization (1995; 1999) and in rich bibliography about this subject. In areas with biological and chemical risk, where the development of activities need of great concentration, noise is a stressor agent and causes direct and indirect effects on the human health as well as affecting the intelligibility of the speech, the performance of activities and causing work accidents. However, the exposition to the main sound sources found in laboratories and hospitals located at the Fundação Oswaldo Cruz (Fiocruz), a Brazilian health institution, could have been avoided if some noise control measures had been employed. This paper presents the main situations observed at the evaluation developed during 2007, in 744 environments of work and shows how environmental noise comes interfering at the workers performance in Fiocruz.

4p THU. PM



## Session 4pAAf

**Architectural Acoustics: Case Studies and Design Approaches II (Poster Session)**

Ingo Witew, Cochair

*Institute of Technical Acoustics, Templergraben 55, Aachen, 52056, Germany*

Byron Harrison, Cochair

*Talaska, 1033 South Boulevard, Oak Park IL 60302, USA*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

**Contributed Papers**

**4pAAf1. Predicting the acoustics of historic Istanbul Tunnel: Simulation, calculation methods and geometrical details.** Filiz Bal Kocyigit (Karabuk University, 232. Str. No: 5/5 Ilkbahar Mah., Cankaya, 06550 Ankara, Turkey, filizbkocyigit@yahoo.com), Sertan Senturk (20th Street 17/6 Bahçelievler, 06490 Ankara, Turkey, sertansenturk@yahoo.com)

The Istanbul Tunnel, which was designed and constructed by Eugene Henry Gavand at 1875 is the 3rd. Metro and 2nd underground railway system in the world after Washington Metro (1868). In this paper, the acoustics of the Istanbul Tunnel are investigated. This is a special case which sets up a challenge to these prediction methods. The shape of the tunnel and therefore focusing the sound, reverberant wall, and therefore fluctuation effect demands high accuracy in predicting the early reflections. The energy dissipates quickly in this type of enclosures and there is little masking effect of the reverberation. Another aspect that has been shown to give very different results in this case study is the geometrical detailing of the models. When the Istanbul Tunnel is compared with modern metro systems; nowadays railway systems are controlled by modern electronic and mechanical system, but in the 19th century these electronic systems were not available. A solution could be found with the shape of the building. The aim of this paper is to clarify some of the problems that can arise in this type of constructions, and give guidelines for how they can be overcome / avoided. Another objective is to emphasize that room acoustic computer simulations although very useful, need careful consideration about the underlying calculation methods.

**4pAAf2. The Royal Church of San Lorenzo in Turin: Guarino Guarini and the Baroque architectural acoustics.** Marco Caniato (University of Trieste, Piazzale Europa, 34100 Trieste, Italy, caniatomarco@vodafone.it), Federica Bettarello (Engineering Dept. - Univ. of Ferrara, Via Saragat, 1, 44100 Ferrara, Italy, federica.bettarello@unife.it), Marco Masoero (Politecnico di Torino, Corso Duca degli Abruzzi 24, 10123 Torino, Italy, marco.masoero@polito.it)

In 1666 the architect Guarino Guarini received from Carlo Emanuele II, Duke of Savoy, the appointment to build in Turin a new church dedicated to S. Lorenzo. The architect conceived a design in Baroque style with a very particular ribbed dome and this peculiarity is a very hard to find feature throughout Europe. Acoustics measurements were performed in S. Lorenzo in order to investigate how this unique architecture affects the response parameters used in architectural acoustics. Results are discussed in the paper, comparing to the methodology suggested by Cirillo and Martellotta in order to characterize the acoustics of churches.

**4pAAf3. Bernardo Antonio Vittone: acoustics and architecture in the XVIII century.** Marco Caniato (University of Trieste, Piazzale Europa, 34100 Trieste, Italy, caniatomarco@vodafone.it), Vilma Fasoli (University of Trieste, Piazzale Europa, 34100 Trieste, Italy, fasoli@tin.it)

Contemporary critics have identified Savoy architect Bernardo Vittone as a major contributor to the establishment of the concept and experience of architecture as the "art of construction" in the 18th century. Vittone grew and worked under the guidance of mathematician Guarino Guarini, whose mastery of theoretical and scientific aspects he was never credited to have matched. Vittone's approach to architecture was mainly practical, his plans chiefly derived from the rules of mechanics and descriptive geometry. Experience was his main source of inspiration. Designs, plans, domes and vaults are but ways to achieve balance and harmony and so are precision and control of the relation between geometry and acoustics, architecture and music. In his book "Alternative instructions on the civil Architecture" moreover he included in the second appendix the "Armonic instructions", within a dissertation on the nature of sound, its propagation in open spaces, in closed ones and its relation with music. In this paper a commented overview is given matching his description of acoustics with present-day knowledge.

**4pAAf4. The engineering method of sound field control for amplifications in sports arena.** Jian Peng (Guangzhou Radio, 231 Huan-shi Zhong Road, 510000 Guangzhou, China, zp24@drexel.edu), Zhao Peng (Drexel Univ., 3175 JFK Blvd, APT 707, Philadelphia, PA 19104, USA, zp24@drexel.edu)

This paper discusses the engineering method of sound field control for amplifications in sports arena, which includes the selection, assembly and adjustment of the loudspeaker system. It also includes the site characteristics of the sports arena, as well as the relations between the loudspeaker and the site. Through the measurements of the rated sensitivity and maximum noise power, the loudspeaker system can be determined as distributed or central system, linear or plane array. With the adjustments on parameters such as the directivity  $D(\theta)$  and the directivity factor  $Q$ , the loudspeaker system will be able to effectively control the sound field during amplification process and to achieve better sound quality in sports arena. Besides determining the effective engineering method for sound field control in the amplification process, this paper also considers the acoustical characteristics of sound sources and coupling effect in the sound field.

**4pAAf5. Correlation between architectural and acoustical parameters in Catholic churches of the colonial era in Ayacucho, Peru.** Carlos R. Jimenez (Pontificia Universidad Catolica Del Peru, Av. Universitaria 1801, San Miguel, 32 Lima, Peru, [cjimene@pucep.edu.pe](mailto:cjimene@pucep.edu.pe))

The paper reports the analysis of the results of an acoustic survey carried out on several Catholic Churches in the city of Ayacucho, a southern city in the Peruvian Andes, built during the colonial era through centuries XVI to XVIII. The study is performed taking into account both room-average values and individual position values of many acoustical parameters measured according to ISO 3382. Effects among architectural characteristics of each church as length, height, room volume, interior surfaces area, total absorbing area, mean absorption coefficient, and some objective acoustical parameters as reverberation time, early decay time, clarity, definition and center time, are investigated.

**4pAAf6. Simulation of the reverberation time of an existing architectural space using a 1:10 scale model.** Luz Del Carmen Gonzalez-Rodriguez (Program of Graduate Studies in Design, CyAD - Universidad Autonoma Metropolitana, Av. San Pablo 180, Edificio D 101, Col. Reynosa Tamaulipas, Delegacion Azcapotzalco, 02200 Mexico, D.F., Mexico, [loozie@gmail.com](mailto:loozie@gmail.com)), Fausto E. Rodriguez-Manzo (Departamento de Procesos y Tecnicas, CyAD, Universidad Autonoma Metropolitana-Azcapotzalco, Av. San Pablo 180, Edificio H-PB, Col. Reynosa Tamaulipas, Delegacion Azcapotzalco, 02200 Mexico, D.F., Mexico, [rfme@correo.azc.uam.mx](mailto:rfme@correo.azc.uam.mx))

Recently the science of acoustics has been reconsidered and valued more than just a tool concerning physics, telecommunications, and music. It is now an important topic of research for the acoustic comfort of places where people work and live, and it is also considered as an essential issue for the physical and mental health of human beings. Sound has to be integrated as a design concept in architecture, but architects need tools for understanding more the way to reach not only good acoustics but also good architectural design. It has been proven that physical scale models are very useful for the prediction of the acoustical behaviour of rooms, therefore this process is analyzed and studied at the Acoustic Design and Analysis Laboratory of the UAM-Azcapotzalco in Mexico City, in order to promote further investigations that provide useful data for the design of architectural spaces as well as the design of sound control devices. This paper shows the work of a master degree thesis where the main objective is to present the measurements of reverberation time taken in an existing room compared with those taken in the physical scale model of the same room.

**4pAAf7. Architectural Acoustics Design using Diffusing Surfaces.** James Heddle (James Heddle Pty Ltd, Unit 2315 / 180 Grey Street, South Bank, 4101 Brisbane, Australia, [acousticdesign@gmail.com](mailto:acousticdesign@gmail.com))

This paper reviews the design challenges and development of acoustic treatments for three acoustical design projects in Australia. The projects that will be outlined were a surround sound widescreen lecture theatre, a contemporary council chambers and the rectification of a problematic brass band practice room. The treatments included sound diffusing surfaces of the geometric type with or without embedded amplitude reflection gratings. Where the gratings were used in multiples the designs also addressed periodicity issues using modulation of the grating patterns with the best known merit factors. The design approach and the development of the acoustical surface features and treatments to solve the design issues, including problems experienced in the building phase, will be discussed.

**4pAAf8. Evaluation of existing sound system designs in mosques and alternative modern solutions.** Wasim Orfali (Nordmannzeile 12, 12157 Berlin, Germany, [orfaliwasim@hotmail.com](mailto:orfaliwasim@hotmail.com)), Wolfgang Ahnert (Ahnert Feistel Media Group, Arkonastr. 45-49, 13189 Berlin, Germany, [wahnert@ada-acousticdesign.de](mailto:wahnert@ada-acousticdesign.de))

Sound intelligibly and audibility was from the old days and will remain a central issue in any sound system design especially in mosques. In other worship spaces sound system may not be needed during specific worshiping modes like congregational prayers. In mosques all worshiping modes require professionally designed sound system to maintain the needed intelligibility and audibility levels in such spaces. Here the current used sound systems in mosques will be discussed and pro and cons will be highlighted. Also, the effect of these systems on mosques applicable sound parameter will be introduced. In the same time, modern sound system design solution for different mosques architectural form and size will be suggested. Case study examples will be used to show the advantages of the new suggested sound system configurations.

**4pAAf9. Room acoustics enhancement system in a Multifunctional Cultural Centre.** Géza Balogh, Jr. (Interton Electroacoustics, Major u. 63, H-1119 Budapest, Hungary, [balogh@interton.hu](mailto:balogh@interton.hu)), Géza Balogh, Dr. (Interton Electroacoustics, Major u. 63, H-1119 Budapest, Hungary, [drbalogh@interton.hu](mailto:drbalogh@interton.hu))

In the lecture, we present our system called DCR (Digital Control of Reverberation), which is capable of changing the reverberation of a room by means of electroacoustical devices. Using the DCR system, we are able to change the enclosure, i.e. we create an electroacoustical enclosure on the walls and on the ceiling. The parameters of the electroacoustical enclosure can be changed in several steps. The system was installed in Hungary for the first time in the city of Debrecen, in the Kölcsey Cultural Centre. The principle of the DCR and the problems and their solutions will be presented via this specific installation. We study the effect of the distance between the microphones and the loudspeakers, and the factors that influence this effect. We also analyse the room acoustics parameters in respect to the frequency, in the different status of the auditorium. Our room acoustics enhancement DCR system is based on the non-inline principle, thus the room acoustics of the large auditorium (1100 seats) of the Kölcsey Centre remained absolutely natural. We use several numbers of independent digital channels in the system, and each channel consists of one microphone, one or two loudspeakers, a power amplifier and a DSP.

**4pAAf10. Translating acoustics into architecture.** Ewart A. Wetherill (Ewart A. Wetherill, AIA, 28 Cove Road, Alameda, CA 94502-7416, USA, [redwetherill@sbcglobal.net](mailto:redwetherill@sbcglobal.net))

Ever since Wallace Clement Sabine's first presentation to the American Institute of Architects in 1898, a substantial gap has often existed between available information on building acoustics and its successful application to the design of buildings. Experience over at least the past 40 years suggests that one of the reasons for this has been the difficulty of adapting construction methods, that were selected to meet other criteria, to satisfy specific acoustical conditions. In a complex project it is important that requirements for individual spaces be identified and resolved early in the design process so that they can be integrated successfully into the overall building design. This paper discusses a procedure that has been found very convenient for establishing acoustical requirements during the schematic design and transferring recommendations efficiently for inclusion in the construction documents. The information is compiled in a book-style format for ease of distribution and for fast reference at any time in design and construction.

## Session 4pAag

## Architectural Acoustics: Prediction Methods in Building Acoustics III (Poster Session)

Berndt Zeitler, Cochair

NRC - Institute for Research in Construction

Catherine Guigou-Carter, Cochair

CSTB

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pAag1. Determination of unknown parameters in impervious layers by inverse method.** Eva Escuder Silla (Escola Politècnica Superior de Gandia, Universitat Politècnica de València, Crtra Natzalet-Oliva s/n, 46730 Gandia, Spain, evescude@fis.upv.es), Jesús Alba Fernandez (Escola Politècnica Superior de Gandia, Universitat Politècnica de València, Crtra Natzalet-Oliva s/n, 46730 Gandia, Spain, jesalba@fis.upv.es), Jaime Ramis Soriano (DFISTS. Univ. de Alicante, Carretera de Sant Vicent del Raspeig s/n, 03690 San Vicente del Raspeig, Spain, jramis@ua.es), Romina Del Rey Tormos (Escola Politècnica Superior de Gandia, Universitat Politècnica de València, Crtra Natzalet-Oliva s/n, 46730 Gandia, Spain, roderey@doctor.upv.es)

In this work, a novel procedure is shown for the determination of unknown parameters in impervious layers used in multilayer structures by inverse method and using scale models. Experimental pressure and velocity data are obtained by Nearfield Acoustic Holography (NAH) for the calculation of the transmission loss of the different multilayer structures mounted on the window of a wooden box designed for that end. These data are used as input data in the inverse method. The forecast model of acoustic insulation in multilayered structures used in this work was Trochidis&Kalaroutis model based on Spatial Fourier Transform (SFT). By applying Trochidis&Kalaroutis model and adjusting by numerical methods the variables that define the impervious layers of the system, the values of the unknown magnitudes of the layers are calculated. For validation purposes the results are compared to those obtained with Ookura&Saito model.

**4pAag2. New Roadway noise modeling to predict noise propagation in front of urban façade.** Hany Hossam Eldien (Suez Canal Univ., 25500 Suez, Egypt, hany.hossam@gmail.com), Philippe Woloszyn (UMR ESO, Université de Haute Bretagne - Rennes II, Place du Recteur Henri Le Moal, CS 24307, 35043 Rennes Cedex, France, philippe.woloszyn@univ-rennes2.fr)

Roadway noise is the most important example of a linear noise source, since it comprises about 80 percent of the environmental noise exposure for humans worldwide. Due to the complexity of the variables, a line source acoustic model is realized by a computer model. In this paper we propose a new approach system based upon Schröder's quadratic residue diffuser modeling for modeling a linear traffic source. 1/10th scale model has been used to simulate the traffic noise propagation in front of the building façade and for a free field. The line source directivity is made quasi-uniform in the aperture angle of interest. The line source directivity measurements were made using a 1/4" microphone Larson Davis type 2530. The sequence of the quadratic residue diffusor is calculated following the Schröder's modulo for-

mula and with 1.60m length, 0.14m height, and 0.11m width. The line source has also been weighted by a full-size equivalent normalised traffic noise spectrum according to the French standard NF EN 1793-3.

**4pAag3. Influence of the conservation state of the facade elements as a variable of prediction of acoustic insulation to airborne sound.** Leonardo Meza (Pontificia Universidad Católica de Chile, Av. Vicuña Mackena 4860, Macul, 7820436 Santiago, Chile, lmezam@uc.cl)

The different prediction methods of acoustic insulation are based on the acoustic properties of the constituent materials of the buildings, however, they don't usually consider the conservation state as a variable within the prediction, ignoring the natural process of degradation that the material and the construction elements suffer which sometimes are increased by the deterioration due to the use of itself. At this paper, the behavior of some construction elements is analyzed by means of acoustic insulation measurements in-situ, such as windows and its woodwork, which presents in many cases a diminish in the acoustic performance due to the damage and deterioration. Finally, the incorporation of this variable is proposed in future prediction models of acoustic insulation.

**4pAag4. Variable Source-Directivity Using Dodecahedron-Loudspeakers.** Gottfried K. Behler (RWTH Aachen University, Templergraben 55, D-52056 Aachen, Germany, gkb@akustik.rwth-aachen.de), Martin Pollow (RWTH Aachen University, Templergraben 55, D-52056 Aachen, Germany, martin.pollow@akustik.rwth-aachen.de)

For room-acoustical measurements dodecahedron loudspeakers are commonplace to achieve a uniform directivity. Therefore all transducers are fed with the same signal. If the signals for the twelve transducers are individually adjustable, the variation of amplitude and phase offers the possibility to achieve a predefined directivity. The goal is to calculate the twelve frequency dependent amplitude- and phase-coefficients for any given directivity with the least possible error. A simple approach like superposition unfortunately does not reveal a correct result, since all transducer interact with each other. The decomposition of spherical functions into spherical harmonics, however, leads to an analytic solution for the prediction of the sound radiation. The acoustical components - like sound pressure and sound velocity - are split up into weighted, orthogonal base functions which can be combined in a way that the mutual coupling between different membrane vibrations is respected. Under these conditions complex filter transfer functions, individually optimized for each one of the twelve transducers can be

computed with respect to the target directivity function. Different approaches for the optimization procedure are discussed. Besides the more theoretical approach measurements with a real multichannel system are shown.

**4pAag5. In situ evaluation of the vibration reduction index Kij.** Arianna Astolfi (Politecnico di Torino, Department of Energetics, Corso Duca degli Abruzzi, 24, 10129 Turin, Italy, arianna.astolfi@polito.it), Alessandro Schiavi (Istituto Nazionale di Ricerca Metrologica, str. delle Cacce, 91, 10135 Turin, Italy, a.schiavi@inrim.it), Cristiana Taricco (Via Tapparelli 31, 12038 Savigliano (CN), Italy, c.taricco@fastwebnet.it), Simone Geroso (Politecnico di Torino, Department of Energetics, Corso Duca degli Abruzzi, 24, 10129 Turin, Italy, geroso@alice.it), Fabrizio Bronuzzi (Politecnico di Torino, Department of Energetics, Corso Duca degli Abruzzi, 24, 10129 Turin, Italy, fabrizio.bronuzzi@polito.it), Andrea Savio (Via Madonnina, 12, 10090 Gassino Torinese (TO), Italy, andrea.savio\_84@alice.it)

As stated in EN 12354-1 and 2 Standards the vibration reduction index Kij is a quantity related to the vibrational power transmission over a junction between structural elements, normalized in order to make it invariant. This quantity allows to quantify the flanking transmission both in air-borne and impact sound insulation between rooms. It is important to underline that the measurement method reported in the EN 10848-1 Standard is only referred to the laboratory measurement, anyway in the Annex E of the EN 12354-1 Standard is indicated that the same methodology is probably useable also in in situ condition. In this work several measurements of the vibration reduction index in situ are reported in order to verify if the methodology is also suitable in uncontrolled conditions as in the field. The measured results are compared with the previsional model results according to the EN 12354-1. In particular the case studies are the T-junctions and the cross-junctions between the ceilings and the vertical walls in some typical rooms in dwellings. From the first experimental results some peculiar building fea-

tures, influencing the measurement and the calculation data, have been evidenced, especially with reference to the south European building typologies in brick and concrete.

**4pAag6. High sound pressure models for microperforated panels backed by an air cavity.** Rostand Tayong (Lab. de Recherche en Mécanique et Acoustique, I.S.A.T - Univ. de Bourgogne, 49, rue Mademoiselle Bourgeois, 58000 Nevers, France, rostand.tayong@u-bourgogne.fr), Thomas Dupont (Lab. de Recherche en Mécanique et Acoustique, I.S.A.T - Univ. de Bourgogne, 49, rue Mademoiselle Bourgeois, 58000 Nevers, France, thomas.dupont@u-bourgogne.fr), Marie-Annick Galland (Centre Acoustique du LMFA, Ecole Centrale de Lyon, 36 avenue Guy de Colongue, 69134 Ecully cedex, France, marie-annick.galland@ec-lyon.fr), Philippe Leclaire (Lab. de Recherche en Mécanique et Acoustique, I.S.A.T - Univ. de Bourgogne, 49, rue Mademoiselle Bourgeois, 58000 Nevers, France, philippe.leclaire@gmail.com)

When submitted to relatively high sound pressure amplitudes, Micro Perforated Panels (MPP) are influenced by certain effects, which are non negligible (vibration of the panel, end radiation and also proximity of the perforations). A model of the total impedance of the MPP is derived from the sum of the contributions of each effect in the case of relatively high sound pressure. The effect of end radiation is supposed to be independent of the propagation inside the apertures. The model is applicable for low Mach numbers. In order to validate the models, various steel MPP specimens were built with different aperture diameters, interstices (distance between two near apertures) and thickness sizes. The experimental method consists in measuring the acoustical pressure before the specimen and the velocity at the aperture entrance. The experimental setup is based on the use of an impedance circular tube. A loudspeaker capable of delivering high sound pressure is used as a source. The excitation is a white noise in a frequency range between 500 Hz and 5000 Hz and the detection is performed with microphones. The comparison between measurements and simulations for the impedance and absorption coefficient is done and discussed.

THURSDAY AFTERNOON, 3 JULY 2008

P2-B, LEVEL 2, 3:40 TO 5:20 P.M.

## Session 4pAAh

### Architectural Acoustics: Measuring Methods and Uncertainty in Building Acoustics II (Poster Session)

Brandon Tinianow, Cochair

Werner Scholl, Cochair

*Physikalisch-Technische Bundesanstalt*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pAAh1. The examination of the influence of standing waves on reverberation time measurements in small reverberant rooms.** Marko Horvat (Faculty of EE and Computing, Unska 3, Department of Electroacoustics, HR-10000 Zagreb, Croatia, marko.horvat@fer.hr), Kristian Jambrosic (Faculty of EE and Computing, Unska 3, Department of Electroacoustics, HR-10000 Zagreb, Croatia, kristian.jambrosic@fer.hr), Hrvoje Domitrovic (Faculty of EE and Computing, Unska 3, Department of Electroacoustics, HR-10000 Zagreb, Croatia, hrvoje.domitrovic@fer.hr)

The measurements of reverberation time are performed in a small and rather reverberant shoe-box type room in order to determine the spatial distribution of reverberation time with respect to the formation of standing

waves in the room itself. The measurements are made in a two-axes raster with equidistant distribution of measurement points. The goal is to investigate the conclusions made in an earlier work by Graber et al. that the distribution of reverberation time values along the fore mentioned raster reaches maximum values at the points of minimum sound pressure level and vice versa. The room in which the measurements are made is deliberately chosen to be relatively small in order to examine the lowest modes of standing waves that form in the room.

**4pAAh2. Effect of the estimation of the wall/floor junction type on the acoustic isolation.** Jesús Alba Fernandez (Escola Politècnica Superior de



Gandia, Universitat Politècnica de València, Crtra Natzaret-Oliva s/n, 46730 Gandia, Spain, jesalba@fis.upv.es), Jaime Ramis Soriano (DFISTS. Univ. de Alicante, Carretera de Sant Vicent del Raspeig s/n, 03690 San Vicente del Raspeig, Spain, jramis@ua.es), Eva Escuder Silla (Escola Politècnica Superior de Gandia, Universitat Politècnica de València, Crtra Natzaret-Oliva s/n, 46730 Gandia, Spain, evescude@fis.upv.es), Romina Del Rey Tormos (Escola Politècnica Superior de Gandia, Universitat Politècnica de València, Crtra Natzaret-Oliva s/n, 46730 Gandia, Spain, roderrey@doctor.upv.es)

Deviations between the laboratory measurement and the final in situ measurement, are common in the acoustic isolation. A fundamental difference between both cases is the effect of the lateral flanks. This effect produce that the global acoustic insulation changes substantially. With the standard UNE-EN-12354-1, it can be estimated the effect of the flank, obtaining the vibration reduction indexes and evaluating the insulation of every way. It is also possible to carried out a measurement procedure with the Standard ISO 10848, in it the index is measured and then the index is calculated. This paper shows the differences of results of airborne sound insulation and impact insulation with in situ measurements of the vibration reduction index and the different estimations that the Standard allow. Different unions are studied, and the influence of these in the global result of the insulation is evaluated.

**4pAAh3. Study in the measurement of noise air insulation in laboratory of the effect in the diffuse field.** Romina Del Rey Tormos (Escola Politècnica Superior de Gandia, Universitat Politècnica de València, Crtra Natzaret-Oliva s/n, 46730 Gandia, Spain, roderrey@doctor.upv.es), Jesús Alba Fernandez (Escola Politècnica Superior de Gandia, Universitat Politècnica de València, Crtra Natzaret-Oliva s/n, 46730 Gandia, Spain, jesalba@fis.upv.es), Jaime Ramis Soriano (DFISTS. Univ. de Alicante, Carretera de Sant Vicent del Raspeig s/n, 03690 San Vicente del Raspeig, Spain, jramis@ua.es), Eva Escuder Silla (Escola Politècnica Superior de Gandia, Universitat Politècnica de València, Crtra Natzaret-Oliva s/n, 46730 Gandia, Spain, evescude@fis.upv.es)

We can obtain, in a transmission chamber, the air transmitted noise insulation. In the standard 140-1 the characteristics of these chambers are described. One of these characteristics tries to ensure diffuse field inside these chambers. Nevertheless, we cannot assure an incident angle between  $0^\circ$  and  $90^\circ$  on the test wall, there is a limit angle lower than  $90^\circ$ . In this paper, we study the valuation of the committed mistake by limit angle in transmission chambers. Expressions used for calculating the insulation in transmission chambers are obtained from transmission description in diffuse field, in which case the angle is  $90^\circ$ . In this work, for different materials commonly used in transmission chambers tests, the influence of the indetermination of a mistake and the global mistake evolution with limit angle is studied.

**4pAAh4. Student project of building an impedance tube.** Mia Suhanek (Faculty of EE and Computing, Unska 3, Department of Electroacoustics, HR-10000 Zagreb, Croatia, mia.suhanek@fer.hr), Kristian Jambrosic (Faculty of EE and Computing, Unska 3, Department of Electroacoustics, HR-10000 Zagreb, Croatia, kristian.jambrosic@fer.hr), Hrvoje Domitrovic (Faculty of EE and Computing, Unska 3, Department of Electroacoustics, HR-10000 Zagreb, Croatia, hrvoje.domitrovic@fer.hr)

This paper describes a student project of building an impedance tube for measuring the absorption coefficient using the transfer-function method, in accordance with the standard ISO 10534-2. This method is well-established and has many advantages compared to the older method using standing wave ratio (ISO 10354-1) in terms of measurement speed and accuracy. For the tube, only inexpensive materials and transducers were used. The tube was designed for the frequency range between 90 and 2000 Hz. In order to achieve this range with one tube, three microphone positions have been used. The resulting absorption coefficient has been calculated using the one- and two-microphone method. Different broadband excitation signals have

been used in order to compare their robustness, such as MLS, frequency sweep and white noise. Various problems with the design and construction are addressed and the optimal configuration is discussed.

**4pAAh5. Uncertainty of airborne sound insulation index measurement in laboratory conditions.** Tadeusz Wszolek (University of Science and Technology, Department of Mechanics and Vibroacoustics, Al.Mickiewicza 30, 30-059 Krakow, Poland, twszolek@agh.edu.pl)

In buildings, airborne sound insulation is used to define the acoustic quality of walls between rooms. However the evaluation of sound insulation is sometimes difficult or even ambiguous, both in field and laboratory measurements, in spite of the fact that there are some unified measurement procedures specified in the ISO 140 standards. There are problems with the reproducibility and repeatability of the measured results. Some difficulties may be caused by non-diffuse acoustic fields, non uniform reverberation time or large spread of the reverberation time measurements especially in low frequency band. Some minor problems are also posed by flanking transmission and the S/N ratio. In the present work partial uncertainty analysis has been carried out for all the above mentioned factors and their influence has been evaluated on the combined uncertainty in 1/3 octave bands and the  $R_w$  index, using the uncertainty propagation law

**4pAAh6. Blind estimation method of reverberation time based on concept of modulation transfer function.** Masashi Unoki (JAIST, 1-1 Asahidai, 923-1292 Nomi, Japan, unoki@jaist.ac.jp), Sota Hiramatsu (JAIST, 1-1 Asahidai, 923-1292 Nomi, Japan, s0610073@jaist.ac.jp)

This paper proposes a method for blindly estimating the reverberation time based on the concept of the modulation transfer function (MTF). This method estimates the reverberation time from the reverberant signal without measuring room acoustics. In the MTF-based speech dereverberation method, proposed by the authors, a process for estimating a parameter related to the reverberation time was incorporated. In this paper, we investigate whether the estimation process, previously presented by authors, works as a blind estimation method and point out a problem with their method. We then propose a new method for blindly estimating the reverberation time to resolve the problem. In the proposed method, the reverberation time is correctly estimated by inverse-MTF filtering in the modulation frequency domain. We evaluated the proposed method with their method using both artificial MTF-based signals and speech signals to show how well the proposed method correctly estimates the reverberation time in artificial reverberant environments. Results suggested that the proposed method correctly estimates reverberation times from the observed reverberant signals. [Work supported by a Grant-in-Aid for Science Research from the Japanese Ministry of Education (No. 18680017).]

**4pAAh7. Realization of a measurement system for physical and acoustic measurements on brick walls.** Luca Barbaresi (DIENCA - Univ. of Bologna, Viale Risorgimento 2, 40136 Bologna, Italy, luca.barbaresi@mail.ing.unibo.it), Massimo Garai (DIENCA - Univ. of Bologna, Viale Risorgimento 2, 40136 Bologna, Italy, massimo.garai@mail.ing.unibo.it), Paolo Guidorzi (DIENCA - Univ. of Bologna, Viale Risorgimento 2, 40136 Bologna, Italy, paolo.guidorzi@mail.ing.unibo.it), Giovanni Semprini (University, DIENCA Dept. Facoltà di Ingegneria, Viale Risorgimento 2, 40136 Bologna, Italy, giovanni.semprini@mail.ing.unibo.it)

Aim of this study is the development of a system for the measurement of frequency values of some physical variables such as the damping factor, the thickness of a structural element, the structural reverberation time and the longitudinal wave speed propagation in light brick walls. Such variables are required by the UNI EN 12354-1 normative for the analytical estimation of the sound reduction index  $R$  of monolithic elements in the laboratory. The results of the calculation, as a function of frequency, will be shown and compared with the measured values of the sound reduction index  $R$ .

**4pAAh8. Swept Sine against MLS in room acoustics with music signals as background noise.** Joel P. Paulo (ISEL, R. Conselheiro Emídio Navarro, 1, 1959-007 Lisbon, Portugal, jpaulo@deetc.isel.ipl.pt), J. Luis Bento Coelho (CAPS, Instituto Superior Técnico, TU Lisbon, Av. Rovisco Pais, P-1049-001 Lisbon, Portugal, bcoelho@ist.utl.pt)

The Swept Sine and the MLS techniques are very popular in room acoustic measurement set-ups. Advantages and disadvantages of both methods have been well investigated and can be found in the literature. However, information regarding the performance of these techniques in the presence of high background music levels is scarce. Since the estimation of the room impulse response is based on the correlation between signals, the likelihood between the test signal and the music contents has an important role on the results accuracy. This paper explores these issues by taking into account the semantic information of the music signals when used as disturbance. The method used for the assessment of the gain between the two techniques consists on splitting each frame in segments and applying a weighting function depending on a likelihood function. The features used for the likelihood function are the rms value of each segment, spectral energy envelope relation, bandwidth and harmonic structure. Several examples are presented for comparison of the performance of the Swept Sine and the MLS techniques. Advantages and disadvantages of each technique are discussed for music signals as noise.

**4pAAh9. Effects of filtering of room impulse responses on room acoustics parameters by using different filter structures.** Csaba Huszty (Budapest University of Technology and Economics, BME Dept. of Telecommunications, Magyar tudósok körútja 2, H-1117 Budapest, Hungary, huszty@hit.bme.hu), Norbert Bukuli (Budapest University of Technology and Economics, BME Dept. of Telecommunications, Magyar tudósok körútja 2, H-1117 Budapest, Hungary, bukuli.norbert@gmail.com), Ákos Torma (Budapest University of Technology and Economics, BME Dept. of Telecommunications, Magyar tudósok körútja 2, H-1117 Budapest, Hungary, tormakos@gmail.com), Fülöp Augusztinovicz (Budapest University of Technology and Economics, BME Dept. of Telecommunications, Magyar tudósok körútja 2, H-1117 Budapest, Hungary, fulop@hit.bme.hu)

Room acoustic evaluation is usually based on post-processing of measured room impulse responses (RIRs), and this often requires some kind of filtering, for instance to derive fractional octave band parameters of a room. In this paper it is shown that the considerable variance of room acoustic parameters of almost any hall is partly caused by the filtering method and the filter properties used in the course of post-processing. The paper proposes new qualification methods and parameters for determining the quality of FIR filter banks, taking their use for acoustic evaluation into account. It suggests practical considerations for the design as well, and shows the analysis and comparison of effects of various filter properties -- such as filter types and topology structures -- on some room acoustics parameters. By using the suggested methods, it is possible to derive more accurate and reliable results in room acoustic evaluation.

**4pAAh10. Reverberation time measuring methods.** Kristian Jambrosic (Faculty of EE and Computing, Unska 3, Department of Electroacoustics, HR-10000 Zagreb, Croatia, kristian.jambrosic@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)

In this paper different well-established methods of reverberation time measurement are compared. Furthermore, the results obtained using these methods are compared to the results provided by some additional methods which could serve as an in situ tool if, for any reason, the reverberation time measurements cannot be carried out using the standardized methods. The methods compared in this paper include the standardized methods (EN ISO 3382:2000), namely the impulse response measured with pink noise, exponential sweep, MLS, but also pistol shots of different calibers, balloon bursts, gated external pink noise, and the B&K filtered burst method. In order to make the comparison, the measurements were performed in four acoustically very different spaces - a rather small and well-damped listening room, a much bigger damped listening room, a rather reverberant atrium, and a large and very reverberant shoebox-shaped room. The results were evaluated according to signal-to-noise ratio criterion as well. Special attention has been given to the influence of room modes on measurement results.

THURSDAY AFTERNOON, 3 JULY 2008

P2-B, LEVEL 2, 3:40 TO 6:40 P.M.

## Session 4pAAi

### Architectural Acoustics and the National Council of Acoustical Consultants: Student Design Competition

Byron W. Harrison, Cochair

*The Talaske Group Inc., 1033 South Boulevard, Oak Park, IL 60302, USA*

Anders Christian Gade, Cochair

*Dept. of Acoustic Technology, Technical University of Denmark, Building 352, DK 2800 Lyngby, Denmark*

The Technical Committee on Architectural Acoustics of the Acoustical Society of America, the National Council of Acoustical Consultants, and the Robert Bradford Newman Student Award Fund are sponsoring the 2008 Student Design Competition that will be professionally judged at this meeting. The purpose of this design competition is to encourage students enrolled in architecture, engineering, physics, and other university curriculums that involve building design and/or acoustics to express their knowledge of architectural acoustics in the design of a new theater performance space as part of a university building project.

This competition is open to undergraduate and graduate students from all nations. Submissions will be poster presentations that demonstrate room acoustics, noise control, and acoustic isolation techniques in building planning and room design. The submitted designs will be displayed in this session and they will be judged by a panel of professional architects and acoustical consultants. An award of USD\$1250.00 will be made to the entry judged "First Honors." Four awards of USD\$700.00 will be made to each of four entries judged "Commendation."

## Session 4pABb

**Animal Bioacoustics, Psychological and Physiological Acoustics, and ECUA: Auditory Brainstem Response and Behavior Correlation III (Poster Session)**

Elizabeth Brittan-Powell, Cochair  
*University of Maryland*

Alexander Supin, Cochair  
*Institute of Ecology and Evolution*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

*Contributed Papers*

**4pABb1. Intense sonar pings induce temporary threshold shift in a bottlenose dolphin (*Tursiops truncatus*).** T Aran Mooney (University of Hawaii, Hawaii Institute of Marine Biology, P.O. Box 1106, Kailua, HI 96734, USA, mooneyt@hawaii.edu), Paul E. Nachtigall (University of Hawaii, Hawaii Institute of Marine Biology, P.O. Box 1106, Kailua, HI 96734, USA, nachtiga@hawaii.edu), Stephanie Vlachos (University of Hawaii, Hawaii Institute of Marine Biology, P.O. Box 1106, Kailua, HI 96734, USA, vlachos@hawaii.edu)

For over a decade it has been suggested that high intensity anthropogenic sounds, such as sonar, could induce a temporary threshold shift (TTS) in odontocetes. Although TTS has been examined in marine mammals, the temporary physiological effects of sonar sounds have yet to be established. This study explored the effects of high-intensity (up to 203 dB re: 1  $\mu$ Pa), mid-frequency sweeps (2-4 kHz) on the hearing of a bottlenose dolphin (*Tursiops truncatus*). The goal was to determine if these sounds could induce TTS and what sound exposure levels (SEL; dB re: 1  $\mu$ Pa<sup>2</sup>s) were necessary for TTS to be induced. Fatiguing sounds were presented to mimic that of mid-frequency sonar. Hearing thresholds were measured before and after exposures using auditory evoked potentials to determine amount of shift and rate of recovery. Temporary threshold shifts of 5-6 dB were measured using SELs of 214 dB, in situations when 15 sonar pings were presented in series. Recovery to normal hearing was rapid, typically within 5 to 10 min. Exposure levels required to induce TTS were high, supporting the notion that

relatively short sounds must be of relatively high intensity to induce threshold shifts.

**4pABb2. The auditory time resolution in bottlenose dolphins: Behavioral experiments versus auditory evoked potential methods.** Gennadi L. Zaslavski (University Authority for Applied Research, RAMOT, Tel-Aviv University, str. Gordon 51 app 7, 42442 Netanya, Israel, gennadi.zaslavski@gmail.com)

Non-invasive auditory evoked potentials (AEP) methods are now widely used to study dolphins' hearing because some auditory characteristics can be obtained much faster compare to behavioral methods. The bottlenose dolphin auditory time resolution assessed using evoked potentials responses to a double click, amplitude modulated tone and periodic click is generally believed to be around 300 microseconds. This assessment is claimed to be in full agreement with behavioral measurements. The intention of this paper is to reevaluate behavioral results which are believed to support AEP methods in light of numerous behavioral results indicative of the bottlenose dolphin time resolution as high as 20-30 microseconds. We found that as long as there are differences in waveforms, bottlenose dolphins are able to discriminate between very short (as short as a bottlenose dolphin sonar click) signals with identical energy spectra as well as between brief noise signals with random energy spectra. Auditory evoked responses do not reveal any differences between such signals whereas the differences are readily indicated by behavioral responses of the dolphins. The auditory temporal analysis of brief signals in bottlenose dolphins seems to be inaccessible by AEP methods, at least in their present form.

## Session 4pABc

## Animal Bioacoustics and ECUA: Odontocete Acoustics II (Poster Session)

David Mellinger, Cochair  
Oregon State Univ. and NOAA

Michel Andre, Cochair  
*Laboratori d'Aplicacions Bioacústiques (Universitat Politècnica de Catalunya)*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pABc1. Decalcifying Protocol of Odontocete Ear Samples with RDO®.**

Maria Morell (Laboratori d'Aplicacions Bioacústiques (Universitat Politècnica de Catalunya), avda. Rambla Exposició s/n, 08800 Vilanova i la Geltrú, Spain, maria.morell@lab.upc.edu), Eduard Degollada (Laboratori d'Aplicacions Bioacústiques (Universitat Politècnica de Catalunya), avda. Rambla Exposició s/n, 08800 Vilanova i la Geltrú, Spain, edmaktub@edmaktub.com), Josep Maria Alonso (Laboratori d'Aplicacions Bioacústiques (Universitat Politècnica de Catalunya), avda. Rambla Exposició s/n, 08800 Vilanova i la Geltrú, Spain, jmalonso@iim.csic.es), Thierry Jauniaux (Université de Liège, Place du 20-Aout, 9, B-4000 Liège, Belgium, T.Jauniaux@ulg.ac.be), Mardik Leopold (Imares, P.O. Box 167, 1790 AD Den Burg, Texel, Netherlands, Mardik.Leopold@wur.nl), Kees Camphuysen (Royal NIOZ (Netherlands Inst. for Sea Research), 1790 AB Den Burg, Texel, Netherlands, camphuys@nioz.nl), Michel Andre (Laboratori d'Aplicacions Bioacústiques, Universitat Politècnica de Catalunya, avda. Rambla Exposició s/n, 08800 Vilanova i la Geltrú, Spain, michel.andre@upc.edu)

The study of the organ of Corti is essential to assess the impact of underwater noise on cetaceans. While classical histology techniques (including EDTA decalcification) have been previously considered, the process is time consuming and artifacts, probably directly deriving from the protocol, often appear and difficult the analysis. However, no matter the choice of the analysis technique, one of the challenging step after extraction and fixation of the samples is to decalcify the bone envelope to access the cochlea without damaging the soft tissues. Here, we propose to use a fast commercial decalcifier (RDO®). 103 ears from 12 different odontocetes species stranded in the Mediterranean, Spanish North Atlantic and North Sea have been used to precisely determine the decalcification time. Specifically, the species processed were: *Phocoena phocoena* (n=45), *Stenella coeruleoalba* (n=12), *Stenella frontalis* (n=12), *Tursiops truncatus* (n=13), *Delphinus delphis* (n=4), *Kogia simus* (n=3), *Kogia breviceps* (n=2), *Globicephala macrohynchus* (n=3), *Globicephala melas* (n=2), *Steno bredanensis* (n=4), *La-*

*genodelphis hosei* (n=2) and *Ziphius cavirostris* (n=1). Depending on the species, the decalcification time ranged from several hours to a few days, allowing a subsequently faster observation of the cochlear structures through routine microscope techniques.

**4pABc2. Mesoplodon densirostris transmission beam pattern estimated from passive acoustic bottom mounted hydrophones and a DTag recording.**

Jessica Ward (NAVSEA, Newport Undersea Warfare Center, Newport, RI RI 02841, USA, wardja@npt.nuwc.navy.mil), David Moretti (NAVSEA, Newport Undersea Warfare Center, Newport, RI RI 02841, USA, MorettiDJ@npt.nuwc.navy.mil), Ronald P. Morrissey (Naval Undersea Warfare Center Division Newport, 1176 Howell Street, Bldg 1351, 2nd Floor, Newport, RI 02841, USA, morrisseyrp@npt.nuwc.navy.mil), Nancy A. Dimarzio (Naval Undersea Warfare Center Division Newport, 1176 Howell Street, Bldg 1351, 2nd Floor, Newport, RI 02841, USA, dimarziona@npt.nuwc.navy.mil), Peter Tyack (Woods Hole Oceanographic Institution, Applied Ocean Physics & Engineering Dept., Woods Hole, MA 02543, USA, ptyack@whoi.edu), Mark Johnson (Woods Hole Oceanographic Institution, Applied Ocean Physics & Engineering Dept., Woods Hole, MA 02543, USA, majohnson@whoi.edu)

The transmission beam pattern of a female *Mesoplodon densirostris* tagged on October 23, 2006 in the Tongue of the Ocean, Bahamas is estimated using data from a Woods Hole Oceanographic Institution DTag and simultaneous recordings from broadband, bottom mounted hydrophones at the Atlantic Undersea Test and Evaluation Center, Andros Island, Bahamas. The bottom mounted hydrophones are used to localize the tagged animal and measure the received level of each click. Pitch, roll and heading from the DTag are used to determine the horizontal and vertical aspect angles relative to the hydrophone. An estimate of the *M. densirostris* horizontal and vertical transmission beam pattern based on four dives will be presented.



## Session 4pABd

## Animal Bioacoustics: General Topics in Animal Bioacoustics I

Richard R. Fay, Cochair

*Loyola University Chicago, Parmly Hearing Institute, 6525 N. Sheridan Rd., Chicago, IL 60626, USA*

Michel Andre, Cochair

*Laboratori d'Aplicacions Bioacústiques, Universitat Politècnica de Catalunya, avda. Rambla Exposició s/n, Vilanova i la Geltrú, 08800, Spain*

## Contributed Papers

5:20

**4pABd1. Vocalizations of the Spotted Hyena (*Crocota crocuta*): Eliciting Acoustic Variation in Groans.** Frédéric E. Theunissen (UC Berkeley, Dept. of Psychology, 3210 Tolmunt Hall, Berkeley, CA 94720-1650, USA, theunissen@berkeley.edu), Steve Glickman (UC Berkeley, Dept. of Psychology, 3210 Tolmunt Hall, Berkeley, CA 94720-1650, USA, glickman@berkeley.edu), Suzanne Page (UC Berkeley, Dept. of Psychology, 3210 Tolmunt Hall, Berkeley, CA 94720-1650, USA, mweldele@berkeley.edu)

Spotted hyenas (*Crocota crocuta*) are highly social animals possessing a complex vocal repertoire. Vocal signals of different types correlate with distinct social circumstances. Groans constitute a very large category in the spotted hyena repertoire which is not well understood. Sounds labeled as groans vary in their acoustical quality from more growling sounds to more tonal vocalizations. Groans are also elicited in many different social interactions. To begin to decipher the meaning of these vocal signals, we examined how the variation in the acoustic properties of groans was correlated with experimentally controlled eliciting conditions. Groans were elicited in adult hyenas presented with three objects: unfamiliar spotted hyena cubs, meaty bones, and the empty transport cage used to contain bones or cubs on other trials. Cubs elicited more groans from more adults than other objects but all objects elicited vocal responses. More importantly, discriminant analysis revealed differences in the acoustic characteristics of groans elicited by cubs and those elicited by other objects. Cubs elicited more prolonged and more tonal groans with higher fundamental frequency. Our study shows that groans can be classified into different groups and that these different sounds were produced in different behavioral contexts.

5:40

**4pABd2. The relationship between complex vocal signaling and immunocompetence in the brown-headed cowbird (*Molothrus ater*).** Samantha Levinson (UC Santa Barbara, 6689 El Colegio Rd. Apt. 46, Goleta, CA 93117, USA, slevinson@umail.ucsb.edu), Loren Merrill (UC Santa Barbara, 6689 El Colegio Rd. Apt. 46, Goleta, CA 93117, USA, lmerrill@lifesci.ucsb.edu)

Male brown-headed cowbirds (*Molothrus ater*) use song in mate attraction and in male-male competition to signal their quality to potential mates and competitors. Singing is an energetically expensive activity for birds, so repertoire size, singing rate, and complexity of songs should be honest indicators of a male's overall quality. Resistance to parasites is a major component of male quality, and some aspects of immunity are inherited by his offspring. Females obtain indirect benefits from mating with healthy males by having healthy offspring, so song may signal immune condition to females. Male cowbirds usually sing between four and six different song types when displaying to females. These songs differ in the amount of frequency modulation required during singing, which indicates that the difficulty of production of these songs by the syrinx also varies. In this study, we investigate the correlation between immune function and singing behavior, specifically the size of a bird's repertoire and the complexity of his songs.

Understanding the relationship between repertoire size, song complexity, and immune function is important to understanding the role of vocal signaling in mate attraction in songbirds.

6:00

**4pABd3. The statistics of plant echoes as perceived by echolocating bats.** Yossi Yovel (Eberhard-Karls-Universität Tübingen, Zool. Institut, Abt. Tierphysiologie, Auf der Morgenstelle 28, 72076 Tübingen, Germany, yossiyoel@hotmail.com), Matthias O. Franz (University of Applied Sciences, Konstanz, Germany, HTWG Konstanz, Brauneggerstr. 55, D-78462 Konstanz, Germany, mfranz@htwg-konstanz.de), Peter Stilz (Eberhard-Karls-Universität Tübingen, Zool. Institut, Abt. Tierphysiologie, Auf der Morgenstelle 28, 72076 Tübingen, Germany, peter.stilz@uni-tuebingen.de), Hans-Ulrich Schnitzler (Eberhard-Karls-Universität Tübingen, Zool. Institut, Abt. Tierphysiologie, Auf der Morgenstelle 28, 72076 Tübingen, Germany, hans-ulrich.schnitzler@uni-tuebingen.de)

To explore the statistics of complex natural plant echoes, we emitted bat-like downsweeps (200-0 kHz) and recorded the echoes of various tree species. A Hilbert transform was used to calculate the envelope of the echoes impulse responses. This corresponds to a one-dimensional representation of the spatial reflector arrangement of the plant. We then calculated the envelope's power spectrum to assess the amount of periodic structures. In control experiments we compared power spectra of a single leaf, a branch and a few branches, and tested the effect of systematically decreasing the leaf density of a plant. On a bi-logarithmic plot, the averaged power spectra of all trees have a sigmoid shape with three approximately linear domains that represent different scales of structure, but differ between species. We hypothesize that the first domain is influenced by the gross skeleton of branches, while the others are associated with smaller scale structures. The control experiments showed a similar dependence between leaf density and power spectrum. Modeling plants as simple three-dimensional textures with stationary statistics was sufficient to predict the characteristics of the spectra. Our findings suggest an interpretable relation between the power spectrum of the echo's envelope and the spatial statistics of the plant.

6:20

**4pABd4. Modelling of the echo generation process in bat echolocation.** Timos Papadopoulos (University of Southampton, SPCG-ISVR, SO17 1BJ Southampton, UK, tp@isvr.soton.ac.uk), Robert Allen (University of Southampton, SPCG-ISVR, SO17 1BJ Southampton, UK, R.Allen@soton.ac.uk)

Very few studies exist that attempt to model the detailed shape of the echoes generated by real targets in bat echolocation. The modelling becomes even more complicated when one attempts to take into account the specific acoustic characteristics of the bat as a source and receiver. Hence, the exact physical acoustics basis that underpins the target detection and classification capabilities demonstrated by bats remains a largely open research question. We use previously published work on real target echo measurement (Simmons and Chen, JASA 1989) as a starting point but modify their ex-

perimental method in a way that allows the incorporation of the bat's source and receiver characteristics in the modelling. Furthermore, we compare our measurements with analytically predicted results and show good agreement. We discuss how our experimental method can be used for the prediction of the binaural signals that constitute the actual input to the bat's auditory system during echolocation.

6:40

**4pABd5. The simulation of bat-oriented auditory processing using the experimental data of echolocating signals.** Su Yeon Kim (University of Southampton, SPCG-ISVR, SO17 1BJ Southampton, UK, syk@isvr.soton.ac.uk), Robert Allen (University of Southampton, SPCG-ISVR, SO17 1BJ Southampton, UK, R.Allen@soton.ac.uk), Daniel Rowan (University of Southampton, SPCG-ISVR, SO17 1BJ Southampton, UK, dr@isvr.soton.ac.uk)

There are various approaches to understanding the echolocation phenomenon of bats. A part of the echolocating process is assessed here by determining what acoustic signal a bat's ears receive during echolocation. It is simplified in an experimental rig to measure the reflections from objects in different horizontal angles which represents a sound localisation task in bats. It has been assumed in this study that the remarkable echolocating ability of bats, which is not shown in the most other mammalian species, is achieved by their different physical shape of head and ears, and specialised auditory processing of echolocating signals. In human studies in terms of sound localisation, physical characteristics are usually modelled as a head-related transfer function (or HRTF) and gammatone filter banks are widely used to simulate auditory processing in the cochlear. A modified filterbank is used here to represent the auditory processing in bats and combined with the experimental data of object reflections. Bat HRTFs will be used subsequently to determine the acoustic reflections at both ears.

7:00

**4pABd6. Steady streaming near model cod otoliths.** Charlotte Kotas (Georgia Institute of Technology, Mechanical Engineering, 771 Ferst Drive, Atlanta, GA 30332-0405, USA, charlotte.kotas@gatech.edu), Peter H. Rogers (Georgia Institute of Technology, Mechanical Engineering, 771 Ferst Drive, Atlanta, GA 30332-0405, USA, peter.rogers@me.gatech.edu), Minami Yoda (Georgia Institute of Technology, Mechanical Engineering, 771 Ferst Drive, Atlanta, GA 30332-0405, USA, minami.yoda@me.gatech.edu)

Typical fish distinguish sounds at about 10-1000 Hz with particle motions as small as 0.1 nm and angular separation near 10° using their ears, which contain dense, bony otoliths weakly suspended in endolymph and tissue. The otoliths oscillate relative to incident sound, inducing flows in the surrounding fluid which are in turn sensed by the hair cells on the overlying macular membrane which project into the groove-like sulcus on the otolith. These fluid flows are then sampled by the hair cells and "heard" by the fish. The irregular geometry of the otolith shapes the flow patterns. The hair cells, which are organized into different ciliary orientation groups on the macula,

preferentially sample the flow patterns along their axes. The steady component of the fluid motion near enlarged models of an actual cod otolith oscillated at 2--20 Hz along different directions was studied in the vicinity of the hair cells using phase-locked particle-image velocimetry (PIV) and pathline visualizations. The possible relationship between the flow patterns and hearing capabilities is discussed. Although the oscillation amplitudes studied are much larger than those for underwater sound, evidence suggests that the flow patterns are amplitude independent. [Supported by ONR.]

7:20

**4pABd7. Sound source segregation by goldfish: Two simultaneous tones.** Richard R. Fay (Loyola University Chicago, Parnly Hearing Institute, 6525 N. Sheridan Rd., Chicago, IL 60626, USA, rfay@luc.edu)

In 1964, van Bergeijk asked, "...given that a fish can discriminate between sounds A and B when they are presented separately, can he still discriminate either one when both are presented simultaneously? Or do the two sounds blend to form a new entity (such as a chord)." This question was a very early query about sound source segregation by fishes. We have investigated the role that frequency separation plays in this phenomenon using classical respiratory conditioning with a stimulus generalization paradigm. Groups of animals were first conditioned to 2-tone mixtures comprised of 150 Hz (A) and another frequency (B) ranging from 300 Hz to 750 Hz. Generalization tests were then carried out using single tones between 50 and 900 Hz. Group mean generalization gradients showed that the two tones were segregated at the widest A-B spacings, and tended to be segregated as the A-B spacing was reduced. The limited resolution of the test procedure did not allow an answer at the narrowest spacings. Source segregation, even of abstract tones, may be a fundamental feature of the sense of hearing of all organisms, and occurs in the absence of auditory cortex and complex cognitive abilities.

7:40

**4pABd8. Passive Acoustic Detection and Monitoring of Schools of Herring.** Thomas R. Hahn (University of Miami, RSMAS, 4600 Rickenbacker Causeway, Miami, FL 33149, USA, t.hahn@miami.edu), Gary L. Thomas (University of Miami, RSMAS, 4600 Rickenbacker Causeway, Miami, FL 33149, USA, gthomas@rsmas.miami.edu)

Passive acoustic detection and monitoring of various marine fishes has recently received much attention in the literature. It has been recognized that passive acoustic techniques have the potential to complement traditional active acoustic surveys and to significantly increase their overall efficiency, if the acoustic signatures of the considered species are well understood. In this paper, the potential of passive acoustic techniques is explored for the specific case of Pacific herring (*Clupea palassii*). It is demonstrated that schools of herring can acoustically be detected by observing the sound of coordinated bubble release, triggered, e.g., by predator activity. This sound not only has identifiable features that can be exploited for determining the presence or absence by simple means, but could also carry abundance and size information. Work supported by ONR and the NMFS via the PWSSC.

**Session 4pAOa****Acoustical Oceanography, Underwater Acoustics, and ECUA: Rapid Environmental Assessment**

John C. Osler, Cochair

*Defence R&D Canada - Atlantic, P.O. Box 1012, Dartmouth, NS B2Y3Z7, Canada*

Yann Stephan, Cochair

*SHOM, 13 rue du Chatellier, CS 92803, Brest cedex 2, 29228, France****Invited Paper*****2:00**

**4pAOa1. Overview of U.S. Navy Operational Oceanographic Models in Support of Acoustic Applications.** Richard Allard (Naval Research Laboratory, NRL Code 7322, Stennis Space Center, MS 39571, USA, allard@nrlssc.navy.mil), Charlie Barron (Naval Research Laboratory, NRL Code 7323, Stennis Space Center, MS 39529, USA, barron@nrlssc.navy.mil), Frank Bub (Naval Oceanographic Office, Balch Blvd, Stennis Space Center, MS 39529, USA, frank.bub@navy.mil), Emanuel F. Coelho (University of Southern Mississippi, Balch Blvd, Stennis Space Center, MS 39529, USA, coelho@nrlssc.navy.mil), James Cummings (Naval Research Laboratory, NRL Code 7322, Stennis Space Center, MS 39571, USA, cummings@nrlmry.navy.mil), J. Pacquin Fabre (Naval Research Laboratory, NRL Code 7322, Stennis Space Center, MS 39571, USA, josie.fabre@nrlssc.navy.mil), Robert Helber (Naval Research Laboratory, NRL Code 7322, Stennis Space Center, MS 39571, USA, helber@nrlssc.navy.mil), Clark Rowley (Naval Research Laboratory, NRL Code 7322, Stennis Space Center, MS 39571, USA, rowley@nrlssc.navy.mil)

The Naval Oceanographic Office operational global 1/8° Navy Coastal Ocean Model assimilates satellite and in-situ data to produce daily 72-hr forecasts. Output includes 3D fields of temperature, salinity, u- and v-components of ocean currents at standard depth levels, and these support derived fields including sound speed and sonic layer depth. The global model provides initial/boundary conditions for nested regional models, primarily relocatable NCOM. The relocatable NCOM modeling system can be set up quickly for areas of interest, includes river and tidal forcing, and is forced with a high-resolution atmospheric mesoscale model. Local and remote observations are incorporated into the models through the Navy Coupled Ocean Data Assimilation system, which assimilates sea surface temperature data from satellite, ships and buoys, profile data from floats and gliders, xbt's, CTD's, fixed and drifting buoys as well as altimeter-derived sea surface heights and ice concentration. In this presentation we will discuss how the operational ocean models feed into acoustic prediction models and tactical decision aids, the role glider observations will play in the modeling strategy, the use of ensembles to provide improved prediction error estimates and guide new observations, and future plans.

***Contributed Paper*****2:20**

**4pAOa2. High-frequency multibeam echosounder classification for rapid environmental assessment.** Kerstin Siemes (Acoustic Remote Sensing Group, Delft Institute of Earth Observation and Space Systems, Delft University of Technology, Kluyverweg 1, 2629 HS Delft, Netherlands, k.siemes@tudelft.nl), Mirjam Snellen (Acoustic Remote Sensing Group, Delft Institute of Earth Observation and Space Systems, Delft University of Technology, Kluyverweg 1, 2629 HS Delft, Netherlands, m.snellen@tudelft.nl), Dick G. Simons (Delft University of Technology, P.O. Box Postbus 5048, 2600 GA Delft, Netherlands, d.g.simons@tudelft.nl), 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), Matthias Meyer (Royal Netherlands Naval College (NLDA) - REA group, PO Box 10000, 1780 Den Helder, Netherlands, mmeyer@ulb.ac.be), Jean-Claude Le Gac (NATO Undersea Research Center, Viale San Bartolomeo 400, 19126 La Spezia, Italy, legac@nurc.nato.int)

For shallow-water naval operations, obtaining rapidly an accurate picture of the environmental circumstances often is of high importance. Hereto a multi-sensor approach is required. In this context, the MREA/BP'07 ex-

periment has been carried out south of Elba (Mediterranean Sea), where several techniques of environmental characterization covering the fields of underwater acoustics, physical oceanography and geophysics have been combined [Le Gac&Hermand, 2007]. The required information typically concerns water-column properties, sea surface roughness, and sediment geo-acoustic properties. Estimating these geo-acoustic parameters from inversion of acoustic data received on drifting sparse arrays has proved to be a promising approach. Part of MREA/BP'07 was therefore dedicated to this type of measurement. For validating the resulting geo-acoustic estimates sediment samples were collected. Additionally, measurements were carried out using a multibeam-echosounder. This system provides depth information, but also allows for seafloor classification. The classification approach taken is model-based employing the backscatter data. It discriminates between sediments in the most optimal way by applying the Bayes decision rule for multiple hypotheses, implicitly accounting for backscatter-strength ping-to-ping variability. Here, results of seafloor classification using the multibeam data and a preliminary comparison with the sediment sample analysis and the geo-acoustic parameter estimates as obtained from the drifting arrays are presented.

2:40

**4pAOa3. Integrated scheme of rapid environmental assessment for shallow water acoustics.** Jean-Claude Le Gac (NATO Undersea Research Center, Viale San Bartolomeo 400, 19126 La Spezia, Italy, legac@nurc.nato.int), 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), Frans Absil (Royal Netherlands Naval College (NLDA) - REA group, PO Box 10000, 1780 Den Helder, Netherlands, fgj.absil@nllda.nl)

Predicting sound propagation in shallow or very shallow water environments requires that the frequency-dependent acoustic properties be assessed for all components of the waveguide, i.e., the water column, sea bottom and sea surface interface. During the Maritime Rapid Environmental Assessment MREA/BP'07 sea trial in April-May 2007, south of Elba Island in the Mediterranean Sea, an integrated MREA scheme has been implemented to provide a full 4D (3D+T) environmental picture that is directly exploitable by acoustic propagation models. Based on a joint multi-disciplinary effort, several standard and advanced techniques of environmental characterization covering the fields of underwater acoustics, physical oceanography and geophysics have been combined within a coherent scheme of data acquisition, processing and assimilation. The paper presents the whole architecture of the implemented scheme. Based on a preliminary analysis of MREA/BP'07 data, advantages and drawbacks of the approach will be discussed. Ways ahead for further improvement and perspectives are finally drawn.

3:00

**4pAOa4. The application of rapid environmental assessment to sonar performance.** Paul C. Hines (Defence R&D Canada - Atlantic, P.O. Box 1012, Dartmouth, NS B2Y3Z7, Canada, Paul.Hines@drdc-rddc.gc.ca), Sean Pecknold (Defence R&D Canada - Atlantic, P.O. Box 1012, Dartmouth, NS B2Y3Z7, Canada, sean.pecknold@drdc-rddc.gc.ca), John C. Osler (Defence R&D Canada - Atlantic, P.O. Box 1012, Dartmouth, NS B2Y3Z7, Canada, john.osler@drdc-rddc.gc.ca)

Naval sonar operations and planning in littoral environments requires information from historical databases, in situ sampling of environmental parameters, and models capable of estimating sonar performance and the uncertainty in the estimate. Defence R&D Canada's approach to enable rapid environmental assessment (REA) for sonar incorporates three components: (1) a GIS-enabled database to manage historical environmental data, (2) measurement tools that operate while underway to provide in situ sampling of water column and seabed properties, and (3) a sensitivity model that examines the relative importance of different environmental parameters in order to quantify the impact of incomplete or degraded environmental information, and to specify the appropriate spatial and temporal scales for sampling. In this presentation, the integration of these aspects of REA will be demonstrated using transmission loss data collected in shallow water. Predictions made using REA data provide a substantially better fit to the measurements than those using historical databases.

3:20

**4pAOa5. Systemic approach to performance prediction and the exploitation of environmental information in naval systems.** Patrick Grenard (DGA/DET/CTSN, BP28, 83800 Toulon Armées, France, Patrick.Grenard@dga.defense.gouv.fr)

Interoperability of systems and implementation of network enabled capabilities are at the heart of defense transformation. By providing extensive capabilities for data gathering, processing and presentations, they achieve a greater tactical advantage through superior knowledge of and use of the operational environment. Within the framework of the deployment of the new French Navy information system SIC21, a system of systems is being developed in order to support the implementation of the NATO concept of Recognized Environmental Picture (REP). The metasystem established allows for the assimilation of Rapid Environmental Assessment (REA) data, and for a local or remote production of the REP. In order to support decision making and to guide actions local exploitation of the REP is carried out through embedded tactical decision aids. They rely on advanced performance prediction tools including propagation modelling capabilities called in simulation scenarios. The aims of the implemented models are to predict performances at the appropriate level of granularity, to provide a confidence level on the proposed tactical picture, to define the appropriate level of marine environmental knowledge that should be acquired, to optimise sensor settings, and finally to provide inversion tools for the REA, ensuring consistency with forward modelling used in performance prediction.

4p THU. PM



**Session 4pAOB****Acoustical Oceanography, Signal Processing in Acoustics, and ECUA: Passive Acoustic Tomography I  
(Poster Session)**

Karim Sabra, Cochair

*Georgia Institute of Technology, School of Mechanical Engineering*

Sergio Jesus, Cochair

*ISR*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

***Contributed Paper*****4pAOB1. Low-frequency acoustic signature of hurricane Ernesto.**

James Traer (Marine Physical Laboratory, Scripps Institute of Oceanography, 8602 La Jolla Shores Drive, La Jolla, CA 92093-0238, USA, jtraer@ucsd.edu), Peter Gerstoft (Marine Physical Laboratory, Scripps Institute of Oceanography, 8602 La Jolla Shores Drive, La Jolla, CA 92093-0238, USA, gerstoft@ucsd.edu), Laura Brooks (Marine Physical Laboratory, Scripps Institute of Oceanography, 8602 La Jolla Shores Drive, La Jolla, CA 92093-0238, USA, lbrook02@gmail.com), William Hodgkiss (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, wsh@mpl.ucsd.edu), David Knobles (Applied Research Laboratories, UT at Austin, P. O. Box 8029, Austin, TX 78713, USA, knobles@arlut.utexas.edu)

The ambient noise level variations produced by Hurricane Ernesto were observed by the SWAMI32, SWAMI52 and SHARK arrays as the storm passed over the SW06 shallow water site. Microseism signals were detected in the water column near 0.1 Hz and were tracked with a beamformer over a period of several days observing variations that were very closely linked to measured surface waves. 5-75 Hz beamforming showed a sound-field dominated by local surface-noise punctuated by brief surges of noise from distant sources. Beamforming and time-domain cross-correlations showed that changes in acoustic environment on the time-scale of hours occurred at all three arrays with good correlation in time and directionality suggesting the storm induced noise-field is homogeneous over many kilometers.

## Session 4pAOc

## Acoustical Oceanography and ECUA: General Topics in Acoustical Oceanography I (Poster Session)

Andone Lavery, Cochair  
*Woods Hole Oceanographic Institution*

Yann Stephan, Cochair  
*SHOM*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pAOc1. The sea and ocean 3D acoustic waveguide: rays dynamics and chaos phenomena.** Alexander V. Glushkov (Odessa University, P.O.Box 24a, 65009 Odessa-9, Ukraine, glushkov@paco.net), Andrey A. Svinarenko (Odessa University, P.O.Box 24a, 65009 Odessa-9, Ukraine, glushkov@paco.net), Olga Y. Khetselius (Odessa University, P.O.Box 24a, 65009 Odessa-9, Ukraine, glushkov@paco.net), Nikolay S. Serbov (Odessa University, P.O.Box 24a, 65009 Odessa-9, Ukraine, glushkov@paco.net)

It has been carried out modeling of the sea and ocean 3D acoustic waveguide. On the basis of Hamiltonian equations of rays it is studied a dynamics of rays. It has been shown that for acoustic waveguide in a shallow sea with non-level bottom under the rays propagation in a waveguide dependence of the of temporal frequency upon the output angle represents a fractal measure in accordance with Abdullaev-Zaslavsky result. For the ocean 3D acoustic waveguide on the basis of solving the eiconal equations in the Hamiltonian form it has been studied the fractal dynamics, including the chaotic one. There are presented the data of numerical solution of equations for the typical acoustic channel in the North-Atlantic region. The conditions for the Arnold diffusion effect realization are discussed.

**4pAOc2. Fuzzy Clustering of Oceanographic Sound Speed Profiles for Acoustic Characterization.** John Dubberley (Naval Research Laboratory, Bldg. 1005 Rm D-23, Stennis Space Center, MS 39529, USA, john.dubberley@nrlssc.navy.mil), Robert Zingerelli (Naval Research Laboratory, Bldg. 1005 Rm D-23, Stennis Space Center, MS 39529, USA, robert.zingerelli@nrlssc.navy.mil)

Historic oceanographic sound speed profiles have traditionally been grouped by area and time period, usually one degree square area and monthly time. After grading the profiles, mean profiles and standard deviations are calculated from the accepted profiles and in the acoustics community they are then used to predict the expected acoustic response of the region. Here the historic profiles in NOAA's World Ocean Database 2005 (WOD2005) will be divided into the same area and time periods, but in subsets with a sufficient number of profiles, fuzzy clustering will be employed

on acoustically relevant oceanographic parameters (mixed layer depth, surface temperature, sound speed gradient, etc) to divide the population into multiple clusters. A parabolic equation acoustic transmission model is then applied on the WOD2005 statistical profiles and on the fuzzy cluster populations. Conclusions will be drawn about the suitability of this clustering to capture the variability of acoustic response at a given time and place.

**4pAOc3. On the consideration of motion effects in underwater geoacoustic inversion.** Nicolas Josso (GIPSA-lab, dep. DIS, 961, rue de la Houille Blanche, 38402 St Martin d'Hères, France, nicolas.josso@gipsa-lab.inpg.fr), Cornel Ioana (GIPSA-lab, dep. DIS, 961, rue de la Houille Blanche, 38402 St Martin d'Hères, France, cornel.ioana@gipsa-lab.inpg.fr), Cédric Gervaise (E3I2 - EA3876, 2 rue François Verny, 29806 Brest Cedex, France, cedric.gervaise@ensieta.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 estimation of an impulse response (IR) of a propagation channel is necessary for a large number of underwater acoustic applications: underwater communication, sonar detection and localization, marine mammal monitoring, etc. Basically, it informs us about the distortions of a transmitted signal in one underwater channel. This operation is usually subject to additional distortions due to the motion of the transmitter-channel-receiver configuration. This paper points on the effects of the motion while estimating the IR for shallow water environments in the very low frequencies bandwidth with matching filtering between the transmitted and the received signals. We propose a methodology to compare between the IR estimation in motionless and motion contexts, respectively. Using this methodology an objective criterion for motion effect analysis is proposed in order to measure the distortions due to the motion phenomena. The proposed methodology is applied to real data sets issued from PASSTIME campaign (SHOM, Bay of Biscaye, 2005) proving also its interest for motion effect analysis.

**Session 4pAOd**

**Acoustical Oceanography and ECUA: Acoustic Characterization of Sea Floor Habitats I (Poster Session)**

Christian De Moustier, Cochair  
*UNH, Center for Coastal and Ocean Mapping*

Dick Simons, Cochair  
*Delft University of Technology*

Xavier Lurton, Cochair  
*Institut Français de Recherche pour l'Exploitation de la Mer*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

***Contributed Paper***

**4pAOd1. Hierarchical spline technique application for real time 3D displaying of seafloor using multibeam sonar data.** Jerzy Demkowicz (Gdansk University of Technology, ul. Narutowicza 11/12, 80-952 Gdansk, Poland, demjot@eti.pg.gda.pl)

Multibeam sonar records have a high resolution raster character. Unfortunately, interpolating and approximating and eventually displaying scattered 3D raster data of high volume leads to some difficulties related to a computer processing power. Usually the problem solution leads to multi-

resolution wavelet approach. The paper presents some advantages of using hierarchical splines as applied to real data from multibeam EM 3002 sonar acquired during acoustic survey on Southern Baltic. The proposed approach is two folded: firstly, all acquired multibeam sonar raw data are interpolated with high density uniform spline interpolation. The knots and control points of interpolated network are saved for defined resolution level. In the next stage, preprocessed high resolution data are combined with low resolution data sets after knot decimation process. Such approach allow real time 3D displaying of multibeam sonar data for different zoom levels

## Session 4pAOg

**Acoustical Oceanography and ECUA: Acoustic Characterization of Sea Floor Habitats II**

Christian De Moustier, Cochair

*UNH, Center for Coastal and Ocean Mapping, Chase Ocean Engineering Lab, 24 Colovos Road, Durham, NH 03824, USA*

Dick G. Simons, Cochair

*Delft University of Technology, P.O. Box Postbus 5048, Delft, 2600 GA, Netherlands*

Xavier Lurton, Cochair

*Institut Français de Recherche pour l'Exploitation de la Mer, NSE/AS, BP 70, Plouzané, 29280, France**Invited Paper*

5:20

**4pAOg1. High frequency scattering measurements for mussel bed characterisation.** M. Snellen (Delft Institute of Earth Observation and Space Systems, Delft University of Technology, Kluyverweg 1, 2629 HS Delft, Netherlands, M.Snellen@tudelft.nl), Dick G. Simons (Delft University of Technology, P.O. Box Postbus 5048, 2600 GA Delft, Netherlands, d.g.simons@tudelft.nl), Rolf Riethmueller (Institute for Coastal Research, GKSS Research Centre Geesthacht, Max-Planck-Str. 1, D-21502 Geesthacht, Germany, rolf.riethmueller@gkss.de)

Several approaches exist towards seafloor classification using high-frequency backscattering measurements. The classification approach taken in this paper is a model-based classification employing backscatter data measured by a multibeam-echosounder (MBES) system. The method discriminates between sediments in the most optimal way by applying the Bayes decision rule for multiple hypotheses, implicitly accounting for the backscatter strength ping-to-ping variability. The method's applicability for seafloor classification has been demonstrated by using 300 kHz MBES data collected in the Cleaver-Bank area (North Sea). The area is well-known from a geological point of view due to extensive sampling campaigns and is characterized by a wide variety of seafloor types. Here we apply the classification method to MBES data acquired in the Oosterschelde estuary (the Netherlands) which is known to contain mussel culture spots. Also recordings using a video camera towed close to the seafloor and core measurements have been taken. From the video recordings estimates of mussel coverage as a function of position were derived. Analysis of the MBES results shows that they clearly reveal the presence of the mussel beds, indicating the usefulness of acoustic classification for habitat mapping. A comparison between MBES analysis results, video recordings and sample analysis will be presented.

*Contributed Papers*

5:40

**4pAOg2. Vertical echosounder versus side-scan sonar mapping of Posidonia Oceanica fields.** Noela Sanchez-Carnero (Universidade da Coruña, Campus de Zapateira s/n, E-15071 A Coruña, Spain, noesanchez@udc.es), Víctor Espinosa (IGIC - Universitat Politècnica de València, Cra. Nazaret-Oliva S/N, E-46730 Gandia, Spain, vespinos@fis.upv.es), Miguel Rodilla (IGIC - Universitat Politècnica de València, Cra. Nazaret-Oliva S/N, E-46730 Gandia, Spain, mrodilla@hma.upv.es), Ester Soliveres (IGIC - Universitat Politècnica de València, Cra. Nazaret-Oliva S/N, E-46730 Gandia, Spain, essogon@epsg.upv.es), Juan Freire (Universidade da Coruña, Campus de Zapateira s/n, E-15071 A Coruña, Spain, jfreire@udc.es)

Posidonia fields in the "Cabo de Gata" marine natural park, located in the south-east mediterranean spanish coast, have been mapped by means of two different acoustical tools: a vertical single-beam scientific echosounder and a side-scan sonar. The measured transects have been dived and recorded with a video camera in order to validate the predictions from the extracted acoustical data. We compare both the results obtained from the application of commercial software for bottom classification, and the processing with alternative algorithms in the case of the vertical echosounder, with those derived from the analysis of the side-scan sonar data.

6:00

**4pAOg3. Comparison of multi-beam and single-beam seabed backscatter and sampling resolution near normal incidence.** Rudy Kloser (GPO Box 1538, 7001 Hobart, Australia, rudy.kloser@csiro.au)

The seabed backscatter difference near normal incidence ( $<30^\circ$ ) is compared between two multibeam (EM300 and EM1002) and two on-axis calibrated single beam EK60 (38 and 120 kHz) echo sounders and found to differ by 7 to 10 dB. A seabed backscatter model supported the calibration of the single beam backscatter near normal incidence. Potential errors due to incorrectly applied equivalent area compensation were found to contribute to this difference but not significantly. Estimating the equivalent area of sampling from multibeam and single beam echosounders highlights the complexities of sampling resolutions as a function of incidence angle and pulse length. These variable sample sizes influence the measured backscatter due to the patchy nature of seabed substrate and fauna. The current backscattering processing method for a national upper slope mapping program relies on producing metrics at the highest spatial resolution. Calibration errors on these metrics for seabed classification for a national mapping program are discussed.



#### 4pAOg4. Using the MBES for classification of riverbed sediments.

Alireza Amiri-Simkooei (Acoustic Remote Sensing Group, Delft Institute of Earth Observation and Space Systems, Delft University of Technology, Kluyverweg 1, 2629 HS Delft, Netherlands, a.amirisimkooei@tudelft.nl), Mirjam Snellen (Acoustic Remote Sensing Group, Delft Institute of Earth Observation and Space Systems, Delft University of Technology, Kluyverweg 1, 2629 HS Delft, Netherlands, m.snellen@tudelft.nl), Dick G. Simons (Delft University of Technology, P.O. Box Postbus 5048, 2600 GA Delft, Netherlands, d.g.simons@tudelft.nl)

For keeping the Dutch rivers suitable for commercial activities measures are required. For example, the bottom of the river Waal, connecting Rotterdam with German industrial areas, is subsiding. Since the subsidence varies along the river, dangerous shoals occur. Sediment suppletions are planned to counteract the subsidence. Appropriate suppletion material is expected to keep the bottom more stable. To monitor the suppletion effectiveness, multibeam-echosounder (MBES) measurements are planned, allowing for simultaneous estimation of bathymetry and sediment composition. For the latter, we apply a method employing the MBES backscatter data. It estimates the number of sediment types present in the survey area and discriminates between them by applying the Bayes decision rule for multiple hypotheses, implicitly accounting for the backscatter strength ping-to-ping variability. The method's applicability was demonstrated in a well-surveyed test area (North Sea). In 2007, MBES measurements were acquired at the Waal, accompanied with extensive sediment grabbing. Contrary to the test area, water depths are very shallow and significant bottom slopes exist, requiring corrections. The lower water depths correspond to smaller beam-footprints, resulting in a higher ping-to-ping variability. Consequently the discriminating power between sediments will decrease. The performance of the classification method for this river environment is assessed.

#### 4pAOg5. Analysis of Backscatter and Seafloor Acoustical Properties for Geosciences and Biodiversity Mapping Studies in Cook Strait, New Zealand.

Xavier Lurton (Institut Français de Recherche pour l'Exploitation de la Mer, NSE/AS, BP 70, 29280 Plouzané, France, lurton@ifremer.fr), Geoffroy Lamarche (National Institute of Water and Atmospheric Research (NIWA), Private bag 14-901, 6021 Wellington, New Zealand, g.lamarche@niwa.co.nz), Anne-Laure Verdier (National Institute of Water and Atmospheric Research (NIWA), Private bag 14-901, 6021 Wellington, New Zealand, a.verdier@niwa.co.nz), Jean-Marie Augustin (Institut Français de Recherche pour l'Exploitation de la Mer, NSE/AS, BP 70, 29280 Plouzané, France, augustin@ifremer.fr), Ian Wright (National Institute of Water and Atmospheric Research (NIWA), Private bag 14-901, 6021 Wellington, New Zealand, i.wright@niwa.co.nz), Ashley Rowden (National Institute of Water and Atmospheric Research (NIWA), Private bag 14-901, 6021 Wellington, New Zealand, a.rowden@niwa.co.nz), Alan Orpin (National Institute of Water and Atmospheric Research (NIWA), Private bag 14-901, 6021 Wellington, New Zealand, a.orpin@niwa.co.nz), Miles Dunkin (National Institute of Water and Atmospheric Research (NIWA), Private bag 14-901, 6021 Wellington, New Zealand, m.dunkin@niwa.co.nz)

A quantitative analysis was conducted over sonar backscatter data collected on the Cook Strait region, central New Zealand, featuring multibeam (~30 kHz) bathymetry and backscatter data, groundtruthed by an extensive geological database (photographs, seabed samples, high-resolution seismics). A first processing step removes the effects of the sounder, seafloor topography, and water column. A second step includes sonar image mosaicing, signal calibration and compensation, speckle noise filtering, image segmentation and textural analysis. Backscatter angular dependence is then extracted from the raw data accounting for the co-registered multibeam bathymetry; it is linked to the various facies of this geologically very active region, forming a catalogue usable for future investigation. Some local fea-

tures are analysed in details, referring to the geological local context. Also the backscatter data from the Haungaroa volcano were used for a proof-of-concept biodiversity mapping exercise. Ecological theory was utilised to predict biodiversity from the seabed substrate heterogeneity, derived from the segmentation of the backscatter data properly pre-processed. The backscatter analysis resulted in the identification of local features with geological, sedimentological, topographic, and possibly biological significance, otherwise not recognised with conventional surveying. This emphasises the potential of backscatter data in submarine seismic hazard studies and large-scale biodiversity mapping.

#### 4pAOg6. Seabed biotope mapping using multi-beam backscatter based on reference sites.

Rudy Kloser (GPO Box 1538, 7001 Hobart, Australia, rudy.kloser@csiro.au)

A multibeam sonar (MBS) was used to discriminate ecological relevant seabed characteristics based on 62 reference sites spanning depths 50 m to 400 m sampled with georeferenced video, sediment grab and rock dredge. The simple ecologically derived terrain characteristics of soft, hard, smooth and rough were found to have the most predictive power for discrimination of the biota using data from video and physical sampling. The acoustic data were corrected for range and incidence angle effects and analysed based on phenomenological characteristics and inversion of a seabed scattering model. Near normal incidence (<16°) the seabed backscatter showed poor correlation to seabed characteristics (cross validation error 32%) and was sensitive to the estimation of the correct seabed incidence angle. Using the trend in backscatter near normal incidence (<30°) greatly improved (cross validation error 4%) the classification but also increased the spatial scale of classification. The length of biotope scales were derived from the video transects with 50% of patches less than 18 m. Referencing the seabed backscatter to a consistent incidence angle (40°) gave the highest spatial resolution derived metric and minimised range, incident angle and beam compensation errors. Using this simple metric, high probabilities of prediction of fauna functional groups were recorded.

#### 4pAOg7. Statistics of seafloor backscatter measured with multibeam sonar systems.

John D. Penrose (Curtin University of Technology, Centre for Marine Sci & Tech, GPO Box U1987, 6845 Perth, WA, Australia, j.penrose@cmst.curtin.edu.au), Alexander Gavrilov (Curtin University of Technology, Centre for Marine Sci & Tech, GPO Box U1987, 6845 Perth, WA, Australia, a.gavrilov@cmst.curtin.edu.au), Iain M. Parnum (Curtin University of Technology, Centre for Marine Sci & Tech, GPO Box U1987, 6845 Perth, WA, Australia, i.parnum@cmst.curtin.edu.au)

A number of theoretical models for seafloor backscatter statistics developed for the recent years show a good agreement with experimental measurements made with sonar systems. However, methods of data collection used in multibeam systems are commonly not taken into consideration when analysing backscatter statistics. Using data collected with a Reson Seabat 8125 system and based on theoretical considerations, it is shown that the seafloor backscatter strength derived from the peak intensity measured as a single value for each beam leads to considerable backscatter overestimation at oblique angles of incidence when the beam footprint is much larger than the insonification area. This occurs because variations of the peak intensity are extreme value distributed, which can be well approximated by the Gumbel distribution. The location parameter of the Gumbel distribution depends on the ratio of the footprint and insonification areas, which results in distorted angular dependence of backscatter strength estimates. On the other hand, the average backscatter strength derived from the integral intensity, i.e. backscatter energy, is a consistent estimate of the actual seafloor backscatter strength. The Gamma distribution is demonstrated to be a good approximation for statistics of the integral intensity, even when the scattering regime is expected to be non-Rayleigh.

**Session 4pAOh****Acoustical Oceanography, Signal Processing in Acoustics, and ECUA: Passive Acoustic Tomography II**

Karim G. Sabra, Cochair

*Georgia Institute of Technology, School of Mechanical Engineering, 771 Ferst Drive, NW, Atlanta, GA 30332-0405, USA*

Sergio Jesus, Cochair

*ISR, Universidade do Algarve, Faro, PT-8005-139, Portugal****Invited Papers*****5:40****4pAOh1. Analysis of passive seabed imaging techniques.** Thomas M. Siderius (HLS Research Inc., 3366 N. Torrey Pines Ct, Suite 310, La Jolla, CA 92037, USA, siderius@hlsresearch.com)

Recently, a passive technique has been developed to image seabed layering. The method exploits naturally occurring acoustic noise generated on the sea-surface, primarily from breaking waves. The processing exploits the noise coherence through cross-correlations between sensors to recover travel times to significant seabed reflectors. To make this a practical tool, beamforming is used with a vertical array of hydrophones and this greatly reduces the required averaging times. Several data sets using moored arrays have shown stable returns from the seabed. Imaging seabed layering over extended areas requires the array to move which has been accomplished by allowing the array to drift. This, however, introduces a number of complications. If the array moves too rapidly, there is potential for the ensonified seabed (in the beam) to change within the averaging time. Another potential problem could be caused by vertical motion of the array (e.g. by surface coupling). In this case, Doppler shifts may cause degradation in the cross-correlation peaks. In some cases, these degrading factors may be reduced through signal processing. In this presentation potential mechanisms that degrade passive seabed imaging will be described with possible mitigating signal processing. Numerical modeling and measured data sets will be analyzed.

**6:00****4pAOh2. Passive tomography of the oceanic environment using ambient noise cross-correlations.** 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)

The random nature of noise and scattered fields tends to suggest limited utility. Indeed, acoustic fields from random sources or scatterers are often considered to be incoherent, but there is some coherence between two sensors that receive signals from the same individual source or scatterer. An estimate of the Green's function (or impulse response) between two points can be obtained from the cross-correlation of ambient noise recorded at these two points. Recent theoretical and experimental studies in ultrasonics, civil engineering, underwater acoustics and seismology have investigated this technique in various environments and frequency ranges. These results provide a means for passive tomography of the ocean environment using only the ambient noise field, without the use of active sources. The coherent wavefronts emerge from a correlation process that accumulates contributions over time from noise sources whose propagation paths pass through both receivers. We will examine the background physics of extracting these coherent structures and present experimental results confirming these theoretical arguments. Further we will present experimental results such as using noise for time synchronization and localization of unconnected acoustic receivers, and for constructing passive tomographic images of the environment.

**6:20****4pAOh3. Passive Geoacoustic Inversion using broadband ship noise in Bay of Biscay shallow water environment.** Cédric Gervaise (E3I2 - EA3876, 2 rue François Verny, 29806 Brest Cedex, France, cedric.gervaise@ensieta.fr)

Our communication proposes a new geoacoustic inversion method for shallow water environments (100 to 300 m). The method relies on the inversion of broadband noise produced by ships of opportunity. The interference patterns generated by the ship movement and the propagation properties of the waveguide are exploited to extract the relative dispersion curves on a chosen bandwidth. These curves are then inverted to estimate geoacoustic properties. This inversion scheme was previously tested (with success) against real data from a very shallow water (10 to 30 m) trial performed off the Southern coast of Barcelona, Spain [1]. To deal with shallow waters, our inversion scheme is improved and tests against real data from PASSTIME trial performed in Bay of Biscay during October 2005. Inversion's results are compared with ground truth and the results showed to be accurate and robust. The proposed technique, which is suitable to a small number of hydrophones and quite easy to implement, offers interesting perspectives for passive geoacoustic inversion. [1] S. Vallez, C. Gervaise, Y. Stephan, M. Andre, Inversion géoacoustique d'un canal très petits fonds à partir des navires en mouvement - traitement incohérent, accepté pour publication *Revue Traitement du Signal*

6:40

**4pAOh4. The passive mode tomography of the ocean using data from short vertical arrays bent by the ocean currents.** Andrey Shurup (Department of Acoustics, Physics Faculty, Moscow State University, Leninskie Gory, 119991 Moscow, Russian Federation, burov@phys.msu.ru), Sergey Sergeev (Department of Acoustics, Physics Faculty, Moscow State University, Leninskie Gory, 119991 Moscow, Russian Federation, sergeev@su29.ru), Valentin Burov (Department of Acoustics, Physics Faculty, Moscow State University, Leninskie Gory, 119991 Moscow, Russian Federation, burov@phys.msu.ru)

The possible realization of passive ocean tomography based on the widely discussed relation between the Green's function and ambient noise cross-correlation is discussed. The problem is considered in the mode representation of acoustic field in adiabatic approximation. The estimated time of signal accumulation required to determine the Green's function with sufficient accuracy reveals the possibility of implementing the schemes of the mode tomography of the ocean based on the measuring the ambient noise field of the ocean. It is shown that the use of the vertical arrays with vector receivers allows a decrease in the accumulation time to one or several hours, depending on the conditions of experiment. The mode structure of acoustic field is determined from the cross-correlation matrix of the noise field received by the hydrophones of short vertical arrays bent by the ocean currents and covering only the part of the sound channel. The proposed algorithm differs from the commonly used mode-filtering procedure and allows a compensation of antenna declination from the vertical profile and takes into account of the finite length of antenna aperture, that is ordinary explored in ocean experiments.

7:00

**4pAOh5. Emergence of the deterministic Green's function from thermal noise in inhomogeneous solids and fluid-solid structures.** Oleg A. Godin (NOAA/ESRL, 325 Broadway, Mail Code R/PSD99, Boulder, CO 80305-3328, USA, Oleg.Godin@noaa.gov)

S. M. Rytov [*A Theory of Electrical Fluctuations and Thermal Radiation* (USSR Academy of Sciences, Moscow, 1953)] was apparently the first to establish theoretically a simple relation between deterministic Green's function and cross-correlation of fluctuations of wave fields generated by random sources. He used reciprocity considerations to analyze fluctuations of electromagnetic fields. An acoustic counterpart of the Rytov's approach was applied in [O. A. Godin, Emergence of the acoustic Green's function from thermal noise, *J. Acoust. Soc. Am.* **121**, EL96-EL102 (2007)] to derive exact and asymptotic relations between acoustic Green's functions and cross-correlation of thermal noise in inhomogeneous moving or motionless fluids. In this paper, these results are extended to solid and fluid-solid media and compared to experimental and theoretical results previously reported in the literature. It is shown that, with the averaging time being sufficiently large, the two-point correlation function of thermal noise contains as much information about the environment as can be obtained acoustically by placing transceivers at the two points. A possibility of retrieving parameters of a fluid motion, such as a flow in a pipe or an oceanic current, from cross-correlations of diffuse noise fields is addressed.

7:20

**4pAOh6. Comparing time domain Green's functions with simulated noise and ambient noise data cross-correlation for a horizontal array.** Stephanie Fried (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, sefried@ucsd.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), 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), William A. Kuperman (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, wkuperman@ucsd.edu)

Previous work has shown that an approximation of the Green's function can be extracted from ambient noise data through cross-correlating the received signals along an array. The resulting Green's function approximation gives accurate time-of-arrivals for the multipaths between hydrophones but can only approximate the magnitude of the arrivals in the time domain Green's function. Nevertheless, some useful information can be obtained from the relative amplitudes of the correlated returns assembled. Further, a Monte Carlo noise model simulation for a similar environment for which noise data was collected reproduces the same cross-correlation arrival structure for the processed ambient noise data and the theoretical time domain Green's function arrival structure. [Research supported by ONR.]

7:40

**4pAOh7. Acoustic modeshape inversion using deep water ambient noise measurements.** Kathleen E. Wage (George Mason University, 4400, University Drive, Fairfax, VA 22030, USA, kwage@gmu.edu), Khalid Almuhan (George Mason University, 4400, University Drive, Fairfax, VA 22030, USA, kalmuhan@gmu.edu)

Assuming that ambient noise can be represented by a sum of uncorrelated acoustic modes, the eigenvectors of the noise covariance matrix for a vertical line array should correspond to the sampled modeshapes. In previous work several authors have investigated using an eigendecomposition of the noise covariance to estimate the mode functions in shallow water, e.g., Wolf et al. [Proc. of the 1993 IEEE Oceans Conf., vol. I, pp. 99-104], Hursky et al. [J. Acoust. Soc. Am., 109(4), pp. 1355-1366], and Nielsen and Westwood [J. Acoust. Soc. Am., 111(2), pp. 748-756]. While the same approach should work for deep water scenarios, only few deep water experiments have deployed arrays with sufficient aperture to resolve the modes, e.g., the work D'Spain et al. [Pure appl. geophys., Vol. 158, pp. 475-512]. This paper explores the problem of inverting for the acoustic modes of a deep water waveguide using ambient noise measurements. In particular the paper focuses on important signal processing issues, including data snapshot requirements, and the effects of array tilt. Data from a deep water propagation experiment will be used to quantify how well the empirical modes match the true modes derived from measured environmental data. [Work supported by an ONR Young Investigator Award.]

## Session 4pBBa

## Biomedical Ultrasound/Bioresponse to Vibration: Ultrasonic Characterization of Bone I

Keith A. Wear, Cochair

*U.S. Food and Drug Administration, Center for Devices and Radiological Health, 10903 New Hampshire Ave, Bldg 62, Rm 3108, Silver Spring, MD 20993, USA*

Kay Raum, Cochair

*Martin Luther University of Halle-Wittenberg, Dept. of Orthopedics, Q-BAM Group, Magdeburger Str. 22, Halle, 06097, Germany*

## Invited Papers

2:00

**4pBBa1. Present state and future trends in ultrasonic characterization of bone.** Pascal Laugier (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, laugier@lip.bhdc.jussieu.fr)

Although it has been over 20 years since the first recorded use of a quantitative ultrasound (QUS) technology to predict bone fragility, the field has not yet reached its maturity. QUS have the potential to predict fracture risk in a number of clinical circumstances and has the advantages of being non-ionizing, inexpensive, portable, highly acceptable to patients and repeatable. However, the wide dissemination of QUS in clinical practice is still limited and suffering from the absence of clinical consensus on how to integrate QUS technologies in bone densitometry armamentarium. There are a number of critical issues that need to be addressed in order to develop the role of QUS within rheumatology. These include issues of technologies adapted to measure the central skeleton, data acquisition and signal processing procedures to reveal bone properties beyond bone mineral quantity and elucidation of the complex interaction between ultrasound and bone structure. In this presentation, we review recent developments to assess bone mechanical properties. We conclude with suggestions of future lines and trends in technology challenges and research areas such as new acquisition modes, advanced signal processing techniques, and models.

2:20

**4pBBa2. Ultrasonic guided waves in bone.** Petro Moilanen (University of Jyväskylä, Department of Physics, PO. Box 35, 40014 Jyväskylä, Finland, pemoilan@cc.jyu.fi)

Recent progress in quantitative ultrasound (QUS) has shown increasing interest towards measuring long bones by ultrasonic guided waves. This technology is widely used in the field of non-destructive testing and evaluation of different waveguide structures. Cortical bone provides such an elastic waveguide and its ability to sustain loading and resist fractures is known to relate to its mechanical properties at different length scales. As guided waves could yield diverse characterization of bone's mechanical properties at the macroscopic level, the method of guided waves has a strong potential over the standardized bone densitometry as a tool for bone assessment. Despite this, development of guided wave methods is challenging, e.g., due to interferences and multiparametric inversion problem. This paper discusses the promises and challenges related to bones characterization by ultrasonic guided waves.

## Contributed Papers

2:40

**4pBBa3. Dual frequency ultrasound measurement of bone - a technique for elimination of soft tissue effects on pulse-echo measurements.** Ossi Riekkinen (University of Kuopio, POB 1627, 70211 Kuopio, Finland, Ossi.Riekkinen@uku.fi), Mikko Hakulinen (University of Kuopio, POB 1627, 70211 Kuopio, Finland, Mikko.Hakulinen@uku.fi), Juha Töyräs (Kuopio University Hospital, POB 1777, 70211 Kuopio, Finland, Juha.Toyras@kuh.fi), Jukka Jurvelin (University of Kuopio, POB 1627, 70211 Kuopio, Finland, Jukka.Jurvelin@uku.fi)

Quantitative ultrasound (US) measurements have been suggested for screening of osteoporosis. However, soft tissues overlying bones affect reliability of the measurements. In this in vitro study, a novel dual frequency ultrasound (DFUS) technique is introduced for elimination of the errors induced by soft tissues on bone pulse-echo US measurements. In DFUS, US reflection from soft tissue-bone interface is measured with two different US frequencies. By knowing the frequency specific US attenuation and speed in adipose and lean tissues, the effect of soft tissue can be determined. DFUS, conducted at frequencies of 2.25 MHz and 5.0 MHz, was validated using

human trabecular bone samples ( $n = 25$ ) covered with heterogeneous soft tissues. DFUS, reduced ( $p < 0.01$ ) the mean error induced by soft tissues from 58.6% to -4.9% and from 127.4% to 23.8% in broadband ultrasound backscattering and integrated reflection coefficients (at 5.0 MHz), respectively. Our results suggest that DFUS is a technique capable to minimize the errors induced by the soft tissue overlying the bone. As no reflection information within soft tissue (adipose-lean tissue interface) is needed in pulse-echo measurements DFUS may enhance the accuracy of ultrasound measurements. Thereby, DFUS shows a significant clinical potential.

3:00

**4pBBa4. Improved standardization methods for clinical measurements of BUA and SOS.** Keith A. Wear (U.S. Food and Drug Administration, Center for Devices and Radiological Health, 10903 New Hampshire Ave, Bldg 62, Rm 3108, Silver Spring, MD 20993, USA, keith.wear@fda.hhs.gov)

**BACKGROUND:** Although calcaneal broadband ultrasound attenuation (BUA) and speed of sound (SOS) are good predictors of osteoporotic fracture risk, BUA and SOS measurements exhibit substantial inter-system



variability. **PURPOSE:** 1) Compare phase insensitive (PI) detection, which suppresses phase cancellation, and conventional phase sensitive (PS) detection for measurement of BUA. 2) Test a new compensation formula for reducing variability in SOS measurements. **METHODS:** Data from 73 women were acquired using a GE Lunar Achilles Insight bone sonometer. Radio frequency data were processed off-line using both PI and PS algorithms. **RESULTS:** BUA measurements (mean  $\pm$  sd) were  $81.4 \pm 21.4$  dB/MHz (PS) and  $67.2 \pm 9.7$  dB/MHz (PI). Compensation of SOS measurements reduced 1) average transit-time-marker-related SOS variability by 75% in 73 women and 2) bandwidth-related SOS variability by 80% in a bone-mimicking phantom. **CONCLUSION:** These new methods will enable a substantial improvement in consistency in bone sonometry. The mention of commercial products, their sources, or their use in connection with material reported herein is not to be construed as either an actual or implied endorsement of such products by the Department of Health and Human Services.

3:20

**4pBBa5. Characterization of Ultrasound Propagation Through Ex-vivo Human Temporal Bone.** Azzdine Y. Ammi (University of Cincinnati, Biomedical Engineering, MSB, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, USA, azzdine.ammia@uc.edu), Douglas T. Mast (University of Cincinnati, Biomedical Engineering, MSB, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, USA, masttd@email.uc.edu), I-Hua Huang (1415 2nd Aveue, Unit 1804, Seattle, WA 98101, USA, helloihua@gmail.com), Todd A. Abruzzo (University of Cincinnati, Dept.s of Radiol., Neurosurg. & Biomed. Engineer., Cincinnati, OH 45267,

USA, Todd.Abruzzo@Healthall.com), Constantin C. Coussios (University of Oxford, Medical Engineering Unit, 43 Banbury Road, OX2 6PE Oxford, UK, constantin.coussios@eng.ox.ac.uk), George J. Shaw (University of Cincinnati, Biomedical Engineering, MSB, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, USA, SHAWGE@UCMAIL.UC.EDU), Christy K. Holland (University of Cincinnati, Biomedical Engineering, MSB, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, USA, Christy.Holland@uc.edu)

Knowledge of cranial and intracranial ultrasonic properties is essential for optimal results in brain vasculature imaging and therapy. The aims of this study were to perform measurements of the intracranial acoustic pressure field, to identify ultrasound parameters that maximize penetration and minimize beam aberration, and to estimate the speed of sound and the attenuation per unit length in the temporal bone (TB). In vitro experiments were conducted on five human skulls. In a water-filled tank, two unfocused (0.12 and 1.03 MHz) and one focused (2.00 MHz) transducers were consecutively placed near the TB of each skull. The acoustic pressure field was measured in a volume estimated to encompass the middle cerebral artery (MCA). For each measurement, the intracranial distance from the position of maximum acoustic pressure to the estimated MCA origin was quantified. The pressure reductions at these locations relative to the free field were also estimated. The intracranial -3 dB depth of field and beam width were investigated as a function frequency. The speed of sound in TB at 1.03 MHz was 1752.1 to 3285.3 m/s. This work provides quantitative information on the cranial and intracranial ultrasound properties, which are needed for optimal insonation of the brain vasculature.

### 3:40-5:20 Posters

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

## Contributed Papers

5:20

**4pBBa6. Estimation of femoral bone density from trabecular direct wave and cortical guided wave ultrasound velocities measured at the proximal femur in vivo.** Reinhard Barkmann (Medizinische Physik, Diagnostische Radiologie, Universitätsklinikum Schleswig-Holstein, Arnold-Heller-Str.9, 24105 Kiel, Germany, barkmann@rad.uni-kiel.de), Stefanie Dencks (Medizinische Physik, Diagnostische Radiologie, Universitätsklinikum Schleswig-Holstein, Arnold-Heller-Str.9, 24105 Kiel, Germany, dencks@rad.uni-kiel.de), Alexander Bremer (Medizinische Physik, Diagnostische Radiologie, Universitätsklinikum Schleswig-Holstein, Arnold-Heller-Str.9, 24105 Kiel, Germany, Alexander.Bremer@gmx.net), Pascal Laugier (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, laugier@lip.bhdc.jussieu.fr), Frederic Padilla (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, Frederic.Padilla@lip.bhdc.jussieu.fr), Kim Brixen (Odense University Hospital, Department of Endocrinology, 5000 Odense, Denmark, Kim.Brixen@ouh.regionsyddanmark.dk), Jesper Ryg (Odense University Hospital, Department of Endocrinology, 5000 Odense, Denmark, Jesper.Ryg@ouh.regionsyddanmark.dk), Claus C. Glüer (Medizinische Physik, Diagnostische Radiologie, Universitätsklinikum Schleswig-Holstein, Arnold-Heller-Str.9, 24105 Kiel, Germany, glueer@rad.uni-kiel.de)

Bone mineral density (BMD) of the proximal femur is a predictor of hip fracture risk. We developed a Quantitative Ultrasound (QUS) scanner for measurements at this site with similar performance (FemUS). In this study we tested if ultrasound velocities of direct waves through trabecular bone and of guided waves through cortical bone could be used to estimate BMD. In two centres, Kiel and Odense, we measured time-of-flight (TOF) of waves through the trabecular greater trochanter and cortical intertrochanter as well as a wave through soft tissue only. TOF was adjusted for leg width using ultrasound echoes reflected from the skin of the leg to yield speed-of-sound (SOS) of different wave components. Data were cross-calibrated and

pooled (62 women). Bivariate correlations and a multivariate model were calculated for the estimation of femur BMD. BMD correlated both with trabecular and cortical SOS but not soft tissue SOS. Coefficient of determination, percentage residual error (RMSE) and level of significance (p) were  $R^2=0.51$ , RMSE=12.6%,  $p<0.0001$  for trabecular and  $R^2=0.53$ , RMSE=12.3%,  $p<0.0001$  for cortical measurements. The combination of trabecular, cortical and soft tissue SOS improved the correlation to  $R^2=0.69$ , RMSE=10.4%,  $p<0.0001$ . Multiwave ultrasound methods allow estimation of femoral BMD with a low residual error.

5:40

**4pBBa7. Estimation of in vivo cancellous bone elasticity.** Takahiko Otani (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, totani@oyoe.jp), Isao Mano (Oyo Electric Co., Ltd., 63-1, Nakamichi-mote Hirakawa, 610-0101 Joyo, Japan, imano@oyoe.jp), Toshiyuki Tsujimoto (Horiba, Ltd., 2, Miyanohigashi-cho, Kisshoin, Minami-ku, 601-8510 Kyoto, Japan, toshiya.tsuji@horiba.com)

Effect of decreasing bone density (a symptom of osteoporosis) is greater for cancellous bone than for dense cortical bone, because cancellous bone is metabolically more active. Therefore, bone density or bone mineral density at cancellous bone is generally used to estimate the onset of osteoporosis. Elasticity or elastic constant is one of fundamental mechanical parameters and directly related to the mechanical strength of bone. Accordingly, elasticity is a preferable parameter to assess the fracture risk. A novel ultrasonic bone densitometer LD-100 has been developed to obtain mass density and elasticity of cancellous bone with a spatial resolution comparable to that of the peripheral quantitative computed tomography system. Bone mass density and bone elasticity are evaluated using ultrasonic parameters based on fast and slow waves in cancellous bone using a modeling of ultrasonic wave propagation path. Elasticity is deduced from measured bone mass density and propagation speed of fast wave. Thus, elasticity of cancellous bone is approximately expressed by a cubic equation of bone mass density.

6:00

**4pBBa8. Fracture and bone defect assessment using quantitative ultrasound wave propagation.** Mikey Szczupak (Stony Brook University, SUNY, Department of Biomedical Engineering, 350 Psychology-A Building, Stony Brook, NY 11794, USA, mszczupak90@yahoo.com), Yi Xia (Stony Brook University, SUNY, Department of Biomedical Engineering, 350 Psychology-A Building, Stony Brook, NY 11794, USA, xia.yi76@gmail.com), Jiqi Cheng (Stony Brook University, SUNY, Department of Biomedical Engineering, 350 Psychology-A Building, Stony Brook, NY 11794, USA, chengjiqi@yahoo.com), Yi-Xian Qin (Stony Brook University, SUNY, Department of Biomedical Engineering, 350 Psychology-A Building, Stony Brook, NY 11794, USA, yi-xian.qin@sunysb.edu)

Nondestructive evaluation of early fracture and monitoring its healing, particularly in non-typical fracture, is critical yet challenge. Quantitative ultrasound test along the long bone is largely dependant on bone density, gap size, the level of maturity of the callus, and mineralized status. We propose that the ultrasonic velocity (UV) is dependent on fracture gap size and the degree of porosity. The objective of the study was to develop an analytical and a multiple ultrasound sensing system for non-invasive fracture assessment. An analytical model was generated as a function between UV and the gap size in long bone. The ultrasound measurements were performed in the intact pig femur with controlled fracture sizes, e.g., 1mm, 3mm and 5mm. A serial ultrasound transducers were aligned 3 cm above bone with an angle approximately equal to the critical angle. The results indicated that the UV decreased with increase of fracture size in a linear manner ( $r=0.96$ ), in which UV decreased 6.7%, 16.3% and 26.8% for 1, 3 and 5 mm gaps, respectively. The simulation result significantly matched the experiment measurement ( $r=0.98$ ). The results demonstrated that quantitative ultrasound has potential to the diagnosis of fracture, monitoring fracture healing, and other bone disorders.

6:20

**4pBBa9. Monitoring of trabecular bone induced microdamage using a nonlinear wave-coupling technique.** Guillaume Renaud (LUSSI, 10 Bd Tonellé, 37032 Tours, France, guillaume.renaud28@etu.univ-tours.fr), Samuel Callé (LUSSI, 10 Bd Tonellé, 37032 Tours, France, calle\_s@med.univ-tours.fr), Jean-Pierre Remenieras (LUSSI, 10 Bd Tonellé, 37032 Tours, France, remenier@med.univ-tours.fr), David Mitton (Laboratoire de biomécanique, 151, Boulevard de l'hôpital, 75013 Paris, France, david.mitton@paris.ensam.fr), Julie Blanchi (Laboratoire de biomécanique, 151, Boulevard de l'hôpital, 75013 Paris, France, julieblanchi@gmail.com), Marielle Defontaine (LUSSI, 10 Bd Tonellé, 37032 Tours, France, defontai@med.univ-tours.fr)

Bone tissue contains microcracks which may affect its mechanical properties as well as the whole trabecular structure. The relationship between crack density and bone strength is nevertheless poorly understood. Efficient nonlinear (NL) ultrasound methods have been widely developed for nondestructive testing and geophysical applications to detect microdamage. Moreover it has been reported that elastic nonlinearities increase with induced damage. We propose to monitor trabecular bone microdamage using a NL wave coupling technique. Ultrasonic short bursts times of flight (TOF) are modulated as result of NL interaction with a low-frequency (LF) wave in the medium. TOF modulation (TOFM), or propagation velocity variations, are directly related to NL elasticity variations. This technique allows measuring the instantaneous TOFM as a function of the LF pressure. It is thus possible to analyze separately elasticity variations in tension and in compression, and to distinguish the tension to compression phase from the compression to tension phase (hysteresis). In several trabecular bone samples, different TOFM amplitudes in tension and in compression are observed, probably due to microdamage. For increasing damage levels progressively induced by quasi-static compression testing, linear and nonlinear ultrasound parameters are compared to biomechanical parameters.

6:40

**4pBBa10. Local measurements of the diffusion constant in multiple scattering media: Application to human trabecular bone imaging.** Alexandre Aubry (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, alexandre.aubry

@espci.fr), Arnaud Derode (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, arnaud.derode@ujf-grenoble.fr), Frederic Padilla (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, Frederic.Padilla@lip.bhdc.jussieu.fr)

Multiple scattering of ultrasound can occur in trabecular bones. A typical signature of multiple scattering is the coherent backscattering effect, and the average trajectory of a wave undergoing multiple scattering can be characterized by the diffusion constant  $D$ . In this work, we present local measurements of the diffusion constant for ultrasonic waves in human trabecular bone, based on near-field observation of the backscattered intensity. The experimental set up consists in an array of programmable transducers placed in front of a slice of bone that we want to image. By achieving Gaussian beam-forming both at emission and reception, an array of virtual sources and receivers located in the near-field is constructed. The time evolution of the incoherent component of the intensity backscattered on this virtual array is shown to represent directly the growth of the diffusive halo as  $(Dt)^{1/2}$ . A matrix treatment is proposed to separate the incoherent intensity from the coherent backscattering peak. Once the incoherent contribution is isolated, local measurements of the diffusion constant  $D$  are achieved around 3 MHz and then a  $D$ -map of the bone is built. These measurements are shown to be strongly correlated with the bone mineral density.

7:00

**4pBBa11. Relationship Between Mechanical Properties and Acoustic Parameters Obtained from Fast and Slow Waves for Cancellous Bone.**

Masaki Harada (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, dth0315@mail4.doshisha.ac.jp), Kazuto Tanaka (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, ktanaka@mail.doshisha.ac.jp), Tsutao Katayama (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, tkatayam@mail.doshisha.ac.jp), Katsunori Mizuno (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, dtg0151@mail4.doshisha.ac.jp), Hiroki Soumiya (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp), Mami Matsukawa (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp)

The bone-density measuring equipment using the characteristics of the ultrasound has been applied for the diagnosis of osteoporosis. The propagated ultrasonic wave in the cancellous bone changes dramatically due to its complicated structure, and is separated into fast wave and slow wave. The parameters obtained from two-waves are closely related to the structure properties and the elastic properties of bone, which is considered to provide important information related to bone strength. However, the relationship between the strength obtained by the mechanical tests and the fast and slow wave properties is not clearly investigated. In this study, compression tests of the cancellous bovine bone were conducted to determine the compressive strength in addition to the ultrasonic measurements. Our results showed that amplitude ratio of fast and slow wave significantly correlated ( $r^2=0.88$ ) with the compressive strength, which reasonably indicates that the bone strength increases in proportional to the density of the cancellous bone.

7:20

**4pBBa12. Two-dimensional simulation of circumferential wave propagation in cortical bone at the femoral neck: sensitivity to bone geometry.**

Julien Grondin (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, Julien.Grondin@lip.bhdc.jussieu.fr), Quentin Grimal (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, quentin.grimal@lip.bhdc.jussieu.fr), Pascal Laugier (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, laugier@lip.bhdc.jussieu.fr)

Assessment of bone status using ultrasound techniques has proved to be efficient to quantify the risk of osteoporotic fracture. A prototype scanner is currently developed (European FEMUS project) to assess bone quality at the hip. Circumferential waves in femoral neck cortical bone are expected to propagate in the prototype configuration. In this work, the relationships

between time of flight (TOF) of circumferential waves and femoral neck cross-section geometrical parameters were investigated. Two-dimensional finite-difference time-domain simulations of through transmission propagation of a plane wave at 0.5 MHz central frequency were performed on eight femoral neck cross-section models reconstructed from X-ray computed tomography data of one human femur. An ellipse with major radius (a) and minor radius (b) was fitted on the external circumference of each cross-section. The TOF was highly correlated to the ellipticity a/b ( $R = -0.9607$ ,  $p < 10^{-3}$ ) and to the area delineated by the endosteal surface ( $R = -0.9717$ ,  $p < 10^{-4}$ ). These results indicate that the TOF is sensitive to the shape of the femoral neck cross-section. This is interesting insofar as previous studies highlighted the importance of the relationships between geometrical parameters and bone strength. In future works, these 2-D results will be challenged in 3-D configurations.

7:40

**4pBBa13. Array transducer applied to low-frequency guided wave ultrasonography: An in vivo study on human radius and tibia.** Vantte Kilappa (University of Jyväskylä, Department of Physics, PO. Box 35, 40014 Jyväskylä, Finland, warma@iki.fi), Petro Moilanen (University of Jyväskylä, Department of Physics, PO. Box 35, 40014 Jyväskylä, Finland, pemoilan@cc.jyu.fi), Tianhui Chen (University of Jyväskylä, Department

of Health Sciences, PO. Box 35, 40014 Jyväskylä, Finland, cth006@hotmail.com), Hongqiang Ma (University of Jyväskylä, Department of Health Sciences, PO. Box 35, 40014 Jyväskylä, Finland, mhqsir0827@hotmail.com), Jussi Timonen (University of Jyväskylä, Department of Physics, PO. Box 35, 40014 Jyväskylä, Finland, Jussi.Timonen@phys.jyu.fi), Sulin Cheng (University of Jyväskylä, Department of Health Sciences, PO. Box 35, 40014 Jyväskylä, Finland, Sulin.Cheng@sport.jyu.fi)

Velocity ( $V_{FAS}$ ) of a first arriving signal for  $f < 0.5$  MHz is expected to have an enhanced sensitivity to endosteal osteoporotic changes as compared to using higher frequencies. In this study a group of males and females (aged 10-87 years) was measured by using a new array probe ( $f_c = 0.4$  MHz) on the radius and tibia. In addition, peripheral quantitative computed tomography was used to assess the bone mineral density (BMD) and cortical thickness (CTh) at the same bone locations. Initial results ( $n=57$ ) showed that the repeatability error of the  $V_{FAS}$  data ( $CV_{rms}$ ) was 0.5%. When  $V_{FAS}$  data for the radius was plotted by age it characterized well, yet better than BMD or CTh, the typical growth and loss curve of bone.  $V_{FAS}$  for the radius was strongly correlated with total BMD ( $r = 0.84$ ,  $p < 0.001$ ), cortical BMD ( $r = 0.77$ ,  $p < 0.001$ ), subcortical BMD ( $r = 0.66$ ,  $p < 0.001$ ) and CTh ( $r = 0.59$ ,  $p < 0.001$ ). Corresponding results for the tibia were similar but clearly weaker than those for the radius. In conclusion, the low-frequency  $V_{FAS}$  had good accuracy and it predicted well both the geometry and material properties throughout the cortex.

THURSDAY AFTERNOON, 3 JULY 2008

P3-B, LEVEL 3, 3:40 TO 5:20 P.M.

## Session 4pBBb

### Biomedical Ultrasound/Bioresponse to Vibration: Ultrasonic Characterization of Bone II (Poster Session)

Keith Wear, Cochair

*U.S. Food and Drug Administration, Center for Devices and Radiological Health*

Frederic Padilla, Cochair

*Université Paris 6*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pBBb1. Cross-sectional ultrasonic tomography of the medullary cavity of child bones - Limits of resolution.** Philippe Lasaygues (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, lasaygues@lma.cnrs-mrs.fr), Régine Guillermin (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, guillermin@lma.cnrs-mrs.fr), Eric Debieu (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, debieu@lma.cnrs-mrs.fr), Jean-Pierre Lefebvre (Laboratory for Mechanics and Acoustics CNRS, 31 chemin Joseph Aiguier, 13009 Marseille, France, lefebvre@lma.cnrs-mrs.fr), Philippe Petit (Pediatric radiology department, 'Timone' Children's Hospital, 13006 Marseille, France, philippe.petit@mail.ap-hm.fr)

In children with bone diseases, ultrasonography has proved to be a highly effective tool for assessing congenital disorders. However, with standard devices, this method of examination is limited and not suitable for

diagnostic large purposes as tumors or sub-periosteal infectious. Authors dealt with the imaging of more adult bones than children, and the main aim has usually been to assess the thickness of the diaphysis and/or to estimate the wave velocity crossing the structure. Our group has been focusing on the cross-sectional radial imaging process, using ultrasonic computed tomography, of child bones. Although this method is known to provide a potentially valuable means of imaging objects with similar acoustical impedance, problems arise when it is proposed to obtain quantitative tomograms of more highly contrasted media. Finding solutions involves either using non-linear schemes. In this paper, we recall the advantages and limitations of ultrasonic computed tomography methods when dealing with highly contrasted scatterers. The results obtained are promising and suggest that the geometrical and acoustical characteristics of children's bones can be efficiently determined using this ultrasonic computed tomography method.



**4pBBb2. Model study on assessment of bone by ultrasonic guided waves through a varying layer of soft tissue.** Alexey Tatarinov (Artann Laboratories, 1753 Linvale-Harbourton, Lambertville, NJ 08350, USA, tatarinov@artannlabs.com), Armen Sarvazyan (Artann Laboratories, 1753 Linvale-Harbourton, Lambertville, NJ 08350, USA, armen@artannlabs.com)

Propagation parameters of ultrasonic guided waves in long bones are sensitive to changes of the cortical thickness, which is one of bone parameters affected by osteoporosis. Meanwhile, the guided waves in bone can be masked by the longitudinal waves in the overlying soft tissues, especially in obese patients. The goal of this study was to explore a possibility to minimize the effect of soft tissue on quantitative assessment of propagation parameters of axial guided waves in long bones. Phantoms and animal bone fragments modeling the axial gradients of the cortical thickness in the human proximal tibia were used. The specimens were covered by a layer of soft tissue of varied thickness. Ultrasonic signals were acquired in the pulse mode at 0.1 MHz by the surface transmission. The specimens were scanned lengthwise and the acquired signals were plotted versus the scanned distance. Analysis of the obtained waveform profiles along the tested specimens allowed quantitative evaluation of the variations of the cortical thickness despite the presence of a 5-6 mm soft tissue layer. The ratio of the acoustic wavelength to the soft tissue layer thickness defines the level of the soft tissue effect.

**4pBBb3. Using Singular Value Decomposition to analyse a low frequency contribution on human cortical bone with a 1MHz axial transmission probe.** Magali Sasso (Echosens, R&D department, 153 avenue d'Italie, 75013 Paris, France, magali.sasso@echosens.com), Maryline Talmant (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, talmant@lip.bhdc.jussieu.fr), Guillaume Haiat (CNRS, Laboratoire de Recherches Orthopédiques, 10, Avenue de Verdun, 75010 Paris, France, haiat@univ-paris12.fr), Pascal Laugier (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, laugier@lip.bhdc.jussieu.fr), Salah Naili (Université Paris 12, B2OA, 61, Avenue du Général de Gaulle, 94010 Créteil, France, naili@univ-paris12.fr)

The configuration of axial transmission technique dedicated to cortical bone characterization generates multiple contributions, associated with different propagation modes. The first arriving signal velocity is classically evaluated using time-of-flight measurements while the analysis of later arrivals requires the development of specific signal processing tools. We focus here on an Energetic Low Frequency (ELF) later contribution acquired by a 1-MHz multi-element bi-directional probe devised in the LIP. Using a procedure adapted from the Singular Value Decomposition (SVD), the ELF contribution was separated from the rest of the signal. The ability of the method to provide an accurate phase velocity estimate of a dispersive wave was established in a controlled-case study on the propagation of Lamb waves on plates using FDTD (Finite-Difference Time-Domain) simulations. The method applied on signals acquired previously in vitro on human radii using the bidirectional device show that the ELF is consistent with the A0 Lamb mode and that its velocity is related to cortical bone thickness ( $R^2 = 0.6$ ,  $p < 10^{-5}$ ). Identification of A0 type wave agree with results reported in the literature obtained with a mono-element device which operates in the 100-300 kHz frequency band.

**4pBBb4. Variability of velocities provided by axial transmission due to irregular geometry of cortical bones.** Thiên-Ly Pham (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, pham@lip.bhdc.jussieu.fr), Maryline Talmant (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, talmant@lip.bhdc.jussieu.fr), Pascal Laugier (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, laugier@lip.bhdc.jussieu.fr)

Early predictions of the velocities of axially transmitted waves on cortical bones were based on waveguides such as plate and tube of regular geometry and uniform thickness. We investigate the role of the actual irregular geometry by means of numerical simulations comparing monodirectional

and bidirectional multielement modalities. Propagation was simulated using a finite difference method in a set of human radii previously examined by a bidirectional device. Individual geometry of the samples was reconstructed from X Ray tomography (pixel=100mm). The material constituting the bone models was considered to be a transverse isotropic medium with chosen fixed elastic properties taken from literature. In addition simulations were performed on plate of either constant or variable thickness. Preliminary results performed on 9 samples show that bidirectional technique reduces the variability of the velocities of axially transmitted waves compared to monodirectional array when irregular geometries are involved. Whereas bidirectional velocity is significantly correlated to the velocity obtained in plates whose thickness is equal to the mean thickness of the specimen ( $r^2 = 0.77$ ,  $p = 0.0018$ ,  $RMSE = 29\text{m/s}$ ), no significant correlation is found for monodirectional velocity. The effect is mainly attributed to the irregular external surface of real samples.

**4pBBb5. A finite element model of the lamellar osteonal structure based on ultrahigh frequency acoustic impedance data.** Kay Raum (Martin Luther University of Halle-Wittenberg, Dept. of Orthopedics, Q-BAM Group, Magdeburger Str. 22, 06097 Halle, Germany, kay.raum@medizin.uni-halle.de), Quentin Grimal (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, quentin.grimal@lip.bhdc.jussieu.fr), Alf Gerisch (Martin Luther University of Halle-Wittenberg, Institut für Mathematik, 06099 Halle, Germany, alf.gerisch@medizin.uni-halle.de)

Materials: A finite element (FE) model was developed, in which the osteon is considered to consist of a central Haversian canal filled with an incompressible fluid and surrounding sets of lamellar units. Each lamellar unit was further subdivided in five sublayers, whereas the orientation of the symmetry axis between adjacent sublayers was shifted clockwise. A sublayer consists of one to ten layers of parallel oriented mineralized collagen fibrils (thickness:  $0.2\text{ }\mu\text{m}$ , constant transverse isotropic stiffness tensor). Results: A variation of the sublayer thicknesses results in either isotropic or anisotropic tissue compound properties. By changing the individual layer thicknesses various degrees of anisotropy could be produced. A good agreement with the lamellar pattern obtained in 1.2-GHz SAM images as well as with the anisotropic elastic coefficients measured at the tissue level (50-MHz ultrasound) was obtained by choosing an asymmetric lamellar unit. Conclusions: With the proposed combination of experimentally derived microelastic and microstructural data by 1.2 GHz SAM and a micromechanical FE the homogenized elastic stiffness tensor of lamellar bone tissue was derived. The tissue anisotropy was explained by the asymmetric twisted plywood structure.

**4pBBb6. Distribution and anisotropy of fast wave speed in the cancellous bone of bovine femur.** Katsunori Mizuno (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, dtg0151@mail4.doshisha.ac.jp), Hiroki Soumiya (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp), Mami Matsukawa (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp), Takahiko Otani (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, totani@oyoe.jp), Masahiko Takada (Omihachiman Community Medical Center, 1379 Tsuchida-cho, 523\*0082 Omihachiman, Japan, takada@design.kyushu-u.ac.jp), Isao Mano (Oyo Electric Co., Ltd., 63-1, Nakamichiomote Hirakawa, 610-0101 Joyo, Japan, imano@oyoe.jp), Toshiyuki Tsujimoto (Horiba, Ltd., 2, Miyano Higashi-cho, Kisshoin, Minami-ku, 601-8510 Kyoto, Japan, toshi yuki.tsujimoto@horiba.com)

The ultrasonic longitudinal wave in the cancellous bone is separated into two waves, fast and slow waves [1]. In this study, the relationship between the fast wave speed and the cancellous bone structure is experimentally investigated. A conventional ultrasonic pulse measurement was performed using a PVDF focus transmitter (Custom made, Toray) and a self-made PVDF receiver. Cylindrical specimens of cancellous bone were taken from the head of bovine femur in the distal part, along the three orthogonal directions. The distribution of fast wave speed was obtained by changing the measurement position along the cylindrical axis. The anisotropy of speed was also investigated by rotating the specimens. The structural parameters



of each specimen were also measured by X-ray micro CT (MCT-12505MF, Hitachi), which gave us the trabecular length and alignment from MIL (mean intercept length) parameters through TRI/3D-Bon software (Ratoc). We found that the fast wave showed large distribution and strong anisotropy depending on the measurement positions and wave propagation directions in the specimens. The fast wave showed the maximum speed in case of wave propagation along the load direction. Reference [1] A. Hosokawa and T. Otani, J. Acoust. Soc. Am., 101, 558 (1997).

**4pBBb7. Ultrasonic wave properties in the bone axis direction of bovine cortical bone.** Kazufumi Yamamoto (Orthopaedic Surgery, Hamamatsu University School of Medicine, 1-20-1 Handayama Higasi-ku, 431-3192 Hamamatsu, Japan, kazyama2323@hotmail.com), Yuichiro Yaoi (Doshisha University, 1-3, Tataru Miyakodani, 610-0321 Kyotanabe, Japan, yuichiro\_y0306@hotmail.co.jp), Yu Yamato (Orthopaedic Surgery, Hamamatsu University School of Medicine, 1-20-1 Handayama Higasi-ku, 431-3192 Hamamatsu, Japan, kazyama2323@hotmail.com), Takahiro Yanagitani (Graduate school of engineering, Tohoku University, 28 Kawau-chi Aoba-ku, 980-8579 Sendai, Japan, mmatsuka@mail.doshisha.ac.jp), Takaaki Koizumi (Doshisha University, 1-3, Tataru Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp), Mami Matsukawa (Doshisha University, 1-3, Tataru Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp), Kaoru Yamazaki (Orthopaedic Surgery, Hamamatsu University School of Medicine, 1-20-1 Handayama Higasi-ku, 431-3192 Hamamatsu, Japan, kazyama2323@hotmail.com), Akira Nagano (Orthopaedic Surgery, Hamamatsu University School of Medicine, 1-20-1 Handayama Higasi-ku, 431-3192 Hamamatsu, Japan, kazyama2323@hotmail.com)

Quantitative ultrasound (QUS) is a good method to measure elastic properties of bone (one indicator of bone quality) in vivo. Bovine cortical bone has two typical microstructures, plexiform and Haversian. In the nanoscopic level, bone consists of calcium phosphate, which forms incomplete hydroxyapatite (HAP) crystal. The preferred orientation of c-axis of HAP crystallites induces anisotropy and inhomogeneity of elastic properties in bone. In this study, relationship between speed of sound (SOS) and HAP crystallites orientation in the axial direction were investigated in two foreign age bovine cortical bones. The dependence of attenuation on the anatomical position was also investigated. Two ring shaped cortical bone samples were made from 36 and 24-month-old bovine femur. SOS was measured by a conventional ultrasonic pulse system, using self-made PVDF transducers. The integrated intensity of (0002) peak obtained using X-ray diffraction was estimated to evaluate the amount of preferred orientation. Regardless of age, a significant correlation between SOS and preferred orientation of HAP crystallite was observed in the parts of the plexiform structure, and gradient of the relation showed a similar tendency. Attenuation seemed to strongly depend on bone microstructure because of its porosity.

**4pBBb8. Anisotropy of ultrasonic longitudinal wave in the cortical bone of bovine femur.** Yuichiro Yaoi (Doshisha University, 1-3, Tataru Miyakodani, 610-0321 Kyotanabe, Japan, yuichiro\_y0306@hotmail.co.jp), Kazufumi Yamamoto (Orthopaedic Surgery, Hamamatsu University School of Medicine, 1-20-1 Handayama Higasi-ku, 431-3192 Hamamatsu, Japan, kazyama2323@hotmail.com), Takaaki Koizumi (Doshisha University, 1-3, Tataru Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp), Mami Matsukawa (Doshisha University, 1-3, Tataru Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp), Kaoru Yamazaki (Orthopaedic Surgery, Hamamatsu University School of Medicine, 1-20-1 Handayama Higasi-ku, 431-3192 Hamamatsu, Japan, kazyama2323@hotmail.com), Akira Nagano (Orthopaedic Surgery, Hamamatsu University School of Medicine, 1-20-1 Handayama Higasi-ku, 431-3192 Hamamatsu, Japan, kazyama2323@hotmail.com)

Quantitative ultrasound (QUS) is a good method to measure elastic properties of bone. It is known that the mammalian cortical bone shows strong anisotropy and inhomogeneity. We have investigated the distribution of ultrasonic longitudinal wave properties in bovine femur, considering the

structure in the nanoscopic level [Yamato et al. Calcified Tissue International, Accepted]. In this study, the anisotropy in the axial-tangential plane of bovine cortical bone is experimentally investigated using an ultrasonic pulse technique. The ultrasonic pulse measurement was performed using a PVDF focus transmitter (Custom made, Toray) and a self-made flat PVDF receiver. Three ring-shaped cortical bone samples were made from a 32-month-old bovine femur. Four cylindrical specimens were taken from one ring-shaped cortical bone sample along the radial direction. The anisotropy of speed was investigated by rotating the specimens. We found that directivities of ultrasonic longitudinal wave in these specimens were similar. In addition, the direction of the fastest wave speed was a little different from the bone axis. The results indicate the complicated anisotropy of the cortical bone.

**4pBBb9. Ultrasonic velocity dispersion in bovine cortical bone.** Guillaume Haiat (CNRS, Laboratoire de Recherches Orthopédiques, 10, Avenue de Verdun, 75010 Paris, France, haiat@univ-paris12.fr), Magali Sasso (Echosens, R&D department, 153 avenue d'Italie, 75013 Paris, France, magali.sasso@echosens.com), Salah Naili (Université Paris 12, B2OA, 61, Avenue du Général de Gaulle, 94010 Créteil, France, naili@univ-paris12.fr), Kazufumi Yamamoto (Orthopaedic Surgery, Hamamatsu University School of Medicine, 1-20-1 Handayama Higasi-ku, 431-3192 Hamamatsu, Japan, kazyama2323@hotmail.com), Mami Matsukawa (Doshisha University, 1-3, Tataru Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp)

The evaluation of cortical bone quality has become possible in clinical practice, but the interaction between a broadband ultrasonic pulse and this complex multiscale medium remains poorly understood. Specifically, the frequency dependence of phase velocity has been sparsely investigated. This study aims at evaluating the determinants of the frequency dependence of phase velocity in bovine femoral cortical bone samples using an in vitro ultrasonic transmission device. Phase velocity is shown to vary quasi linearly in a 1 MHz restricted bandwidth around 4 MHz, which enables dispersion evaluation. Axial dispersion is significantly higher than radial and tangential dispersions. Significant differences in dispersion are obtained according to the anatomical location. The microstructure of each sample is determined using an optical microscope, which allows assessing the dependence of dispersion on the type of bone microstructure. Mostly positive, but also negative values of dispersion are measured. Negative dispersion is obtained mostly in samples constituted of mixed microstructure, which may be explained by phase cancellation effects due to the presence of different microstructures within the same sample. Dispersion is shown to be related to broadband ultrasonic attenuation values, especially in the radial direction. This dependence is compared with results derived from the local Kramers-Krönig relationships.

**4pBBb10. Diffuse ultrasonic backscatter from cortical bone.** Joseph A. Turner (University of Nebraska, Dept. of Engineering Mechanics, W317.4 NH, Lincoln, NE 68588, USA, jaturner@unl.edu), Goutam Ghoshal (University of Nebraska, Dept. of Engineering Mechanics, W317.4 NH, Lincoln, NE 68588, USA, goutamghoshal@rediffmail.com)

Diffuse ultrasonic backscatter techniques have been used primarily for characterization of microstructure in structural materials such as polycrystalline metals. Such measurements exploit the spatial variance of the signals from a modified pulse-echo technique. In this presentation, experiments are discussed using this technique on samples of porcine cortical bone at center frequencies of 15, 20, and 25 MHz. The time domain results obtained are analyzed with respect to a single-scattering model that includes statistical information about the microstructure. In addition, the model includes a rigorous description of the transducer beam pattern as it interacts with the liquid-sample. These results provide information regarding the ability of single-scattering models to capture the ultrasound propagation in such materials.

**4pBBb11. Effects of bone marrow on the ultrasonic propagation in the cancellous bone - Comparative study on experiment and simulation.** Takashi Saeki (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, dth0915@mail4.doshisha.ac.jp), Masahiro Emura (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, dth0915@mail4.doshisha.ac.jp), Katsunori Mizuno (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, dtg0151@mail4.doshisha.ac.jp), Mami Matsukawa (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp), Yoshiki Nagatani (Nara Medical University, 840 Shijo-cho, 634-8522 Kashihara, Japan, naramed-u@nagatani.ne.jp)

Longitudinal ultrasonic wave in cancellous bone separates into fast and slow waves depending on the bone structure. This phenomenon seems useful for the diagnosis of osteoporosis. In this study, we have investigated the influences of soft tissue (bone marrow) in the cancellous bone on the propagation of waves, in order to investigate the mechanism of this phenomenon. First, we have experimentally investigated the temperature dependence of longitudinal wave velocity and attenuation in bovine bone marrow, using a conventional ultrasonic pulse method. We used the ultrasonic wave at 1MHz. Then, we simulated the wave propagation in cancellous bone. For simulation, we used the 3 dimensional elastic FDTD (Finite Difference Time Domain) method. Here, we used the X-ray CT pictures of actual cancellous bone obtained from the head of left bovine femur as the bone model. By changing the velocity and the attenuation values in the soft tissue among trabeculae from those of marrow to the water, we have found the changes in the waveforms of both fast and slow waves. This indicates the changes in both wave properties, due to the properties of soft tissue.

**4pBBb12. Measurement of the tortuosity and the viscous characteristic length of human cancellous bone via transmitted waves.** Naima Sebaa (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, naima.sebaa@univ-lemans.fr), Laurens Boeckx (Lab. ATF, Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, laurens.boeckx@fys.kuleuven.be), Zine Fellah (CNRS-Laboratoire de Mécanique et d'Acoustique, 31 Chemin Joseph Aiguier, 13402 Marseille, France, fellah@lma.cnrs-mrs.fr), Mouna Naas (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, mouna.naas.etu@univ-lemans.fr), Tom L. Goossens (Technische Universiteit Eindhoven, Den Dolech 2, 5600 MB Eindhoven, Netherlands, tomgoos@gmail.com), Steven Boonen (Afdeling Geriatrie, UZ Herestraat, 3000 Leuven, Belgium, Steven.Boonen@med.kuleuven.be), Georges Van Der Perre (Afd. Biomechanica en Grafisch Ontwerpen, Celestijnenlaan 300, 3000 Leuven, Belgium, Georges.VanderPerre@mech.kuleuven.be), Philippe Leclaire (Lab. de Recherche en Mécanique et Acoustique, I.S.A.T - Univ. de Bourgogne, 49, rue Madoiselle Bourgeois, 58000 Nevers, France, philippe.leclaire@gmail.com), Claude Depollier (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, cclaude.depollier@univ-lemans.fr), Walter Lauriks (Lab. ATF, Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, Walter.Lauriks@fys.kuleuven.be)

For a few years several authors have proposed the model of Biot to describe the ultrasonic wave propagation in cancellous bone. One drawback of this model is the number of parameters which it needs. Two significant parameters to describe the geometry of the trabeculae are tortuosity and viscous characteristics length. In this communication, we propose two methods to measure these parameters. The first method consists in estimating these two parameters by inversion from the transmission coefficient. In the second method, we get the tortuosity from direct measurement using the focused transducers. The scan of a layer of cancellous bone shows a good correlation between direct measurements and the results of the inversion.

**4pBBb13. Simulation of ultrasound wave propagation through trabecular bone samples with and without bone marrow.** Liesbet Goossens (Afdeling Biomechanica en Grafisch Ontwerpen, Departement Werktuigkunde, Katholieke Universiteit Leuven, Afd. Biomechanica en Grafisch Ontwerpen, Celestijnenlaan 300, 3000 Leuven, Belgium, liesbet.goossens@mech.kuleuven.be), Jef Vanderroost (K.U.Leuven, Celestijnenlaan 300c, bus 2419, 3001 Heverlee, Belgium, jef.vanderroost@mech.kuleuven.be), Siegfried Jaecques (Department of Dentistry, Oral Pathology and Maxillo-Facial Surgery, BIOMAT Research Cluster, Katholieke Universiteit Leuven, Kapucijnenvoer 7 - bus 7001, 3000 Leuven, Belgium, siegfried.jaecques@med.kuleuven.be), Steven Boonen (Afdeling Geriatrie, UZ Herestraat, 3000 Leuven, Belgium, Steven.Boonen@med.kuleuven.be), Jan D'Hooge (Division of Cardiovascular Imaging and Dynamics, Katholieke Universiteit Leuven, UZ Herestraat 49 - bus 7003, 3000 Leuven, Belgium, jan.dhooge@uz.kuleuven.be), Walter Lauriks (Lab. ATF, Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, Walter.Lauriks@fys.kuleuven.be), Georges Van Der Perre (Afd. Biomechanica en Grafisch Ontwerpen, Celestijnenlaan 300, 3000 Leuven, Belgium, Georges.VanderPerre@mech.kuleuven.be)

For the clinical assessment of osteoporosis, ultrasound has been proposed as an alternative or supplement to the Dual-Energy X-ray Absorptiometry technique. However the interaction of ultrasound waves with (trabecular) bone remains relatively poorly understood. The aim of the present study was to improve the understanding of this interaction by simulating ultrasound wave propagation in fifteen trabecular bone samples from the human lumbar spine, using  $\mu$ CT based Finite Elements Modelling. The model included only the solid bone, without the bone marrow. Two structural parameters were calculated: the bone volume fraction (BV/TV) and the structural (apparent) elastic modulus ( $E_s$ ), and the ultrasound parameter Speed Of Sound (SOS). At 1 MHz, correlations between SOS and the parameters BV/TV and  $E_s$  were rather weak but the results can be explained from the specific features of the trabecular structure and the intrinsic material elastic modulus  $E_t$ . The correlation found between the simulated SOS values and those calculated from the simple bar equation was poor when the three directions are considered separately. However at lower frequencies (50-300 kHz), this correlation significantly improved. Currently we investigate the correlations between SOS and the structural parameters when the bone marrow is included in the FE model.

**4pBBb14. Modeling the ultrasonic scattering in trabecular bone.** Andrzej Nowicki (Institute of Fundamental Technological Research, Swietokrzyska 21, 00-049 Warsaw, Poland, anowicki@ippt.gov.pl), Jerzy Litniewski (Institute of Fundamental Technological Research, Swietokrzyska 21, 00-049 Warsaw, Poland, jlitn@ippt.gov.pl)

Computer simulations conducted to investigate the properties of the ultrasound pulse-echo signal, as it is received on the transducer surface after scattering in trabecular bone was developed. It can be used to yield an ideal environment in which, the effects of various parameters (scatterer mechanical and geometrical properties, scatterers' concentration), the shape of incident wave and experimental conditions influencing the scattering of ultrasonic waves in trabecular bone structure can be examined individually. The Wear's scattering model of a cancellous bone was applied with modifications that allow for changes of mechanical and geometrical properties of individual trabeculae as well as their spatial density variation. The model also enables considering the groups of scatterers with varying mean values, e.g. thick and thin trabeculae of cancellous bone. Also, the real interrogating pulses are considered, thus the pulse shape, the emitted field structure and the frequency transfer function of the transmitting-receiving transducer are applied in simulations. The results proved that the computer simulation is a useful tool for gaining a better understanding of the scattering of ultrasonic waves in biological tissue and has a particular relevance in studying scattering in cancellous bone which may be approximated as a collection of cylindrical trabeculae.

**4pBBb15. Dual frequency ultrasound technique enables determination of soft tissue composition and improves reliability of *in vivo* ultrasound bone densitometry.** Janne P. Karjalainen (University of Kuopio, POB 1627, 70211 Kuopio, Finland, janne.karjalainen@uku.fi), Juha Töyräs (Kuopio University Hospital, POB 1777, 70211 Kuopio, Finland, Juha.Toyras@kuh.fi), Toni Rikkinen (University of Kuopio, POB 1627, 70211 Kuopio, Finland, toni.rikkinen@uku.fi), Jukka Jurvelin (University of Kuopio, POB 1627, 70211 Kuopio, Finland, Jukka.Jurvelin@uku.fi), Ossi Riekkinen (University of Kuopio, POB 1627, 70211 Kuopio, Finland, Ossi.Riekkinen@uku.fi)

Soft tissues diminish reliability of the bone ultrasound backscatter measurements. In this study, the ability of a single broadband transducer dual frequency ultrasound (DFUS) technique to monitor the changes in soft tissue was investigated in a body builder during a 21 week training and dieting period, inducing a weight loss of 16.5 kg (18%). Then, DFUS was applied to correct the errors induced by soft tissues on the measurements of integrated reflection coefficient (IRC) in human distal femur. In DFUS, US reflection from soft tissue-bone interface is determined with two different US frequencies and, by knowing the frequency specific US attenuation and speed in adipose and lean tissues, their content can be determined. The dual energy X-ray absorptiometry (DXA) indicated that significant changes in quantity and composition of soft tissue, but not in bone density, took place during the diet. As compared with DXA, the single transducer DFUS could determine local soft tissue composition ( $r^2 = 0.88$ ,  $n=8$ ,  $p < 0.01$ ). The change in uncorrected IRC associated significantly with the change in body composition ( $r^2 = 0.56$ ,  $n=8$ ,  $p < 0.05$ ). The IRC values, corrected by DFUS, showed only minor variation ( $SD = \pm 1.26$  dB) during the diet.

**4pBBb16. Simulation and modeling of a new quantitative ultrasound imaging device using cylindrical crossed beam forming arrays.** Sylvain Hupert (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, sylvain.hupert@lip.bhdc.jussieu.fr), Djelloul Reguieg (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, reguieg@gmail.com), Frederic Padilla (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, Frederic.Padilla@lip.bhdc.jussieu.fr), Marielle Defontaine (LUSSI, 10 Bd Tonellé, 37032 Tours, France, defontai@med.univ-tours.fr), Pascal Laugier (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, laugier@lip.bhdc.jussieu.fr)

Real-time quantitative ultrasound imaging (QUI) produces images of bone properties with the advantages of being non-ionizing, portable and repeatable. A real-time QUI technique has been proposed, based on two 2-D arrays (24x24 transducer elements) to produce a confocal spherical focusing in transmit and receive modes. However, the electronics to manage beam scanning and focusing is complex and remains expensive. To overcome these disadvantages, a new device has been developed, based on two 1-D transducer arrays (72x1 transducer elements) and confocal cylindrical

crossed-beam forming. The intersection of two orthogonal cylindrical focus lines replaces the spherical focused beams. Initial results obtained with this approach showed a distortion and a shift of the spectrum in the low frequency range compared to the reference confocal spherical focusing technique. The aim of the study was to explain the observed differences between spherical and cylindrical focusing techniques using numerical simulations of elementary impulse responses calculated in the confocal and reception planes. The orthogonal configuration of the transmit and receive focusing line results in orthogonal curvatures of the transmitted wavefront and receiving aperture, with summation by the receiving aperture of an out-of-phase wavefront. We show that this effect is the major source for the frequency response artifacts.

**4pBBb17. Longitudinal assessment of human bone quality using scanning confocal quantitative ultrasound.** Yi-Xian Qin (Stony Brook University, SUNY, Department of Biomedical Engineering, 350 Psychology-A Building, Stony Brook, NY 11794, USA, yi-xian.qin@sunysb.edu), Yi Xia (Stony Brook University, SUNY, Department of Biomedical Engineering, 350 Psychology-A Building, Stony Brook, NY 11794, USA, xia.yi76@gmail.com), Wei Lin (Stony Brook University, SUNY, Department of Biomedical Engineering, 350 Psychology-A Building, Stony Brook, NY 11794, USA, wei.lin@sunysb.edu), Jiqi Cheng (Stony Brook University, SUNY, Department of Biomedical Engineering, 350 Psychology-A Building, Stony Brook, NY 11794, USA, chengjiqi@yahoo.com), Jesse Muir (Stony Brook University, SUNY, Department of Biomedical Engineering, 350 Psychology-A Building, Stony Brook, NY 11794, USA, jmuir@ic.sunysb.edu), Clint Rubin (Stony Brook University, SUNY, Department of Biomedical Engineering, 350 Psychology-A Building, Stony Brook, NY 11794, USA, crubin@sunysb.edu)

Microgravity and aging induced bone loss is a critical skeleton complication occurred particularly in the weight-supporting skeleton, which leads to osteoporosis and fracture. Advances in quantitative ultrasound (QUS) provide a unique method for evaluating both bone strength and density. Using an imaging-base confocal scanning ultrasound diagnostic system (SCAD), the goals of this work were to non-invasively characterize bone quality at proximal femur, and longitudinally monitor effectiveness of calcaneus bone loss in a 90-day bedrest. QUS scanning was performed at proximal femur (cadaver) and calcaneus (bedrest subjects) regions with QUS images of 80x80 mm<sup>2</sup> for hip and 40x40 mm<sup>2</sup> for calcaneus. QUS was processed to calculate the ultrasound attenuation (ATT; dB), wave ultrasound velocity (UV), and the broadband ultrasound attenuation (BUA; dB/MHz). Human cadaver proximal femurs have been measured with the SCAD, micro-CT, DXA, and mechanical strength test. Human calcaneus of bedrest subjects were measured using SCAD and DXA in day 0 (baseline), day 60 and day 90. Results demonstrated that QUS measurement has the capability to predict bone BMD, microstructure and mechanical properties in human bone, and indicated significant sensitivity to the progressive change of bone quality, particularly in the trabecular bone region with remodeling activities.



## Session 4pBBc

**Biomedical Ultrasound/Bioresponse to Vibration: Theoretical and Computational Models of Ultrasonic Propagation in Bones II (Poster Session)**

James Miller, Cochair  
Washington University

Pascal Laugier, Cochair  
Université Paris 6

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

*Contributed Papers*

**4pBBc1. Investigation of effect of trabecular microstructure on ultrasound propagation through cancellous bone using finite-difference time-domain simulations.** Atsushi Hosokawa (Akashi National College of Technology, 679-3 Nishioka, Uozumi, 674-8501 Akashi, Japan, hosokawa@akashi.ac.jp)

Using finite-difference time-domain (FDTD) numerical simulations, it was investigated how the trabecular microstructure could affect the propagation of ultrasound waves through cancellous bone. Three-dimensional (3D) numerical models of cancellous bone were reconstructed from 3D micro-computed tomography images of bovine femoral bone with oriented trabecular structure. In these models, the trabecular elements were eroded to increase porosity using an image processing technique. Three erosion procedures were given to realize different changes in the trabecular microstructure with increasing porosity. FDTD simulations of the ultrasound pulse waves propagating through the cancellous bone models were performed in two cases of the propagations parallel and perpendicular to the main trabecular orientation, and the porosity dependences of the propagation properties, attenuation and propagation speed, were derived for various trabecular changes. It was demonstrated from the simulated results that the propagation properties in both directions parallel and perpendicular to the trabecular orientation could be affected by the trabecular microstructure. In addition, the effects of the major (or plate-like) and minor (or rod-like) trabecular elements on the ultrasound propagation were respectively investigated for both the parallel and perpendicular propagations.

**4pBBc2. Excess ultrasonic attenuation due to inhomogeneities in porous media.** Mouna Naas (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, mouna.naas.etu@univ-lemans.fr), Naima Sebaa (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, naima.sebaa@univ-lemans.fr), Zine Fellah (CNRS-Laboratoire de Mécanique et d'Acoustique, 31 Chemin Joseph Aiguier, 13402 Marseille, France, fellah@lma.cnrs-mrs.fr), Mohamed Fellah (Laboratoire de Physique Théorique, Institut de Physique, USTHB, BP, 16111 Alger, Algeria, fellah1@yahoo.com), Walter Lauriks (Lab. ATF, Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium, Walter.Lauriks@fys.kuleuven.be), Claude Depollier (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, claude.depollier@univ-lemans.fr)

While Biot's theory seems well adapted to model the acoustical waves propagation in cancellous bone, some of its predictions do not agree with the experimental results. The excess of attenuation of the fast wave is one of

these discrepancies. In this paper we propose a modified Biot's model which takes into account the fluctuations of the physical parameters and their correlations. As a result of this model, we show that this excess of attenuation is due to several processes: i) classical Biot's attenuation, ii) scattering leading to the extension of the wave path, iii) mode conversion. Some comparison between experimental results and numerical simulations are proposed.

**4pBBc3. Relationship between QUS parameters and a cellular model-based estimation of bone strength.** Guillaume Haiat (CNRS, Laboratoire de Recherches Orthopédiques, 10, Avenue de Verdun, 75010 Paris, France, haiat@univ-paris12.fr), Frederic Padilla (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, Frederic.Padilla@lip.bhdc.jussieu.fr), Pascal Laugier (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, laugier@lip.bhdc.jussieu.fr)

The final goal of quantitative ultrasound (QUS) techniques is to assess bone strength. However, the translation of QUS results into bone strength remains elusive because the physical principles of ultrasonic propagation in bone are not fully understood yet. Here, the sensitivity of Broadband Ultrasonic Attenuation and Speed of Sound to variations of bone strength is derived. Therefore, a cellular model is combined to a multiple regression analysis resulting from the analysis of finite-difference time domain (FDTD) simulations coupled with imaging techniques. The variation of QUS variables induced by a variation of strength of 10%, realized either by a change in material properties or a change in bone volume fraction (BV/TV) is investigated. Except when BV/TV is high, the variations of BUA in response to a variation in strength realized by a pure change of BV/TV is higher than the technique imprecision and thus can be detected. When the variation of strength is realized by changes of elastic properties, the response in QUS properties is dominated by the variation in C11 over C44. The interpretation of these data, however, is not straightforward due to sparse description of elastic properties at the tissue level, which is a limitation of the cellular model.

**4pBBc4. Bulk waves velocities are dependent on frequency in cortical bone.** Cécile Baron (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, norabelic@yahoo.fr), Quentin Grimal (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, quentin.grimal@lip.bhdc.jussieu.fr), Maryline Talmant (Université



Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, talmant@lip.bhdc.jussieu.fr), Pascal Laugier (Université Paris 6, Laboratoire d'Imagerie Paramétrique, 15, rue de l'Ecole de Médecine, 75006 Paris, France, laugier@lip.bhdc.jussieu.fr)

Any model of ultrasound propagation in cortical bone requires the knowledge of the effective elastic properties of bone at, let's say, the mm scale. It is well known that the porous network of cortical bone interacts with ultrasonic waves and plays a major role in the mechanical behaviour. The purpose of the present work is to emphasize that, in cortical bone, there is some variation of the effective bulk wave velocities, and consequently of

the elastic properties, with frequency in the MHz range due to the effect of the porous network. A Finite Difference Time Domain (FDTD) code is used to simulate the ultrasonic propagation of compression and shear waves, transverse to, and along the bone axis, in cortical bone volumes reconstructed from micro-computed tomography (microCT). The resolution of the microCT data allowed to model the 3D networks of resorption cavities and Haversian pores. It is found that, for porosities typically above 5-6 %, the effective phase velocity is dependent on the frequency. Preliminary results indicate that the influence of the frequency is more important in the range 1-5 MHz.

THURSDAY AFTERNOON, 3 JULY 2008

P3-B, LEVEL 3, 3:40 TO 5:20 P.M.

### Session 4pBBd

## Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Light and Sound for Medical Imaging and Therapy II (Poster Session)

Ronald Roy, Cochair  
*Boston University*

Emmanuel Bossy, Cochair  
*Laboratoire Photons et Matière, ESPCI/CNRS*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pBBd1. Selective ultrasonic focusing towards an optical contrast agent by use of photoacoustic-guided time-reversal.** Arik R. Funke (Laboratoire Photons et Matière, ESPCI/CNRS, 10 rue Vauquelin, 75231 Paris Cedex 05, France, arik.funke@espci.fr), Emmanuel Bossy (Laboratoire Photons et Matière, ESPCI/CNRS, 10 rue Vauquelin, 75231 Paris Cedex 05, France, emmanuel.bossy@espci.fr), Jean-François Aubry (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, jean-francois.aubry@espci.fr), Mathias Fink (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, mathias.fink@espci.fr), Albert-Claude Boccara (Laboratoire Photons et Matière, ESPCI/CNRS, 10 rue Vauquelin, 75231 Paris Cedex 05, France, boccara@optique.espci.fr)

We recently demonstrated experimentally that optically absorbing regions embedded at centimetric depths in an optically diffusive medium can be used, through the photoacoustic effect, as ultrasound sources for time-reversal techniques [Bossy *et al.*, APL **89**(18), 2006]. If the optically absorbing region is spatially localized, the emission of the time-reversed photoacoustic waves provides a method to automatically focus ultrasound towards the absorbing region. In biological tissue, this approach has the potential to automatically guide high-intensity focused ultrasound (HIFU) beams towards a targeted region labelled with an optically absorbing contrast agent. However, such an approach is conceivable only if the HIFU beam is guided towards the sole targeted region, and not towards other potentially strong endogenous optical absorbers (such as blood vessels). In this work, we present a method to selectively focus time-reversed photoacoustic waves towards an exogenous wavelength-dependent optical contrast agent. Experiments were performed *in vitro* on tissue-mimicking phantoms, with nanoparticles as contrast agents. A tunable nanosecond pulsed laser was used with different optical wavelengths to generate photoacoustic waves. A multi-

channel time-reversal ultrasound system was used to emit ultrasound signals derived from time-reversed photoacoustic signals obtained with different optical wavelengths.

**4pBBd2. Point spread function of the limited-view array transducers in optoacoustic tomography: numerical and experimental studies.** Ivan Pelivanov (International Laser Center of Moscow State University, Vorob'yovy Gory, 1, 119992 Moscow, Russian Federation, pelivanov@ilc.edu.ru), Tatiana Khokhlova (International Laser Center of Moscow State University, Vorob'yovy Gory, 1, 119992 Moscow, Russian Federation, t\_khokhlova@ilc.edu.ru), Varvara Simonova (International Laser Center of Moscow State University, Vorob'yovy Gory, 1, 119992 Moscow, Russian Federation, simvar@mail.ru), Alexei Podolsky (International Laser Center of Moscow State University, Vorob'yovy Gory, 1, 119992 Moscow, Russian Federation, uzh@yandex.ru), Alexander Karabutov (Moscow State University, MSU, 1, building 2, GSP-2, Leninskiye Gory, 119992 Moscow, Russian Federation, akarabutov@gmail.com)

Optoacoustic (OA) tomography utilizes pulsed laser radiation for thermoelastic excitation of wideband ultrasonic pulses. The distribution of laser-induced heat release can be reconstructed if the ultrasonic pulses are detected by an array of transducers. The resolution provided by an array depends on various aspects of its design - the number of elements, element size, spacing and arrangement - and is determined by the point spread function (PSF) of the array. In present work PSFs of arrays with different characteristics are calculated using the Rayleigh integral for simulating the direct problem, and backprojection algorithm for solving the inverse problem of OA tomography. Universal dependencies of the resolution provided by an array on the dimensions and arrangement of its elements are found. Numerical studies are validated by experimental measurements of PSFs of different arrays.

**4pBBd3. A combined platform for b-mode and real-time optoacoustic imaging based on raw data acquisition.** Marc Fournelle (Fraunhofer IBMT, Ensheimer Strasse 48, 66386 Sankt Ingbert, Germany, marc.fournelle@ibmt.fhg.de), Kirsten Maass (Fraunhofer IBMT, Ensheimerstrasse 48, 66386 Sankt Ingbert, Germany, kirsten.maass@ibmt.fhg.de), Heinrich Fonfara (Fraunhofer IBMT, Ensheimerstrasse 48, 66386 Sankt Ingbert, Germany, heinrich.fonfara@ibmt.fhg.de), Hans-Joachim Welsch (Fraunhofer IBMT, Ensheimerstrasse 48, 66386 Sankt Ingbert, Germany, hansjoachim.welsch@ibmt.fhg.de), Holger J. Hewener (Fraunhofer IBMT, Ensheimer Strasse 48, 66386 Sankt Ingbert, Germany, holger.hewener@ibmt.fraunhofer.de), Christian Günther (Fraunhofer IBMT, Ensheimerstrasse 48, 66386 Sankt Ingbert, Germany, christian.guenther@ibmt.fhg.de), Robert M. Lemor (Fraunhofer IBMT, Ensheimer Strasse 48, 66386 Sankt Ingbert, Germany, robert.lemor@ibmt.fhg.de)

Optoacoustic imaging is a new promising medical imaging modality combining the benefits of optical and acoustical methods. Optoacoustics allow to make the high intrinsic optical contrast in biological tissue accessible to acoustical detection. In addition, the possibility of using nanoscaled contrast agents makes of optoacoustics an ideal candidate for molecular imaging. While optoacoustics are an emerging imaging modality with poor clinical experience, ultrasound is widely used for diagnosis. Accordingly, optoacoustic images are much harder to interpret than b-mode images. For this reason, we developed a hardware platform which allows combined b-mode and optoacoustic imaging using a 2-in-1 transducer with arrays of different frequency for the two modalities. The system supports simultaneous data acquisition of 128 channels with a sample rate of 80 MSamples allowing the usage of transducers with frequencies up to 20 MHz. The unprocessed data is transferred to a PC where the images are reconstructed with algorithms adapted to both modalities. A software for hardware control, data processing and visualization in real-time was developed. B-mode and optoacoustic images of tissue phantoms were generated and different types

of nanoparticles were used as optoacoustic contrast agent. Further, first in-vivo measurements underlying the high potentials of the combined system were obtained.

**4pBBd4. A maximum likelihood method for obtaining integrated attenuation from ultrasound transmission mode measurements.** Rene G. Willemlink (University of Twente, Signals and Systems Group (SaS), Faculty of Electrical Engineering, Mathematics and Computer Sciences (EWI), P.O. Box 217, 7500AE Enschede, Netherlands, G.H.Willemlink@utwente.nl), Srirang Manohar (University of Twente, Biophysical Engineering Group (BPE), Faculty of Science and Engineering (TNW), P.O. Box 217, 7500 AE Enschede, Netherlands, s.manohar@tnw.utwente.nl), Kees H. Slump (Univ. of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, c.h.slump@utwente.nl), Ferdi Van Der Heijden (Univ. of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, f.vanderheijden@utwente.nl), Ton G. Van Leeuwen (Univ. of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, a.g.j.m.vanleeuwen@tnw.utwente.nl)

In photoacoustic imaging applications an image is reconstructed of the optical absorption distribution the imaged object. The photoacoustic measurements however are also dependent on acoustic properties of the imaged object. By estimating the acoustic properties from photoacoustic measurements, we can improve the performance of the optical absorption reconstruction and allow for the imaging of separate acoustic modalities. We derive and evaluate a maximum likelihood estimator for the measurements of integrated acoustic attenuation. This estimator is applicable to media like soft tissue. In this kind of media, the attenuation due to dissipative effects obeys a frequency power law. By measuring the propagation of ultrasound signals through such a medium, the parameters that describe the attenuation can be estimated. In this paper a new method is introduced for estimating the attenuation of ultrasound media by means of transmission mode measurements. The method is based on analyzing the noise characteristics of the received signals and the formulation of a maximum likelihood estimator. The new estimator is compared to existing methods and was found to be a better estimator in terms of the RMS error than previous methods.

THURSDAY AFTERNOON, 3 JULY 2008

ROOM 353, 2:00 TO 7:40 P.M.

## Session 4pEAa

### Engineering Acoustics: Transducers I

Robert M. Koch, Chair

NUWC, 1176 Howell Street, Code 8232, Bldg. 1302, Newport, RI 02841-1708, USA

#### Contributed Papers

2:00

**4pEAa1. Analytical modeling of piezoelectric ceramic transducers based on coupled vibration analysis.** Boris Aronov (UMassD, 22 Seabeds Way, apt. 12, Needham, MA 02494, USA, baronov@comcast.net)

The energy method for analyzing piezoelectric ceramic transducers [B. S. Aronov, JASA, 117(1), 210-220.] is applied to the treatment of transducers with mechanical systems that can be considered as two-dimensional. Analysis is made following the general outline of the theory of coupled vibration in two degree-of-freedom systems and its extension to calculating the resonance frequencies of elastic bodies, as suggested by Gibbe and Blechshmidt [Ann. Physik, Ser. 5, 18, No. 5, 417-485 (1933)]. The approach to the problem is illustrated with examples of piezoelectric rectangular plates, stripes and thin-walled cylinders at various orientations relative to crystallographic coordinate system. For all of the examples the resonance frequencies and effective coupling coefficients are presented as functions of the dimensional aspect ratios. Equivalent electromechanical circuits are introduced that permit calculation of the transducers performance under dif-

ferent loading conditions. Practical recommendations for improving the electromechanical properties of the transducers by use of appropriate aspect ratios are discussed.

2:20

**4pEAa2. Resonance frequencies of the multilayered piezotransducers.** Francisco J. Arnold (CESET - Unicamp, R. Paschoal Marmo, 1888 - JD. Nova Italia, 13484332 Limeira, Brazil, arnold@ceset.unicamp.br)

Piezoelectric transducers used in high power ultrasonic applications are composed of piezoelectric ceramics, metallic blocks and a central bolt that pre-stresses the assembly. The impedance characterization of these transducers has been done by using numerical methods but, in many cases, one-dimensional simplified models are enough to a good physical interpretation of the problem. In this work was used the Mason's equivalent electric circuit for one-dimensional modeling of composed transducers. It was derived Thevenin's equivalent circuit to simplify the problem and evaluate the ef-

fects of the central bolt on the resonances. Some transducers were mounted and the transmission line circuit was used to measure the resonance frequencies. The results show a good fitting between the experimental and simulated by electric circuit. By using the presented model it was identify other longitudinal vibration modes further that considered in previous models and, so that, it was increased the possibility of investigate the behavior of the transducers using the frequency spectrum.

2:40

**4pEAa3. Integrated transducers for marine animal tags using thick film PZT.** Rasmus Lou-Møller (Ferroperm Piezoceramics A/S, Hejreskovvej 18A, 3490 Kvistgaard, Denmark, rm@ferroperm.net), Erling Ringgaard (Ferroperm Piezoceramics A/S, Hejreskovvej 18A, 3490 Kvistgaard, Denmark, er@ferroperm.net), Tomasz Zawada (Ferroperm Piezoceramics A/S, Hejreskovvej 18A, 3490 Kvistgaard, Denmark, tz@ferroperm-piezo.com), Sigmar Gudbjörnsson (Star-Oddi, Vatnagardar 14, 104 Reykjavik, Iceland, sigmar@star-oddi.com), Haraldr Hilmarsson (Star-Oddi, Vatnagardar 14, 104 Reykjavik, Iceland, sigmar@star-oddi.com)

The technology of printing PZT ceramic layers onto curved substrates using pad printing is demonstrated by the application of a transmitter/receiver system printed on alumina cylinders. The cylinder is used as housing for a Data Storage Tag (DST) for marine animals. DST's are used for monitoring behaviour and lifecycle of animals, providing valuable knowledge for marine biologists and researchers. Electrodes and PZT ceramic is printed directly onto the alumina housing using the pad printing technique, which eliminates the need for complex assembly procedures and other post processing steps. Since the PZT is situated on the external surface of the housing, direct contact between the acoustic elements and the marine animal is ensured and the elements do not take up valuable space inside the tag. The manufacturing of the active elements and the technology of pad printing PZT thick film is presented. The piezoelectric properties of the film and the acoustic properties have been tested under laboratory conditions. According to the obtained results one can conclude that the transmitter/receiver system was able to assure sufficient sensitivity in the required distance keeping the power consumption on an extremely low level. Second generation devices will be ready for commercialisation in the near future.

3:00

**4pEAa4. High-Output and High-Fidelity Microsized Driver System.** Keehoon Kim (Physical Optics Corporation, 20600 Gramercy Pl #103, Torrance, CA 90501, USA, kkim@poc.com), Tao Xu (Physical Optics Corporation, 20600 Gramercy Pl #103, Torrance, CA 90501, USA,

txu@poc.com), Reginald Daniels (Air Force Research Laboratory, 2610 Seventh St. Bldg 441, Wright Patterson AFB, OH 45433, USA, Reginald.Daniels@WPAFB.AF.MIL)

Active noise reduction (ANR) for aircrew hearing protection in harsh noise environments such as military and civilian flight operations requires a high-performance, microsized driver that provides not only high acoustic power, at least 130 dB SPL, but also high fidelity sound quality, especially in low-frequency ranges below 100 Hz. A new microsized smart material actuator (MSMA) is being developed by using high-density and direct-conversion piezoelectric exciters combined with a unique acoustic structure, unlike conventional moving coil/magnet-type drivers. The dimensions of the MSMA are 6 mm in diameter by 7 mm long, small enough for ear-canal application as an earplug driver, maximizing ANR effectiveness. The MSMA system consists of a microscale earplug driver, a protective package made of medical-grade stainless steel, and a compact actuation amplifier easily connected to or embedded in an external ANR controller. Performance tests show the system produces over 130 dB SPL in a 1 cc trapped cylindrical volume with very flat frequency responses even below 100 Hz. The total harmonic distortion of the MSMA is lower than 4% over all audible frequency ranges, without phase delay or discontinuities.

3:20

**4pEAa5. MEMS-based magnetic and electrostatic acoustic microspeakers.** Michael Pedersen (Novusonic Corporation, P.O. Box 183, Ashton, MD 20861, USA, info@novusonic.com)

Acoustic microspeakers remains a challenging application area for MEMS technology. This is mostly due to the inherently low transduction factors normally achievable in MEMS technology. While established electromagnetic transduction principles may be adapted for MEMS technology, there are also other transduction methods, such as electrostatic, with high performance potential. In this paper, the design, fabrication, and testing of two different types of MEMS microspeakers is presented. Firstly, an electromagnetic microspeaker is shown consisting of a MEMS diaphragm with integrated moving and an external rare earth permanent magnet and yoke structure. Secondly, a novel MEMS electrostatic microspeaker based on rolling contact is presented. In this device, high transduction forces are achieved by the strong electrical field over a solid insulator, and the forces are translated to a MEMS diaphragm by an integrated cantilever structure. Measurements on the electromagnetic microspeakers show a sensitivity of 83 dB SPL/mW at 1 kHz in a B&K 4153 acoustic coupler. The resonance frequency of the 6mm diameter diaphragm was 3.2 kHz. Numerical simulations suggest that simple changes to the moving coil geometry and material may increase the sensitivity to 108 dB SPL/mW.

### 3:40-5:20 Posters

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

### Contributed Papers

5:20

**4pEAa6. Sound pressure estimation at an electroacoustic transducer's voicing face by way of all-electrical sensing.** Florian Sandoz (Ecole Polytechnique Fédérale de Lausanne, EPFL STI LEMA, Station 11, CH 1015 Lausanne, Switzerland, florian.sandoz@epfl.ch), Hervé Lissek (Ecole Polytechnique Fédérale de Lausanne, EPFL STI LEMA, Station 11, CH 1015 Lausanne, Switzerland, herve.lissek@epfl.ch)

When it comes to designing active control disposals, sound pressure sensing is usually required. In the framework of active control of acoustic impedance, this work aims at conceiving a digital estimator that processes

voltage and current measurement to compute an estimation of the pressure in the vicinity of the diaphragm of an electro-dynamic speaker. After dealing with modeling concerns of the actuator, the bilinear transform is used to obtain equations of two IIR filters. An adaptive identification using the LMS algorithm and a SVD-based identification are tested to compute filters coefficient, both leading to poor experimental results. Accordingly, the actual transfer functions of the speaker are measured and optimal IIR filters are designed using the simulated annealing algorithm. It is then shown that good results can be achieved experimentally when the speaker is loaded electrically by a short circuit or an open circuit. Finally, performances of the process are discussed regarding other operating conditions.

**4pEaA7. Virtual Prototyping of Electrodynamical Loudspeakers by Utilizing a Finite Element Method.** Reinhard Lerch (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, reinhard.lerch@lse.eei.uni-erlangen.de), Manfred Kaltenbacher (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, manfred.kaltenbacher@lse.eei.uni-erlangen.de), Martin Meiler (Simetris GmbH, Am Weichselgarten 7, 91058 Erlangen, Germany, martin.meiler@simetris.de)

To speed up the development of electrodynamic loudspeakers, computer tools have to be applied. With appropriate computer simulations, the costly and lengthy fabrication of prototypes, as required within conventional experimental design, can be reduced tremendously. Present computer modeling tools are mainly based on equivalent circuit representations. The main drawback of these models, however, stems from the fact that the circuit element parameters have to be measured on a prototype, first. Therefore, the need for appropriate numerical simulation tools based on finite element method (FEM) arises, since as input parameters they suffice with geometrical and material data. However, present finite element tools suffer from their incompleteness in respect to full field couplings and nonlinear features. In this paper, a new finite element scheme is introduced and its utilization within the computer-aided design of electrodynamic loudspeakers is demonstrated. This scheme allows the precise and efficient calculation of the electromagnetic, mechanical and acoustic fields, including their couplings. Furthermore, nonlinear effects in the mechanical behavior of the spider as well as magnetic nonlinearities due to the nonhomogeneity of the magnetic field are taken into account.

**4pEaA8. Tactile Touch Plate with Variable Boundary Conditions.** Ros Kiri Ing (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, ros-kiri.ing@espci.fr), Didier Cassereau (Laboratoire Ondes et Acoustique, 10, rue Vauquelin, 75231 Paris, France, didier.cassereau@espci.fr), Mathias Fink (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, mathias.fink@espci.fr), Jean-Pierre Nikolovski (Laboratoire Ondes et Acoustique, ESPCI, Université Paris 7, CNRS, 10 rue Vauquelin, 75005 Paris, France, jean-pierre.nikolovski@cea.fr)

The touch screen device is becoming more and more widespread because it is a very user friendly human/machine interface. In acoustic domains, several approaches are used to realize such a device. Triangulation or Rayleigh waves absorption are such classical methods. However, these approaches are limited because they need a large number of sensors and are only applicable to plates of constant thickness and homogeneous materials. To remedy these limitations, a new approach is proposed using only two sensors. In this approach, one sensor is used to excite the plate, either continuously or impulsively. The second sensor is used to detect the acoustic waves generated in the plate. When an object comes into contact with the plate, some acoustic wave characteristics change. These changes affect different frequencies and depend on the position of the contact point. Comparing these changes with pre-recorded values, it is possible to achieve a tactile touch plate that only responds to specific touch locations. Several experiments with different types of plates were conducted and the results will be presented.

**4pEaA9. Personal sound system design for mobile phone, monitor, and television set: cylindrical shape approach.** Ji-Ho Chang (Center for Noise and Vibration Control, Korea Advanced Institute of Science and Technology, Dep. of Mechanical Engineering, Guseong-dong, Yuseong-gu, 305-701 Daejeon, Republic of Korea, chang.jiho@gmail.com), Jin-Young Park (Center for Noise and Vibration Control, Korea Advanced Institute of Science and Technology, Dep. of Mechanical Engineering, Guseong-dong,

Yuseong-gu, 305-701 Daejeon, Republic of Korea, jypark1979@kaist.ac.kr), Yang-Hann Kim (Center for Noise and Vibration Control, Korea Advanced Institute of Science and Technology, 4114, Department of Mechanical Engineering, Guseong-dong, Yuseong-gu, 305-701 Daejeon, Republic of Korea, yanghannkim@kaist.ac.kr)

Personal sound system that focuses sound on the user and reduces in the other zone has great interest in these days because it has significant needs to be applied in personal devices such as mobile phone, monitor, and television set. We have shown the feasibility of the personal sound system using a line array of loudspeaker units [C.-H. Lee et. al., J. Acoust. Soc. Am., 122, 3053 (2007)] based on acoustic contrast control [J.-W. Choi, Y.-H. Kim, J. Acoust. Soc. Am. 111, 1695(2002)], with the successful result of about 20dB difference between the front and the side region for 800-5kHz range. Continuing this research, we try to apply acoustic contrast control in cylindrical shape instead of two-dimensional planar shape that was used before in order to reduce the level of side lobes more. That is, acoustically bright zone and dark zone are determined as cylindrical shape surrounding the array of loudspeakers. Computer simulation and experimental result will be addressed and evaluated by comparing to the previous result. (This work was supported by the Korea Science and Engineering Foundation(KOSEF) through the National Research Lab. Program funded by the Ministry of Science and Technology(M10500000112-05J0000-11210).)

**4pEaA10. Efficient Directivity Pattern Control for Spherical Loudspeaker Arrays.** Franz Zotter (Institute of Electronic Music and Acoustics, Inffeldgasse 10 / 3, 8010 Graz, Austria, zotter@iem.at), Hannes Pomberger (Institute of Electronic Music and Acoustics, Inffeldgasse 10 / 3, 8010 Graz, Austria, pom@sbox.tugraz.at), Andrew Schmeder (Center for New Music and Audio Technologies, 1750 Arch Street, Berkeley, CA 94720, USA, andy@cnmat.berkeley.edu)

With an appropriate control system, directivity pattern synthesis can be accomplished with spherical loudspeaker arrays, e.g. in the shape of Platonic solids or spheres. The application of such devices for the reproduction of natural or artificial directivity patterns poses a relatively young field of research in computer music and acoustic measurements. Using directivity measurements with microphones, the directivity patterns of the individual speakers on the array can be determined. Usually, the directivity of the whole array may be regarded as a linear combination of these patterns. In order to gain control, the measurement data of the linear system need to be inverted. Given L loudspeakers and M microphones, this inversion yields the desired control system, an expensive LxM multiple-input-multiple-output (MIMO) filter. We introduce discrete spherical harmonics transform and decoder matrices to reduce the number of channels required for this control system, thus reducing the computational effort. However, this step often leads to a sparse MIMO-system, in which many off-diagonal transfer functions vanish. If applicable, the computation of the non-zero transfer functions only can be done at even much lower cost. A case study for an icosahedral loudspeaker array is given, showing the properties of the sparse MIMO-system.

**4pEaA11. Effects of enclosure design on the directivity synthesis by spherical loudspeaker arrays.** Alexander Mattioli Pasqual (Universidade Estadual de Campinas, Rua Mendeleiev, 200, Cidade Universitária "Zeferino Vaz", 13083-970 Campinas, Brazil, pasqual@fem.unicamp.br), José Roberto Arruda (Universidade Estadual de Campinas, Rua Mendeleiev, 200, Cidade Universitária "Zeferino Vaz", 13083-970 Campinas, Brazil, arruda@fem.unicamp.br), Philippe Herzog (Laboratoire de Mécanique et d'Acoustique - CNRS, 31 chemin Joseph Aiguier, 13402 Marseille, France, herzog@lma.cnrs-mrs.fr)

Spherical loudspeaker arrays have been used to generate non-uniform directivity patterns. It is known that the poor radiation efficiency of spherical sources and the loudspeaker electroacoustic behavior impose constraints on the directivity synthesis at low frequencies, which are aggravated as the source volume is made smaller. In this work, the effects of the enclosure design on the loudspeaker signal powers are analyzed. Two different approaches have been reported in literature, although quantitative comparisons



have not been provided. In the first approach, the drivers share the same enclosure volume and in the second, they have their own independent sealed cavities. Here, an analytical model that takes into account the interior and exterior acoustic coupling is used in order to evaluate the voltages that must feed the array drivers. It is shown that the signal powers can be reduced at low frequencies by letting the drivers share the same enclosure volume. However, this leads to controllability problems, since some natural frequencies of the enclosure are in the operation range of the spherical array. If controllability at natural frequencies is neglected, a simple lumped parameter model of the enclosure presents good agreement with the continuous model, indicating that heavy calculations may be unnecessary.

7:20

**4pEa12. Relative calibration and characterization of 1/4" condenser microphones under different environmental conditions.** Cécile Guianvarc'H (Institut National de Métrologie (LNE-INM/Cnam), 61 rue du Landy, 93210 La Plaine Saint Denis, France, cecile.guianvarch@cnam.fr), Paolo Alberto Giuliano Albo (Istituto Nazionale di Ricerca Metrologica (INRiM), Strada delle Cacce, 91, 10135 Torino, Italy, albo@inrim.it), Roberto Gavioso (Istituto Nazionale di Ricerca Metrologica (INRiM), Strada delle Cacce, 91, 10135 Torino, Italy, r.gavioso@inrim.it), Giuliana Benedetto (Istituto Nazionale di Ricerca Metrologica (INRiM), Strada delle Cacce, 91, 10135 Torino, Italy, gbenedet@inrim.it), 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)

Measurement condenser microphones are commonly used in air at ambient temperature and pressure. However, several applications require to use such microphones in environments which are significantly different. In particular, for the determination of the Boltzmann constant by an acoustic method, measurements take place in a cavity filled with pure argon or helium over a wide pressure range at the temperature of the triple point of water. For this application, it is important to determine the microphone frequency response and acoustic input impedance with a low uncertainty in these gas conditions. A few previous works have examined the influence of static pressure, temperature and gas composition on microphone sensitivity. In one case, these results were supported by a theoretical investigation using a lumped-element model. The aim of the present work is to compare theoretical results from different lumped-element models with experimental relative calibration results obtained using an electrostatic actuator technique. Measurements are performed on 1/4" condenser microphones maintained in argon and helium environments, at 273.16 K, in the pressure range between 50 kPa and 700 kPa. The results are used to test the existing theoretical models and to compare the microphone properties with the manufacturer's data.

THURSDAY AFTERNOON, 3 JULY 2008

ROOM 342B, 2:20 TO 3:40 P.M.

### Session 4pEAb

## Engineering Acoustics, Underwater Acoustics, Acoustical Oceanography, and ECUA: Acoustics in Marine Archeology

Kenneth Walsh, Cochair

*K & M Engineering LTD, 51 Bayberry Lane, Middletown, Rhode Island 02842, USA*

Tuncay Akal, Cochair

*SUASIS: Underwater Systems Technology Development, Tubitak-Marmara Research Center, Tech. D. Free Zone, Block A - L4, Gebze, Kocaeli, 41470, Turkey*

### Invited Papers

2:20

**4pEAb1. Green's theorem as the foundation of interferometry.** Adriana Citlali Ramírez (WesternGeco, 10001 Richmond Ave, Houston, TX 77042, USA, aperez50@slb.com), Arthur B. Weglein (University of Houston, Physics Department, 617 S&R Bldg1, Houston, TX 77204-5005, USA, aweglein@uh.edu)

A prerequisite for applying full wavefield theory to marine exploration is the completeness and proper sampling of recorded data, which can be satisfied with data extrapolation/interpolation techniques. Green's theorem can be applied to acoustic measurements of the Earth's subsurface to obtain exact equations that incorporate boundary conditions for the retrieval of the Earth's Green's function in positions where it was not measured. Recently, a number of papers on seismic interferometry have shown methods to reconstruct the Green's function between a pair of receivers by using data cross correlations. Current interferometry methods require dual measurements (pressure field and its normal derivative) which are not always available. The lack of dual measurements has encouraged the arrival of algorithms using high frequency and one-way wave approximations to the normal field derivative. The approximations are compromises to the exact theory and, hence, produce artifacts. We present a unifying framework for a broad class of interferometry techniques using Green's theorem. This framework and foundation allows errors and artifacts to be anticipated and fully explained as a consequence of approximations made within Green's theorem. We also develop a set of more effective interferometry methods, where fewer (or none) approximations are made and the result is improved.

**4pEAb2. Surveillance and Protection of Underwater Archaeological Sites: Sea-Guard.** Tuncay Akal (SUASIS: Underwater Systems Technology Development, Tubitak-Marmara Research Center, Tech. D. Free Zone, Block A - L4, Gebze, 41470 Kocaeli, Turkey, tuakal@yahoo.com), Piero Guerrini (NURC Nato Undersea Research Centre, Viale San Bartolomeo 400, 19026 La Spezia, Italy, guerrini@nurc.nato.int), William A. Kuperman (MPL, Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093-0238, USA, wkuperman@ucsd.edu), Kerem Koprulu (SUASIS: Underwater Systems Technology Development, Tubitak-Marmara Research Center, Tech. D. Free Zone, Block A - L4, Gebze, 41470 Kocaeli, Turkey, keremkpr@yahoo.com), 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), 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)

Recent advances in underwater detection and survey techniques have increased the possibility of finding the wrecks of vessels and their cargoes. Since prehistoric times, many populations have used vessels for transportation and trade and built harbors and it is now estimated that there are about a million antique wrecks that lie underwater, still to be discovered. Many of these wrecks and ancient harbours lie within shallow depths and can be discovered and accessed easily by humans using available hardware such as side-scan sonars, multi-beam echo sounders, ROVs, GPS navigation, diving gear. Even though the majority of known archaeological sites on land are well protected by guards and high tech equipment, there is no technology for surveillance and protection of underwater archeology sites. This paper presents a new system specifically adapted for the surveillance and protection of underwater archaeological sites (SEA-GUARD). Options available to the acoustic sensor suite of this system are pressure hydrophones and/or vector sensors. The system is based on an underwater acoustic monitoring of the noise field near an archaeological site sending preprocessed signals as an alarm to the authorities on shore via GSM/GPRS or Satellite network.

### *Contributed Paper*

3:00

**4pEAb3. Acoustic methods in extremely shallow water for reconstruction of ancient environments.**

Fantina Madricardo (CNR-Istituto di Scienze Marine, Riva Sette Martiri - Castello 1364/a, 30122 Venice, Italy, fantina.madricardo@ismar.cnr.it), Sandra Donnici (CNR-Istituto di Scienze Marine, Riva Sette Martiri - Castello 1364/a, 30122 Venice, Italy, sandra.donnici@ismar.cnr.it), Alberto Lezziero (PHAROS sas, Via della Libertà 12, 30175 Venice, Italy, alezziero@gmail.com), Federica De Carli (CNR-Istituto di Acustica 'O.M .Corbino', Via del Fosso del Cavaliere, 100, 00133 Rome, Italy, federica.decarli@gmail.com)

A large area of the Venice Lagoon was surveyed through a modified traditional echosounder. For the first time, acoustic methods together with geological analysis were used systematically to investigate natural and anthropogenic morphologies buried in the lagoonal sediments. The shallowness (with depths often < 1m) of the Venice Lagoon represents a challenge for underwater acoustic methods. The results we present in this paper show that such methods are very useful for detailed geomorphological and archaeological reconstruction and can be extended to other similar environments. In this context, a general methodology of multidisciplinary data collection was developed. As a synthesis of our acoustic and geoarchaeological investigation, maps of the ancient lagoonal environment were produced.

### *Invited Paper*

3:20

**4pEAb4. Precision positioning for deep-water archaeology.** Brian Bingham (Franklin W. Olin College of Engineering, 1000 Olin Way, MH 368, Needham, MA 02492, USA, bbing@olin.edu)

Underwater archaeology makes use a variety of tools for deep-water exploration, but the defining characteristic of archaeology, what distinguishes the endeavor as scientific in purpose, is the quantifiable techniques employed for precision navigation. Precise positioning enables the collocation of archaeological results into a common reference frame, creating a persistent data products with the spatial context to enable interpretation. With this common context, data products such as site-maps, micro-bathymetric maps and photomosaics afford the capability to perform measurements with known accuracy and resolution, a key differentiator of the science of archaeology. Acoustic ranging is the meter stick of marine archeology. Archaeology has leveraged standard navigation techniques: long baseline (LBL) positioning, ultra-short baseline (USBL) tracking, Doppler velocity log (DVL) dead-reckoning, etc. However, the extreme precision required for archaeology has driven the development of new precise ranging instruments. This paper presents the capabilities and limitations of today's precision ranging instruments which use discrete sequence spread spectrum (DSSS) signal processing to provide the requisite performance. In addition to a technical discussion, the paper presents the results of recent field work in the Aegean, the Mediterranean and the East Coast of the United States.

## Session 4pEAc

## Engineering Acoustics: Ultrasonic Acoustic MEMS II (Poster Session)

F. Levent Degertekin, Cochair  
*Georgia Institute of Technology*

Dominique Certon, Cochair  
*Lab. LUSSI Inserm U930 CNRS FRE 2448 Univ. François Rabelais*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pEAc1. A surface acoustic wave impedance loaded sensor for wireless humidity measurement.** Rung-De Wang (Institute of Applied Mechanics, National Taiwan University, No. 1, Sec. 4, Roosevelt Road, 106 Taipei, Taiwan, r94543020@ntu.edu.tw), Yung-Yu Chen (Department of Mechanical Engineering, Tatung University, No. 40, Sec. 3, Chungshan N. Rd., 104 Taipei, Taiwan, yychen@ttu.edu.tw), Tsung-Tsong Wu (Institute of Applied Mechanics, National Taiwan University, No. 1, Sec. 4, Roosevelt Road, 106 Taipei, Taiwan, wutt@ndt.iam.ntu.edu.tw)

The control of humidity is required in various areas such as the smart living space, industry, agriculture, and medicine. To reduce cost and increase lifetime, a wireless humidity sensor without an additional power supply should be developed to fulfill their needs. Recently, there are more and more efforts on surface acoustic wave (SAW) based radio frequency identification (RFID) system. The RFID system primarily consists of a reader and transponders, composed of SAW tags and antennas. Due to the capabilities of passive operation and wireless connection, SAW tags are also suitable for remote sensing. Therefore, in this study, a wireless humidity sensor is accomplished by integrating a 433MHz 128°YX-LiNbO<sub>3</sub> SAW tag with a resistive humidity sensor. The SAW tag is designed as an impedance loaded sensor. Moreover, because nanostructure sensing materials possess high surface-to-volume ratio, large penetration depth, and fast charge diffusion rate, CSA-doped polyaniline nanofibers are synthesized by the interfacial polymerization method and further deposited on the resistive humidity sensor as sensitive film to enhance the sensitivity. Finally, the humidity sensor was constructed and measured. Results indicate that our proposed sensor exhibits not only good linearity but also high sensitivity. Furthermore, it is indeed capable of passive remote sensing.

**4pEAc2. Periodically poled transduction structures built on thinned single crystal Lithium Niobate layers bonded onto Silicon.** Emilie Courjon (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, emilie.courjon@femto-st.fr), Gwenn Ulliac (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, gwenn.ulliac@femto-st.fr), Jérôme Hauden (Photline Technologies, 16 rue Jouxoux, 25000 Besançon, France, jerome.hauden@photline.com), Sylvain Ballandras (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, sylvain.ballandras@femto-st.fr)

The demand for highly coupled high quality acoustic wave devices has generated a strong innovative activity, yielding the investigation of new excitation principles and wave guide structures. In a recent work, we have investigated the interest of periodically poled transducers (PPTs) on single crystal LiNbO<sub>3</sub> Z-cut plates. The PPT simply consists of a periodically poled ferroelectrics layer metallized on its two faces for acoustoelectric excitation.

In this work, we have fabricated PPTs on single crystal LiNbO<sub>3</sub> Z-cut thinned layers reported on Silicon to develop a new kind of wave guide. Their fabrication is based on a home-made wafer bonding technique based on a metal-metal adhesion, the lithium niobate being lapped and polished in order to obtain a layer exhibiting a few tenth micrometer thick. The corresponding devices have successfully fabricated and tested in the frequency range 100-300 MHz. Comparison between experimental measurements and theoretical analysis (using a combination of finite and boundary element analysis) show that the modes are well controlled and that different kind of wave polarization may be excited.

**4pEAc3. Acoustical properties characterization of a composite made of SU-8 and nanoparticles for BioMEMS application.** Julien Carlier (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, julien.carlier@univ-valenciennes.fr), Pierre Campistron (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, pierre.campistron@univ-valenciennes.fr), Dorothée Callens (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, dorothee.callens@univ-valenciennes.fr), Caroline Soyer (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, caroline.soyer@univ-valenciennes.fr), Bertrand Nongaillard (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, Bertrand.Nongaillard@univ-valenciennes.fr), Shengxiang Wang (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, shx\_wang@126.com), Xingzhong Zhao (Department of Physics, Wuhan University, 430072 Wuhan, China, xzzhao@whu.edu.cn)

The photoresist SU-8 (often used in microtechnology) has been acoustically characterized at the frequency of 1 GHz thanks to ZnO transducers. This material will be used to achieve acoustical matching between silicon and water at this frequency and a gain of about 5 dB has been obtained using the combination of this material with a silicon oxide layer. The acoustical characterization of a composite material made of this photoresist SU-8 doped with nanoparticles (TiO<sub>2</sub> for example) is also presented. The aim is to achieve matching layers with specific mechanical impedance around 5 MRay and depending on the nanoparticles concentration. The targeted application concerns the integration of high frequency ultrasonic transducer in Lab-on-Chip for biological cell characterization, especially adhesion properties or biological cells mechanical behaviour. The mechanical impedance and the attenuation of this composite material are characterized around 1 GHz thanks to a standing wave ratio method measuring S11 parameter and extracting the targeted parameters thanks to signal processing.

**4pEAc4. High Overtone Bulk Acoustic Resonators Based on Thinned Single-crystal Piezoelectric Layers: Filters and Frequency Sources Applications.** Dorian Gachon (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, dorian.gachon@femto-st.fr), Jeremy Masson (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, jeremy.masson@femto-st.fr), Emilie Courjon (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, emilie.courjon@femto-st.fr), Sylvain Ballandras (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, sylvain.ballandras@femto-st.fr)

The thin film bulk acoustic wave resonators exploiting the thickness-extensional vibration mode of piezoelectric thin films is a key technology as alternative solutions to standard SAW resonators. Lakin have emphasized the capability of High Overtone Bulk Acoustic Resonators to present high quality factors at frequencies in the GHz range. HBAR spring from the conjugation of the strong coupling coefficient of deposited piezoelectric thin films and of the high intrinsic quality of used substrates. The piezoelectric film and the two electrodes on its both sides are used as transducer whereas the acoustic energy is mainly trapped in the substrate. The resonant frequency corresponds to a half wavelength in the entire thickness of the device and, in opposition to FBAR, we can utilize both odd and even harmonics. The fundamental, generally in the vicinity of 10 MHz, has no specific interest but Q.f products around  $1.1 \times 10^{14}$  have already been obtained for high overtones using aluminum nitride thin films deposited onto sapphire. In view of improving the Q factor of thin films, it is desirable to use a single-crystal piezoelectric material such as lithium niobate. We show and compare the fabrication in both approaches. Different measurement results are exposed for both approaches for the fabrication of oscillator and filters are shown and discussed.

**4pEAc5. Micromachined probe performance assessment.** Mathieu Legros (Vermon SA, 180 rue Général Renault, 37000 Tours, France, m.legros@vermon.com), Cyril Meynier (Vermon SA, 180 rue Général Renault, 37000 Tours, France, c.meynier@vermon.com), Guillaume Ferin (Vermon SA, 180 rue Général Renault, 37000 Tours, France, g.ferin@vermon.com), Rémi Dufait (Vermon SA, 180 rue Général Renault, 37000 Tours, France, r.dufait@vermon.com)

In recent years, Capacitive Micromachined Ultrasound Transducer (CMUT) technology was widely investigated and functional prototypes have been released by several R&D teams. CMUT technology offers outstanding characteristics in acoustic, interconnect packaging capabilities or in integration features that are exciting criteria for new medical applications. We propose a full acoustic characterization report of a CMUT probe. A linear array was fully packaged with electronic preamplifier boards integrated. A complete acoustic characterization of the probe is then performed and presented. In a second phase, a linear probe with piezocomposite technology is realized. The conception is done in regard to the geometric characteristics and to the acoustic response of the micromachined probe. Then an electro-acoustical benchmark CMUT / piezocomposite is realized in the closest conditions. Using a commercial ultrasound imaging platform, an image assessment is performed. The images are first analysed in a quantitative way with a tissue mimicking phantom, using a computerized tool who considered imaging parameters such as contrast, resolution or signal to noise ratio. In a second way, a clinical perspective is discussed with in vivo images.

**4pEAc6. CMUT membrane model based on theory of stratified plates.** Nicolas Senegond (Lab. LUSSI Inserm U930 CNRS FRE 2448 Univ. François Rabelais, 10, Boulevard Tonnellé, 37032 Tours, France, nicolas.senegond@univ-tours.fr), Franck Teston (Lab. LUSSI Inserm U930 CNRS FRE 2448 Univ. François Rabelais, 10, Boulevard Tonnellé, 37032 Tours, France, franck.teston@univ-tours.fr), Cyril Meynier (Vermon SA, 180 rue Général Renault, 37000 Tours, France, c.meynier@vermon.com), Dominique Certon (Lab. LUSSI Inserm U930 CNRS FRE 2448 Univ. François Rabelais, 10, Boulevard Tonnellé, 37032 Tours, France, dominique.certon@univ-tours.fr)

To optimize and characterize capacitive micromachined transducers, static and dynamic mechanical behavior of the membrane have to be considered and modeled. Most of cMUT models are based on the classical plate

equation. The flexural rigidity is homogenized for multi-layer structure taking into account properties of each material. Homogenization relations are available provided the flexural plate equation can be separated from the in-plane deformation equation, so-called the membrane-like behavior equation. Practically, for membrane partially covered with electrode, at the metallized/non-metallized discontinuity, these equations cannot be solved separately and homogenization procedure is no more available. Moreover, initial membrane deflection introduces supplementary coupling between in-plane and flexural displacements requiring modification of homogenization relations. A complete analytical formulation of the plate equation is developed in this paper. A finite difference meshing has been used to numerically solve the new set of plate equations. First, a cMUT with simple geometry is modeled for validation and comparison with Finite Element simulation (COMSOL Multiphysics software). Then, the validity domain of the "simple" flexural plate equation is discussed. Finally, an example of membrane optimization is given, where a multi-layer structure is proposed to reduce influence of static pre-stresses in the membrane on collapse voltage and resonance frequency.

**4pEAc7. Pure-shear mode BAW resonators consisting of (11-20) textured ZnO films.** Takahiko Yanagitani (Faculty of Engineering Tohoku University, 6-6-05 Aramaki Aoba Aoba-ku, 610-0321 Kyotanabe, Japan, yanagi@ecei.tohoku.ac.jp), Kiuchi Masato (National Institute of Advanced Industrial Science and Technology, 1-8-31 Midorigaoka, Osaka, 563-8577 Ikeda, Japan, yanagitani@usl.doshisha.ac.jp), Mami Matsukawa (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp), Yoshiaki Watanabe (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, kwatanab@mail.doshisha.ac.jp)

Thickness pure-shear mode film bulk acoustic wave resonators (FBARs) made of (11-20) textured ZnO films have been fabricated. We have also fabricated FBAR structure consisting of two layers of the (11-20) textured ZnO film with opposite polarization directions. This FBAR structure operated in second overtone pure-shear mode, and allowed shear-mode FBARs at higher frequency. The effective electromechanical coupling coefficients  $k_{eff}^2$  of pure-shear mode FBAR and second overtone pure-shear mode FBAR in this study were found to be 3.3% and 0.8%, respectively. The temperature coefficient of frequency (TCF) of thickness extensional mode FBAR, pure-shear mode FBAR, and second overtone pure-shear mode FBAR were measured in the temperature range of 10-60 °C. TCF values of -63.1 ppm/°C, -34.7 ppm/°C, and -35.6 ppm/°C were found for the thickness extensional mode FBAR, the pure-shear mode FBAR, and the second overtone pure-shear mode FBAR, respectively. These results demonstrated that pure-shear mode ZnO FBARs have more stable temperature characteristics than the conventional thickness extensional mode ZnO FBARs.

**4pEAc8. Influence of the shape of the membrane of computed cMUT using FEA/BEM analysis.** Stanislas Clatot (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, stanislas.clatot@femto-st.fr), Emeline Sadoulet (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, emeline.sadoulet-reboul@univ-fcomte.fr), Olivia Arbey (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, olivia.arbey@femto-st.fr), William Daniau (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, william.daniau@femto-st.fr), Julien Garcia (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, julien.garcia@femto-st.fr), Joseph Lardies (FEMTO-ST Applied Mechanics, 24 chemin de l'épithaphe, 25000 Besançon, France, joseph.lardies@univ-fcomte.fr), Sylvain Ballandras (CNRS FEMTO-ST, 32 Avenue de l'Observatoire, 25044 Besançon, France, sylvain.ballandras@femto-st.fr), Marc Berthillier (University of Franche-Comté, Institute FEMTO - LMARC, 24 rue de l'Épithaphe, 25000 Besançon, France, marc.berthillier@univ-fcomte.fr)

The possibility to excite and detect acoustic waves in fluids using capacitive micro-machined ultrasonic transducers (cMUT) built on silicon using clean room techniques offers attractive opportunities for manufacturing high quality low cost imaging probes. CMUTs developed for acoustic imaging exploit the first flexural mode of thin and stiff membranes, leading to



bandwidth larger than 100%. These transducers can be accurately designed using mixed finite element analysis/boundary element methods (FEA/BEM). Periodic FEA particularly allows for the simulation of devices exhibiting complicated shape interfaces and involving materials of different nature. BEM also are particularly well-suited to provide an accurate description of any stacked medium assuming flat interfaces for the radiation area and the layer interface for 2 and 3D structures as well. In this work, we have

analysed the influence of the shape of the MUT membrane on the spectral response of the transducer. 3D Computations have been conducted considering radiation conditions in the substrate on the backside (silicon) and in viscous water on the front side. We have particularly focused the computation on solutions allowing for reducing parasitic modes in the actual operation of the transducer. The efficiency of the different configuration are compared in terms of emitted pressure contributions.

THURSDAY AFTERNOON, 3 JULY 2008

P3-C, LEVEL 3, 3:40 TO 5:20 P.M.

### Session 4pEAd

## Engineering Acoustics and Signal Processing in Acoustics: Transducers and Signal Processing for the Oil and Gas Industry II (Poster Session)

Fernando Garcia-Osuna, Cochair  
*Schlumberger*

Benoit Froelich, Cochair  
*Etudes et Productions Schlumberger*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pEAd1. Multimode evaluation of cement behind steel pipe.** Benoit Froelich (Etudes et Productions Schlumberger, 1, rue Becquerel, BP 202, 92142 Clamart, France, [bfroelich@clamart.oilfield.slb.com](mailto:bfroelich@clamart.oilfield.slb.com))

Oil wells are usually cased with a steel pipe, and cement is injected in the annulus between pipe and rock in order to provide hydraulic isolation of the reservoir. The traditional ultrasonic technique to evaluate cement behind steel pipe has been to excite the pipe thickness mode, and to extract the acoustic impedance of the material from the pulse-echo response. However, this technique suffers from some limitations, such as a limited depth of investigation or a poor discrimination between mud and light cement. To alleviate such limitations, the thickness mode can be combined with the pipe flexural mode. Although dispersive, the flexural mode has the unique property of a quasi constant group velocity in a specific frequency domain which is high enough to provide azimuthal resolution. Such property provides the means for an accurate measurement of the time of arrival and the amplitude of the different echoes generated by a particular geometry. In particular, the echo traveling within the annulus and reflected by the rock can be easily detected from within the pipe, and provide novel information on the annulus fill.

**4pEAd2. Data fusion technique applied to steam wastage estimation and fault detection in an industrial process heating application.** Sivaram Nishal Ramadas (Institute of Sound and Vibration Research, University of Southampton, University Road, Highfield, S017 1BJ Southampton, UK, [nishal@ieee.org](mailto:nishal@ieee.org))

Data fusion, the process of combining information obtained together from many heterogeneous sensors to form a single composite picture of the environment, is used widely in many applications. Modern steam heating systems consist of mechanical devices known as 'traps', which are robust and reliable but inevitably over time can wear and fail, with the possibility of leaking steam. To diagnose such faulty steam traps and the level of leakage under operating conditions in a closed system is difficult. This paper presents the preliminary work carried out to integrate together data recorded from commercial sensors (such as piezoelectric acoustic emission devices, pressure transmitters, and thermocouples) to estimate steam wastage and fault detection in a steam system. Experimental data were acquired from a purpose built steam wastage test rig (built similar to the method outlined in the British Standard for determination of steam loss from traps), capable of simulating varying condensate loads by injecting preheated water into a steam test line. The captured composite data is then used to develop a signal processing algorithm to diagnose effective trap operation and quantify the rate of steam loss in the system and the results are discussed.

## Session 4pEAe

## Engineering Acoustics: Transducers II (Poster Session)

Robert M. Koch, Chair  
NOWC

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pEAe1. Condenser microphone as parametric electroacoustic system and its time-domain modelling via equivalent electrical circuit in SPICE software.** Michal Vlk (Czech Technical University, Faculty of Electrical Engineering, Department of Radioengineering, Technická 2, 166 27 Praha 6 - Dejvice, Czech Republic, vlkm3@fel.cvut.cz)

Condenser microphone usually works with constant polarisation voltage. Where special attentions are prescribed for spectrum of self noise there may be useful to use high frequency voltage (pump) applied to transducer instead of DC charge. When frequency of pump is orders of higher than frequency of acoustic signals, large parametric gain occurs in the transducer. Method for describing such a system via equivalent electric network with approach of electronic filter design blocks was developed and special analogies were used for convergency improvement. Method was applied to time-domain simulation of pressure transducer with current-mode diode discriminator in SPICE computer analysis software. Simulations of transient behavior after switching system on was demonstrated.

**4pEAe2. Modelling of radiofrequency MEMS bulk acoustic wave resonators with Legendre polynomial method.** Antoine A. Raherison (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, raherson\_antoine@yahoo.fr), Faniry Emilson F. Ratolofanahary (LAPAU, Université de Fianarantsoa, 301 Fianarantsoa, Madagascar, faniry@univ-fianar.mg), Jean-Etienne J. Lefebvre (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, jean-etienne.lefebvre@univ-valenciennes.fr), Lahoucine L. Elmaïmouni (Faculté polydisciplinaire d'Ouarzazate, Université Ibn Zohr, BP 638, Ouarzazate, Morocco, la\_elmaïmouni@yahoo.fr), Tadeusz T. Gryba (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, tadeusz.gryba@univ-valenciennes.fr)

Radio-Frequency electro-acoustic resonators and filters are key components for wireless communication devices. The next generation of mobile systems requires for these components a strict size reduction, higher operating frequency and better power resistance. So, MEMS BAW (Bulk Acoustic Wave) devices have emerged as an efficient alternative to established Surface Acoustic Wave filtering solutions. Knowledge of the electromechanical coupling coefficient and the Q-factor are of primary importance for the MEMS BAW device design. In our model, the Legendre polynomial method which describes the structure and incorporates automatically the boundary conditions in constitutive and propagation equations is used to calculate these parameters. It is the first time this method is applied to study standing rather than propagative waves. The advantage of this approach are, in a unique formulation, to take into account the presence of sources, existence of electrodes' losses and dissimilarity of the constituent materials of the resonator. To validate this approach, it is applied to a Al/ZnO/Al resonator. Through harmonic and modal analysis, the influence of electrodes

properties on MEMS BAW resonator performances are illustrated; the results are compared with those published earlier. This method is an efficient tool for designing MEMS BAW filter.

**4pEAe3. Design of dispersive layered SAW filters with CMOS low noise amplifier.** Tadeusz T. Gryba (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, tadeusz.gryba@univ-valenciennes.fr), Julien Carlier (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, julien.carlier@univ-valenciennes.fr), Etienne Ntagwirumugara (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, entagwir@yahoo.fr), Victor Y. Zhang (IEMN-CNRS, Av. Poincaré, Cite Scientifique, B.P. 60069, 59652 Villeneuve d'Ascq Cedex, France, victor.zhang@iemn.univ-lille1.fr), Jean-Etienne J. Lefebvre (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, jean-etienne.lefebvre@univ-valenciennes.fr)

Up to now, RF front-end surface acoustic wave (SAW) filters for mobile communication are mainly fabricated on LiNbO<sub>3</sub> and LiTaO<sub>3</sub> substrates. A monolithic integration of these filters on Si substrates is highly desirable, but Si is non piezoelectric. One alternative is the deposition of a piezoelectric film on the semiconductor substrate. CMOS technology is very attractive for integrating the radio frequency modules in a single chip. This paper presents the analysis and realisation of a SAW passband filter on silicon substrate based on a piezoelectric ZnO thin film working near 1 GHz, integrated with a CMOS low-noise amplifier. We propose a modified coupling of modes (COM) approach for a layered ZnO/Si surface acoustic wave filter. The COM parameters in this formulation are the Rayleigh wave velocity, the electromechanical coupling coefficient, the complex reflection coefficient, the transduction coefficient and the inter-digital capacitance C. This is a dispersive SAW layered filter some parameters of which become frequency dependent due to the phase velocity dispersion. We present the theoretical and experimental results of the filter integrated with a CMOS low noise amplifier.

**4pEAe4. A stepped plate transducer as ultrasonic range sensor.** Yub Je (Postech, San 31, Hyoja-Dong, Namgu, 790-784 Pohang, Republic of Korea, effortjy@postech.ac.kr), Jong-Kyu Park (Postech, San 31, Hyoja-Dong, Namgu, 790-784 Pohang, Republic of Korea, chong@postech.ac.kr), Dong Hoon Yi (Postech, San 31, Hyoja-Dong, Namgu, 790-784 Pohang, Republic of Korea, effortjy@hotmail.com), Haksue Lee (Postech, San 31, Hyoja-Dong, Namgu, 790-784 Pohang, Republic of Korea, jolary@postech.ac.kr), Wonkyu Moon (Postech, San 31, Hyoja-Dong, Namgu, 790-784 Pohang, Republic of Korea, wkmooon@postech.ac.kr)

A new type of highly directional ultrasonic transducer is designed and tested as an ultrasonic range sensor by using the parametric acoustic arrays. To get intensive primary waves, a modified stepped plate transducer is proposed. Gallego-juarez et al.(1978) first proposed stepped plate transducer for high power radiation at one frequency. The steps with the height of half-

wavelength of sound in air compensate discrete phase difference on the vibrating plate. However, two collimated beams are required for the parametric acoustic array. The position and height of the steps are modified to compensate the flexural vibration for two frequencies in this transducer. The transducer diameter is 50mm. The optimal primary frequencies are designed as 80kHz and 120kHz to generate difference frequency of 40kHz efficiently. The proposed transducer has the HPBW of 5° that is much higher directivity than general ultrasonic range sensor (Generally 20° on same size). The maximum SPL is 130dB at primary frequencies and 95dB at difference frequency on 75Vpk input. These experimental results show that the proposed transducer can successfully improve the spatial resolution of ultrasonic sensor. [Research is partly supported by MIC/IITA Intelligent Robot Sensor and partly supported by DAPA and ADD UD070054AD]

**4pEAe5. A study on the adaptive design of loudspeaker driver parameters for a given enclosure.** Seokjin Lee (Applied Acoustics Lab., INMC, 132-302, Seoul National University, San 56-1, Sillim-dong, Kwanak-gu, 151-742 Seoul, Republic of Korea, sjlee@acoustics.snu.ac.kr), Junguk Noh (Applied Acoustics Lab., INMC, 132-302, Seoul National University, San 56-1, Sillim-dong, Kwanak-gu, 151-742 Seoul, Republic of Korea, junoh37@acoustics.snu.ac.kr), Koeng-Mo Sung (Applied Acoustics Lab., INMC, 132-302, Seoul National University, San 56-1, Sillim-dong, Kwanak-gu, 151-742 Seoul, Republic of Korea, kmsung@snu.ac.kr)

In the conventional methods of electro-dynamic loudspeaker design, driver units are selected first and then the enclosures are designed for the best compensation of frequency responses of the given drivers. However, in the industry of nowadays, these conventional methods are not appropriate; the enclosure design is determined first, and after that, a proper driver unit is selected by trial and error. In this paper, an adaptive design method of loudspeaker driver parameters for a given enclosure is studied. We estimated loudspeaker parameters such as diaphragm size, acoustic compliance, acoustic resistance, etc., to obtain an optimal alignment or desired response curve for a given enclosure. In the proposed method, the target frequency response curve in low-frequency range is determined first. Second, the optimal size of driver is calculated. The other driver parameters, such as driver suspension, Q of driver, etc., are designed lastly. Easier and faster design can be achieved by the proposed design method of electro-dynamic loudspeakers.

**4pEAe6. Consideration on design of the sensitivity in piezoelectric vibratory tactile sensor.** Subaru Kudo (School of Science and Engineering, Ishinomaki Senshu University, 1 Shinmito, Minamisakai, 986-8580 Ishinomaki-city, Miyagi, Japan, kudou@isenshu-u.ac.jp)

The piezoelectric vibratory tactile sensors have been used for measuring the softness and hardness of an object. They make use of changes in the resonance frequency when their vibrating sections are brought into contact with an object. In this study, the sensitivity of the frequency change on the tactile sensors is experimentally considered. The longitudinal-bar-type and the fixed-free-bar type resonators are used as the tactile sensors. Then, the experimental characteristics of the manufactured tactile sensors are shown by measuring the softness and hardness of the test pieces. The differences of the characteristics of the sensitivity are discussed from the viewpoint of the resonance frequency, vibration mode and the dimensions of the resonator. It is clarified that the sensitivity on the tactile sensor is inversely proportional to the mass of the resonator. Moreover, the effects of the resonance frequency change on the tactile sensor by the load force and the additional mass, in case of contacting with an object, are analyzed using the finite element method. The obtained results will be useful for designing the piezoelectric vibratory tactile sensor.

**4pEAe7. Improvement of low frequency signal radiation performance for piezoelectric loudspeakers.** Juro Ohga (2-24-3, Tamanawa, 247-0071 Kamakura, Japan, johga@nifty.com)

Piezoelectric loudspeaker is characterized by their very simple constructions, but their radiation performance at low frequency region is poor because their diaphragms are too stiff for large amplitude vibration. This paper proposes two innovative methods for improvement of low frequency radiation performance of piezoelectric direct-radiator loudspeakers. One is a flat-shape loudspeaker unit with a tuck-shaped flexible diaphragm by a sheet of piezoelectric film. The other is a paper cone loudspeaker unit with a large radiator driven by revolution of piezoelectric ultrasonic motors. A loudspeaker system with satisfactorily wide frequency range for music is constructed by combination of these two sorts of loudspeaker units.

**4pEAe8. Geometrical optimization of vented boxes.** Guillaume Pellerin (CNAM Paris, 5, rue du Vert Bois, 75003 Paris, France, guillaume.pellerin@cnam.fr)

Today, low frequency reproduction with loudspeakers in vented box is limited by two factors: the volume of the box, and non-linear airflow through the vent. We propose a novel approach that takes into account aerodynamical parameters, leading to original profiles and an improved functioning of the vented box. For example, under certain alignment conditions, there exists a second cut-off frequency below the first one, localized on the lower impedance hump. Using this lower cut-off frequency and an adapted vent profile makes it possible to radiate frequencies under 40 Hz with box volumes smaller than 1 liter and small drivers. A prototype will be demonstrated.

## Session 4pEaf

## Engineering Acoustics: Sound Fields II (Poster Session)

Raymond Kirby, Chair  
Brunel University

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pEaf1. Application of Advanced Features in Computational Acoustics.**

Reinhard Lerch (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, reinhard.lerch@lse.eei.uni-erlangen.de), Manfred Kaltenbacher (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, manfred.kaltenbacher@lse.eei.uni-erlangen.de), Martin Meiler (Simetris GmbH, Am Weichselgarten 7, 91058 Erlangen, Germany, martin.meiler@simetris.de)

Modern numerical simulation tools allow the analysis of the generation and propagation of sound. However, a variety of computational features enhancing their applicability are missing in these codes. Therefore, it is sometimes cumbersome to come to practically useful results when applying such codes to real life problems. Here, some of these lacking features will be addressed and, furthermore, their implementation in a new finite element code is reported. These features are: Frequency dependent damping propagation medium, nonmatching grids for computing sound in neighbouring domains with quite different propagation velocities and perfectly matched layers (for handling of open domain problems without reflections at boundaries of the computational domain). A finite element environment including pre- and post-processing has been established to support the development of electroacoustic devices. The following real life examples will be reported: electrodynamic loudspeakers, noise emission from power transformers, ultrasound devices for medical imaging such as conventional piezoceramic arrays as well as CMUTs (capacitive micromachined ultrasound transducers). It will be also shown that the code is able to handle nonlinear effects in transducing mechanisms and acoustic wave propagation, as occurring in HIFU (high intensive focused ultrasound) applications.

**4pEaf2. Theoretical Study of Near 3D Sound Field Reproduction Based on Wave Field Synthesis.**

Toshiyuki Kimura (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, t-kimura@nict.go.jp), Yoko Yamakata (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, yamakata@nict.go.jp), Michiaki Katsumoto (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, katsumoto@nict.go.jp)

It is very important to develop near 3D sound field reproduction techniques in order to realize the ultra-realistic communication such as 3D TV and 3D tele-conference. In this report, the principle of the near 3D sound field reproduction technique using wave field synthesis is defined from Kirchhoff-Helmholtz integral equation and two methods (dipole control method and directional point control method) are proposed. The performance of two proposed methods is studied by computer simulation and it is shown that the dipole control method has good performance and that the

directional point control method has good performance if the directivity of loudspeakers is unidirectional or shotgun.

**4pEaf3. Performance Evaluation of 3D Sound Field Reproduction System with a Few Loudspeakers and Wave Field Synthesis.**

Munenori Naoe (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, nm.s512.ex@nict.go.jp), Toshiyuki Kimura (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, t-kimura@nict.go.jp), Yoko Yamakata (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, yamakata@nict.go.jp), Michiaki Katsumoto (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, katsumoto@nict.go.jp)

A conventional 3D sound field reproduction system using wave field synthesis places a lot of loudspeakers around the listener. However, since such a system is very expensive and loudspeakers come into the listener's field of vision, it is very difficult to construct an audio-visual system with it. We developed and evaluated a 3D sound field reproduction system using eight loudspeakers placed at the vertex of cube and wave field synthesis. We compared the sound localization of a loudspeaker array with that of seventeen loudspeakers placed around the listener and found that their localization capabilities were equivalent except the normal direction of cube's planes.

**4pEaf4. Synthesis of wave front in the sound field recording/reproduction system based on the boundary surface control principle.**

Seigo Enomoto (Advanced Telecommunication Research Institute International, 2-2-2 Hikaridai, 'Keihanna Science City', 6190288 Kyoto, Japan, seigo.enomoto@atr.jp), Yusuke Ikeda (Advanced Telecommunication Research Institute International, 2-2-2 Hikaridai, 'Keihanna Science City', 6190288 Kyoto, Japan, yusuke.ikeda@atr.jp), Shiro Ise (Urban and Environmental Engineering, Graduate School of Engineering, Kyoto University, C1-386, Kyodai-Katsura, Nisikyo-ku, Kyoto-si, 6158540 Kyoto, Japan, ise@archi.kyoto-u.ac.jp), Satoshi Nakamura (Advanced Telecommunication Research Institute International, 2-2-2 Hikaridai, 'Keihanna Science City', 6190288 Kyoto, Japan, satoshi.nakamura@atr.jp)

Based on the boundary surface control (BSC) principle, a new recording/reproduction system is developed to realize high fidelity three-dimensional sound field reproduction. Theoretically, by using this new system, physically faithful reproduction could be achieved in any acoustic environments. Sound recording/reproduction systems based on the BSC principle require many loudspeakers and many microphones. In this new system, the microphone array system to record 3D sound field consists of 70 microphones, and the loudspeaker system to reproduce the recorded 3D sound field consists of 62 full-range units and 8 sub-woofer units. To evaluate the ability of this system, the wave front which is measured in the sound-



proofed room is compared with the reconstructed wave front within this system. The experiment shows that the reconstructed secondary wave front is very similar to the primary wave front in lower frequency.

**4pEAf5. Three-Dimensional FDTD Method for Analysing of the acoustic characteristics of Compact Acoustic Reproduction Systems.** Yuichiro Koga (Kansai University, Kobe-Nishiku-Gakuenhigasimati5-1-512, 651-2102 Kobe, Japan, koga0814@joho.densi.kansai-u.ac.jp)

We propose a method for analysing of the acoustic characteristics of compact acoustic reproduction systems with Three-Dimensional FDTD method. We reproduce actual structure of compact acoustic reproduction systems and take the frequency characteristic of sound pressure level at a point with Three-Dimensional FDTD method. To compare the frequency characteristic of sound pressure level of actual measurement of compact acoustic reproduction systems to measurement of the proposed method, we show availability of the proposed method for analysing of the acoustic characteristics of compact acoustic reproduction systems. The proposed method could explain actual phenomena on compact acoustic reproduction systems well.

**4pEAf6. Sound localization in multiple regions: theory and applications.** Yang-Hann Kim (Center for Noise and Vibration Control, Korea Advanced Institute of Science and Technology, 4114, Department of Mechanical Engineering, Guseong-dong, Yuseong-gu, 305-701 Daejeon, Republic of Korea, yanghannkim@kaist.ac.kr), Ji-Ho Chang (Center for Noise and Vibration Control, Korea Advanced Institute of Science and Technology, Dep. of Mechanical Engineering, Guseong-dong, Yuseong-gu, 305-701 Daejeon, Republic of Korea, chang.jiho@gmail.com), Jin-Young Park (Center for Noise and Vibration Control, Korea Advanced Institute of Science and Technology, Dep. of Mechanical Engineering, Guseong-dong, Yuseong-gu, 305-701 Daejeon, Republic of Korea, jypark1979@kaist.ac.kr)

It is often required to have several listening zones, which allows us to have different sounds that we select. For example, in a room, someone wants to listen sound from TV set, and other wants to listen music. In a car, a driver might wants to hear the information coming from his/her navigator system, and the passenger at the back side wants have a quiet zone so that he/she can sleep. To accomplish this kind of acoustic zones, we need to generate multiple sound zones by using multiple speakers. The performance has to be evaluated in accordance with how well one can listen the sound that is expected to have. We proposed to maximize the acoustic contrast between the zones that are defined. The basic concept associated with this approach was proposed by Choi and Kim [J. Acoust. Soc. Am., Vol.111(4), 1695-1700, Apr. 2002.], but this paper extend this fundamental idea to multiple zone cases. Theoretical formulation which shows what we have proposed is well addressed and several practical cases, including car audio system will be demonstrated. [This work was supported by the Korea Science and Engineering Foundation (KOSEF) through the National Research Lab. Program funded by the Ministry of Science and Technology (M10500000112-05J0000-11210)]

**4pEAf7. Sound focused personal audio system design: Performance improvement in acoustic contrast control by spatial weighting for obtaining spatially averaged acoustic potential energy.** Jin-Young Park (Center for Noise and Vibration Control, Korea Advanced Institute of Science and Technology, Dep. of Mechanical Engineering, Guseong-dong, Yuseong-gu, 305-701 Daejeon, Republic of Korea, jypark1979@kaist.ac.kr), Ji-Ho Chang (Center for Noise and Vibration Control, Korea Advanced Institute of Science and Technology, Dep. of Mechanical Engineering, Guseong-dong, Yuseong-gu, 305-701 Daejeon, Republic of Korea, chang.jiho@gmail.com), Chan-Hui Lee (Center for Noise and Vibration Control, Korea Advanced Institute of Science and Technology, Dep. of Mechanical Engineering, Guseong-dong, Yuseong-gu, 305-701 Daejeon, Republic of Korea, Chance99@kaist.ac.kr), Yang-Hann Kim (Center for Noise

and Vibration Control, Korea Advanced Institute of Science and Technology, 4114, Department of Mechanical Engineering, Guseong-dong, Yuseong-gu, 305-701 Daejeon, Republic of Korea, yanghannkim@kaist.ac.kr)

In acoustic contrast control [J.-W. Choi and Y.-H. Kim, J. Acoust. Soc. Am. 111, 1695 (2002)], spatially averaged acoustic potential energy is used as a representative spatial parameter because acoustic contrast control aims to maximize spatially averaged acoustic potential energy ratio between acoustically bright and dark zone. Therefore, spatial averaging process should be concerned carefully because control performance is sensitive to how to determine bright and dark zone. We have already got successful result in feasibility study for personal audio system with over 20dB difference between bright (frontal) and dark (side) zone [C.-H. Lee et al., J. Acoust. Soc. Am. 111, 3053 (2007)], without any spatial weighting in spatial averaging of acoustic potential energy. Recently, we're trying to improve the control performance by giving spatial weighting in spatial averaging process because how to give spatial weighting has to do with the improvement of control performance to satisfy the original purpose of personal audio system more closely. In this paper, it will be covered the investigation for how to give spatial weighting in averaging process and shown experimental evaluations for a sound focused personal audio system. [Supported by the Korea Science and Engineering Foundation (KOSEF) through the National Research Lab. Program funded by the Ministry of Science and Technology (M10500000112-05J0000-11210)]

**4pEAf8. 3D sound field rendering under non-idealized loudspeaker arrangements.** Alois Sontacchi (Institute of Electronic Music and Acoustics, Inffeldgasse 10 / 3, 8010 Graz, Austria, sontacchi@iem.at), Franz Zotter (Institute of Electronic Music and Acoustics, Inffeldgasse 10 / 3, 8010 Graz, Austria, zotter@iem.at), Robert Höldrich (Institute of Electronic Music and Acoustics, Inffeldgasse 10 / 3, 8010 Graz, Austria, hoeldrich@iem.at)

The approach to realise periphonic sound field reproduction based on spherical harmonics (multi-pole theory) has already been well-known as Ambisonics and Higher Order Ambisonics, respectively. By the aid of an N-dimensional orthogonal set of vectors any arbitrary source free sound field can be described. Reproduction is realized by projection of the encoded sound field on a regular loudspeaker distribution over a spherical surface. The used set of vectors exhibits a defined hierarchic with interesting symmetries. In the original scheme sound sources represented by plane waves (sources in far distance) can be encoded independent of the decoding process on the regular loudspeaker layout. Usually, in practice - in contrast to theory, 3D loudspeaker layouts are requested for the upper hemisphere. This restriction is caused by the physical configuration. First of all that demand bounds the reproduction of sound sources to the upper area. Furthermore caused by these facts idealized regular layouts considering the 3 dimensions are impossible. Within this contribution we will show how the symmetries of the spherical harmonics can be used to obtain optimized decoding rules and to overcome insufficient irregular loudspeaker arrangements.

**4pEAf9. Measurements of head-related transfer function in sagittal and frontal coordinates.** Takashi Nakado (Nagoya University, Furo-cho, Chikusa-ku, 4648603 Nagoya, Japan, nakado@sp.m.is.nagoya-u.ac.jp), Takanori Nishino (Nagoya University, Furo-cho, Chikusa-ku, 4648603 Nagoya, Japan, nishino@media.nagoya-u.ac.jp), Kazuya Takeda (Nagoya University, Furo-cho, Chikusa-ku, 4648603 Nagoya, Japan, kazuya.takeda@nagoya-u.jp)

3D sounds can be generated by using a head-related transfer function (HRTF), which is defined as the acoustic transfer function between a sound source and the entrance to the ear canal. Since HRTF depends on a subject and the sound source direction, many HRTF measurements were conducted. In most case, HRTFs were measured in horizontal coordinates. However, HRTF measurements in other coordinates are also useful. In previous researches, HRTFs measured in sagittal coordinates were used to investigate the relation between spectral cues and vertical angle perception. Although HRTF measurement in frontal coordinates is rarely conducted, there is an advantage to measure HRTFs densely in the front and rear where sound localizations are very sensitive. Therefore, we measured HRTFs for about

2,300 directions in sagittal and frontal coordinates and constructed a database. The measurements were conducted in a soundproof chamber with two head-and-torso simulators (B&K 4128 and KEMAR). The HRTF database can be downloaded at <http://www.sp.m.is.nagoya-u.ac.jp/HRTF/>.

**4pEaf10. A New Sound Field Reproduction Method Based on the Spatial Covariance.** Yoshinori Takahashi (NHK Science & Technical Research Laboratories, 1-10-11 Kinuta, Setagaya-ku, 157-8510 Tokyo, Japan, yoshinori@ieee.org), Akio Ando (NHK Science & Technical Research Laboratories, 1-10-11 Kinuta, Setagaya-ku, 157-8510 Tokyo, Japan, ando.aio@nhk.or.jp)

This work investigates a sound field reproduction method based on the sound field statistics. The covariance among recorded multi-channel signals might represent the relative relationship and the mutual magnitude characteristics between each point in the sound field that time invariant. The author Y. Takahashi et al. proposed a reproduction method minimizing the difference in the spatial covariance between the original and reproduced sound fields [19th ICA, RBA15-012]. We call the method "Sound field Reproduction method based on the sPAatial Covariance (SORPAC)". However it has not been clarified that the theoretical background and the characteristics in listening. The SORPAC doesn't require the information of sound source locations and transfer functions. In this work, we described the reproduction theory of SORPAC in frequency domain. And we showed the relationship between SORPAC and the general wave surface control. Then we confirmed that the SORPAC is able to reproduce the sound source direction without the

wave surface control. As a application of SORPAC, we demonstrated a multi-channel contents down-mixing experiment and evaluated the result from the binaural-listening point of view.

**4pEaf11. Optimal array pattern synthesis with desired magnitude response.** Alexander Mattioli Pasqual (Universidade Estadual de Campinas, Rua Mendeleiev, 200, Cidade Universitária "Zeferino Vaz", 13083-970 Campinas, Brazil, pasqual@fem.unicamp.br), José Roberto Arruda (Universidade Estadual de Campinas, Rua Mendeleiev, 200, Cidade Universitária "Zeferino Vaz", 13083-970 Campinas, Brazil, arruda@fem.unicamp.br), Philippe Herzog (Laboratoire de Mécanique et d'Acoustique - CNRS, 31 chemin Joseph Aiguier, 13402 Marseille, France, herzog@lma.cnrs-mrs.fr)

Letting Euclidean norm be the performance parameter, the task of finding the best approximation of a complex function in a finite dimension subspace leads to a convex optimization problem that can be easily solved by the least-squares method. However, this procedure leads to a sub-optimal solution in applications that have no phase requirements on the approximated function. In this case, semidefinite programming has been used to obtain optimal magnitude responses. In this work, this non-convex optimization problem is dealt with by using an iterative method based on the least-squares, which is illustrated on directivity synthesis by spherical loudspeaker arrays. Usually, instead of synthesize directly the desired pattern, the strategy adopted is to reproduce its truncated spherical harmonic representation. The truncation order is determined by the number of drivers of the spherical array. It is shown that truncation error and signal powers can be significantly reduced if phase error is neglected, providing potential means to improve directivity synthesis for applications requiring only magnitude response.

THURSDAY AFTERNOON, 3 JULY 2008

ROOM 202/203, 2:20 TO 6:20 P.M.

## Session 4pED

### Education in Acoustics: Acoustics Education Software

Ralph T. Muehleisen, Cochair

*Illinois Institute of Technology, Civil, Architectural and Environmental Engineering, 3201 S. Dearborn St., Room 228, Chicago, IL 60616, USA*

Catherine Potel, Cochair

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

### Invited Papers

2:20

**4pED1. Education software for numerical acoustics.** Olivier Dazel (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, olivier.dazel@univ-lemans.fr)

Numerical computations tools and scientific computing programs are more and more used by acousticians. This induces that higher education institutions have to include numerical techniques in their programs. Several general aspects should be considered while designing such courses; they are the initial level of the students, their ability with mathematics and/or physics and the aim of the formation. Particular aspects on the choice of a type of software and the methodology need also to be studied. This talk will detail these aspects around the numerical courses at Université du Maine in Le Mans (France) whose formations ranges are from sound and vibration technicians to PhD students. Some typical examples and numerical tools will be presented to illustrate the purpose.

2:40

**4pED2. Computer tools for architectural acoustics education.** Jian Kang (School of Architecture, University of Sheffield, Western Bank, S10 2TN Sheffield, UK, j.kang@sheffield.ac.uk)

Calculating/simulating acoustic performance of architectural spaces and building elements plays an important role in architectural education. Unfortunately, architectural students often hesitate to use theoretical formulae/models commonly applied by acousticians. This study aims to develop simple calculation/simulation tools to help architectural students to understand basic acoustic principles.

Focusing on the effectiveness of various key parameters, as well as on scientific visualisation way of presenting teaching materials, five tools have been developed: (1) sound distribution behind an environmental noise barrier, with parameters including barrier height, source-barrier distance, and source height; (2) sound distribution in a rectangular street canyon, with parameters including street length, width, building height, boundary absorption coefficient, air absorption, and the height of receiver plane; (3) reverberation time calculation in a rectangular space, with parameters including room dimensions and boundary absorption coefficients, where a database of absorption coefficients is also included; (4) absorption of perforated panel absorbers, especially micro-perforated panel absorbers, with parameters including hole size, hole spacing, panel thickness, and depth of airspace; (5) digital audio animation for urban soundscape design, considering idealised cross-streets and squares in a 2D environment, where the sound file with multiple sources can be played back, with reverberation effects. [Work supported by the Theodore John Schultz Grant]

3:00

**4pED3. Live computer-based demonstrations in musical acoustics education.** Donald M. Campbell (Edinburgh University, 4201 JCMB, Kings Buildings, Mayfield Road, EH9 3JZ Edinburgh, UK, d.m.campbell@ed.ac.uk), David Skulina (Edinburgh University, 4201 JCMB, Kings Buildings, Mayfield Road, EH9 3JZ Edinburgh, UK, dskulina@ph.ed.ac.uk)

Live demonstrations play a vital pedagogical role in any science education programme. For a subject as interdisciplinary as musical acoustics, embracing as it does mathematics, physics, engineering, psychology, cognitive science, neurophysiology, musicology and music technology, demonstrations of psychoacoustical phenomena and performances on musical instruments provide invaluable methods for involving and intriguing a general audience. The recent dramatic increase in the memory and speed of laptop computers has opened new possibilities for live computer-based demonstrations. This talk explores some of these possibilities, including demonstrations of the analysis and display of musical instrument sounds using real-time spectrographic programs and the explanation of important musical instrument characteristics using physical modelling synthesis.

3:20

**4pED4. A computer model to study the properties of guided waves.** Michael J. Lowe (Imperial College London, Department of Mechanical Engineering, SW7 2AZ London, UK, m.lowe@imperial.ac.uk)

The author's research group has a specialist interest in developing guided wave techniques for the Non Destructive Testing (NDT) of structures. Guided waves travel along the structure and so can be used to inspect large lengths or areas very much faster than the traditional point-by-point ultrasonic methods. The development of these inspection methods requires careful study and understanding of the properties of the guided waves. To address this need, the author's team has created a general purpose modelling tool. This can model waves in waveguides consisting of an arbitrary number of layers which can be flat or cylindrical, elastic or damped, isotropic or anisotropic solids, or perfect or viscous fluids, and can be immersed in a fluid or embedded in a solid. The primary output of the program is the dispersion curves, that is the frequency-velocity relationships of any modes which could travel in the structure. The program also predicts the attenuation of the modes, caused by leakage into surrounding materials or by material damping, and the mode shapes. The presentation will explain the basis of the modelling tool and illustrate the way in which it can be used to understand the guided waves.

3:40-5:20 Posters

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

### *Invited Paper*

5:20

**4pED5. DELTAEC is also an acoustics teaching tool.** Steven Garrett (Penn State, Applied Research Laboratory, PO Box 30, State College, PA 16804, USA, sxg185@psu.edu)

A major revision of the Los Alamos thermoacoustics code, renamed DELTAEC, was released in 2007. It takes advantage of a user-friendly, menu-driven Windows environment and has indigenous plotting capabilities. DELTAEC is a differential equation solver that analyzes one-dimensional acoustical networks defined by a series of "segments" representing ducts, compliances, speakers, etc. This talk will relate experiences using this software to teach a first-year graduate core course on acoustics in fluids. Examples include illustration of effective length and quality factor of Helmholtz resonators, as well as the more challenging standing wave solutions within a resonator of variable cross-section. The plotting feature allows immediate illustration of the pressure and velocity fields as well as power flow within the resonator. DELTAEC will also adjust gas mixture concentration to match a specified frequency. Segments representing electrodynamic loudspeakers, radiation loading, and flow resistance in porous media will be used to demonstrate the coupled-oscillatory behavior of a bass-reflex enclosure's complex electrical impedance vs. frequency. Calculation of the modes of a catenoidal horn of finite length will be presented. The "thermo-physical property" feature provides fluid and solid properties at the students' choice of pressure, temperature and frequency, making it useful as a "handbook" for other assignments.

5:40

**4pED6. CNAQ: a MATLAB acquisition tool for computing transfer functions over audio cards.** Guillaume Pellerin (CNAM Paris, 5, rue du Vert Bois, 75003 Paris, France, guillaume.pellerin@cnam.fr)

Since transfer function measurements require very precise buffer management, it is not easy to find any free and/or educational software which processes them with consumer audio cards, cheaper than professional ones but useful. Some people often use musical sequencers to generate and acquire signals but data can't be easily analyzed in other specific engineering codes because the lengths of the samples can't be easily fixed for example. Therefore, we decided to develop a specific software for general audio acoustic measurements with these main constraints: choose a programming language compatible with existing educational and professional tools, promote multi-platform interpreted languages, be able to use consumer audio cards, publish the source code under the terms of a free software license. CNAQ provides a graphical interface under MATLAB to process sweep sine measurements for any kind of acoustical or electronic application in the audio range. The current features of the software and some development ideas, such as implementing convolution based Farina's method, will be also presented.

6:00

**4pED7. Noise Metric - Environmental Noise Analysis Software.** Milan Stojiljković (Faculty of Technical Sciences, Department of Environmental Engineering, Trg Dositeja Obradovića 6, 21000 Novi Sad, Yugoslavia,

noisemetric@gmail.com), Vlado Delic (Faculty of Technical Sciences, Department of Telecommunications, Laboratory for Acoustics and Speech Technologies, Trg Dositeja Obradovića 6, 21000 Novi Sad, Yugoslavia, vdelic@uns.ns.ac.yu)

Noise Metric software suite addresses the need for a modern low-cost acquisition, post-processing and analysis system that accommodates professional grade instruments used in typical environmental noise assessments. Software is designed to comply with national and international legislation and to provide a comprehensive overview of environmental acoustics theory in an interactive help file. Educational use is emphasized by providing measurement wizards and example case histories, with included demonstration measurement files for student self study. Acknowledging tight student financial budgets, the system is not limited to use with professional grade equipment. It can be configured for less expensive and less precise operation by providing integration with standard PC sound boards and low-cost microphones. Current beta version focuses on community noise and traffic noise assessment, with future modules planned for occupational noise and basic acoustic calculations. Off line analysis module is supplemented by a real time module for recording and logging the measurements and a browser for viewing the analysis results, noise rating level calculation and report generation. Every attempt is being made to automate as many analysis tasks as possible and thus minimize the human induced errors in the assessment process. Preliminary product testing was successfully completed by comparison with proprietary software products.

THURSDAY AFTERNOON, 3 JULY 2008

AMPHI MAILLOT, 1:40 TO 7:00 P.M.

## Session 4pMUa

### Musical Acoustics: Control of Natural and Synthetic Musical Sounds I

Marcelo M. Wanderley, Cochair

*Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada*

Jean Kergomard, Cochair

*Laboratoire de Mécanique et d'Acoustique CNRS UPR-7051, 31, Chemin Joseph Aiguier, Marseille Cedex 20, 13402, France*

### Invited Papers

1:40

**4pMUa1. Systems control theory for sound synthesis: an overview.** Brigitte D'Andréa-Novel (Ecole des Mines de Paris, 60 Bd. Saint-Michel, 75006 Paris, France, brigitte.dandrea-novel@ensmp.fr), Denis Matignon (Univ. Toulouse; ISAE. Dept. MIA, 10 av. Edouard Belin, 31055 Toulouse, France, denis.matignon@isae.fr)

The aim of this talk is to present some classical techniques and methods from control theory to deal with the problem of sound synthesis based on physical models. A first part will be devoted to the case of finite-dimensional state space, illustrated by: 1. Kelly-Lochbaum structure for 1D wave equations; 2. Realization theory for passive impedance of rational type. The second part considers the infinite-dimensional case. More precisely the authors will present: 1. The operator associated to PDEs in acoustics, its diagonalization leading to "Riemann invariants" useful to study stability properties; 2. A Lyapunov method to elaborate boundary control to ensure stabilization of an equilibrium regime and some kind of robustness; 3. Diffusive realizations of fractional time derivatives, useful when modeling visco-thermal losses in acoustical ducts. These tools are well adapted to the class of "wind instruments".



**4pMUa2. Extracting Reed Control Parameters Using Acoustic Measurements and Inverse Filtering.** Tamara R. Smyth (Simon Fraser University, 2728 West 5th Ave, Vancouver, BC V6K 1T4, Canada, tamaras@cs.sfu.ca), Jonathan Abel (Stanford University, 752 College Ave, Menlo Park, CA 94025, USA, jonathan.abel@comcast.net)

The control of virtual musical instruments often relies on either a specially-developed controller on which the performer has usually not gained sufficient virtuosity to play musically, or an existing multipurpose general controller with control parameters not always easily, or intuitively, mapped to the synthesis parameters of the virtual instrument being performed. A response to this problem is to obtain control information from a musical performance where the performer uses an instrument with which s/he is sufficiently familiar. In this work, we incorporate a previously developed measurement technique to transform a measured clarinet signal into a sequence of pulses corresponding to the reed displacement as a function of time. The measurement technique, shown to obtain accurate reflection functions from various tube structures, is used to obtain a filter modeling the bore and bell of the wind instrument used in the performance. The "reed pulse" waveform is then isolated by inverse filtering the measured clarinet signal. The characteristics of this residual waveform, which evolve with the performer's control of the instrument, may be extracted and remapped to the synthesis parameters of a physical model.

### Contributed Papers

2:20

**4pMUa3. Modeling and automatic control of a slide flute.** Brigitte D'Andréa-Novel (Ecole des Mines de Paris, 60 Bd. Saint-Michel, 75006 Paris, France, brigitte.dandrea-novel@ensmp.fr), Benoit Fabre (Institut Jean Le Rond d'Alembert / LAM (UPMC / CNRS / Ministère Culture), 11, rue de Lourmel, 75015 Paris, France, fabreb@ccr.jussieu.fr), Jean-Michel Coron (Lab. Jacques-Louis Lions, UPMC and Institut Universitaire de France, 4 place Jussieu, 75005 Paris, France, Jean-Michel.coron@math.u-psud.fr)

In this paper, we consider the problem of modeling and control of a slide flute: a kind of recorder without finger holes but which is ended by a piston mechanism to modify the length of the resonator. A previous study has been done, but with a very simple boundary condition for the mouth, corresponding to an ideal situation assuming that the acoustic pressure is zero at the entrance of the resonator. In this work, we have taken into account a more realistic model, describing the coupling effects between the jet and the pipe. The jet is obtained by blowing through a flue channel and formed by flow separation at the flue exit, and finally directed towards a sharp edge, called the labium. The resulting structure will be described by two linear PDEs coupled with nonlinear ODEs describing the boundary conditions: for the mouth, taking into account the jet dynamics, and for the piston. A modal analysis is performed using the linearized boundary conditions which can also be used to compute the suitable blowing pressure and the suitable pipe

length to obtain a desired fundamental frequency or equivalently a desired pitch. This will constitute the basis of our control algorithm.

2:40

**4pMUa4. Physical parameters of the violin bridge changed by active control.** Henri Boutin (Institut Jean le Rond d'Alembert, Lab. d'Acoustique Musicale, 11, rue de Lourmel, 75015 Paris, France, boutin@lam.jussieu.fr), Charles Besnainou (Institut Jean le Rond d'Alembert, Laboratoire d'Acoustique Musicale, 11, rue de Lourmel, 75015 Paris, France, chbesnai@ccr.jussieu.fr)

The physical parameters of a violin bridge have a significant influence on the tonal colouration of its sound. The resonance peaks of the bridge shape the response of the violin body. Reinicke and Cremer developed a simple bridge model that shows a typical broad frequency peak around 2.5kHz, because it incorporates the coupling to the violin body and the soundpost. By using the same model, Jim Woodhouse revealed the effect of some parameters of the bridge (mass, stiffness and foot spacing) on the instrument frequency response. Here the parameters of the violin resonance peaks are changed in real time, by applying an active control method. Such a technique, very useful in noise reduction, enabled to change separately the position and the shape of each peak of the bridge input admittance. On the bridge, 2 actuators and an accelerometer are placed at strategic positions in order to change the peak frequency and the damping factor values. The system behaviour is controlled by a Digital Signal Processor. Some sound results achieved with a real violin back up the theoretical equations.

### Invited Paper

3:00-3:20 Break

3:20

**4pMUa5. Between the frog and the tip - bowing gestures and bow-string interaction in violin playing.** Anders Askenfelt (Dept. of Speech, Music and Hearing, Royal Institute of Technology (KTH), Lindstedtsvägen 24, SE-100 44 Stockholm, Sweden, andersa@speech.kth.se)

The motion of the bow gives a natural visualization of a string performance. Watching the player's bowing may augment the communicative power of the music, but all relevant bow control parameters are not easy to capture by a spectator. The string player controls volume of sound and tone quality continuously by coordination of three basic bowing parameters (bow velocity, bow-bridge distance, and bow force), which set the main conditions for the bow-string interaction. At a more detailed level of description, the tilting of the bow, which among other things controls the effective width of the bow hair, enters into the model. On a longer time scale, pre-planned coordination schemes ('bowing gestures'), including the basic bowing parameters and the angles between the path of the bow and the strings, builds the performance. Systems for recording bowing parameters will be reviewed and results from old and current studies on bowing gestures presented. The player's choice and coordination of bowing parameters are constrained both in attacks and 'steady-state' according to bow-string interaction models. Recent verifications of these control spaces will be examined. Strategies for starting notes and examples of how players do in practice will be presented and compared with listeners' preferences.

*Invited Papers*

5:00

**4pMUa6. The guitar as an extension of the voice - Phonetic gestures underlying guitar timbre perception and description.**

Caroline Traube (Laboratoire informatique, acoustique et musique, Faculté de musique, Université de Montréal, C.P. 6128, succursale Centre-Ville, Montréal, QC H3C 3J7, Canada, caroline.traube@umontreal.ca), Maryse Lavoie (Laboratoire informatique, acoustique et musique, Faculté de musique, Université de Montréal, C.P. 6128, succursale Centre-Ville, Montréal, QC H3C 3J7, Canada, maryse.lavoie@umontreal.ca)

The guitar is an instrument that gives the player great control over timbre. Different plucking techniques involve varying the finger position along the string and the inclination between the finger and the string. Guitarists perceive subtle variations of these parameters and have developed a rich vocabulary to describe the tones they produce on their instrument. Vocal imitations - onomatopoeias - is another way to intuitively describe instrumental tones. The data that we collected showed that guitar tones can be consistently associated with different types of vowels depending on instrumental gesture parameters, suggesting that guitar tones can evoke "phonetic gestures" as defined in the motor theory of speech perception. In addition, these phonetic gestures seem to be at the origin of a large subset of commonly used verbal descriptors: open, oval, round, thin, closed, nasal, hollow, etc. These analogies can be explained by the comb-filter shaped spectral envelope of plucked-string instruments which emphasizes energy in the region of vocal formants. We conclude that, when technical difficulties have been surmounted, performers can use their musical instrument as an extension of their voice, the musical instruments allowing a virtuosic control of sound parameters similar to those involved in the paralinguistic elements of speech.

5:20

**4pMUa7. Production and perception of goal-points and coarticulations in music.** Rolf Inge Godøy (University of Oslo, Department of Musicology, P.B. 1017 Blindern, N-0315 Oslo, Norway, r.i.godoy@imv.uio.no), Alexander Refsum Jensenius (University of Oslo, Department of Musicology, P.B. 1017 Blindern, N-0315 Oslo, Norway, a.r.jensenius@imv.uio.no), Kristian Nymoen (University of Oslo, Department of Musicology, P.B. 1017 Blindern, N-0315 Oslo, Norway, krisny@hf.uio.no)

From our studies of sound-related movement (<http://musicalgestures.uio.no>), we have reason to believe that both sound-producing and sound-accompanying movements are centered around what we call goal-points, meaning certain salient events in the music such as downbeats, or various accent types, or melodic peaks. In music performance, these goal-points are reflected in the positions and shapes of the performers' effectors (fingers, hands, arms, torso, etc.) at certain moments in time, similar to what is known as keyframes in animation. The movement trajectories between these goal-points, similar to what is known as interframes in animation, may often demonstrate the phenomenon of coarticulation, i.e. that the various smaller movement are subsumed under more superordinate and goal-directed movement trajectories. In this paper, we shall present a summary of recent human movement research in support of this scheme of goal-points and coarticulations, as well as demonstrate this scheme with data from our ongoing motion capture studies of pianists' performance and other researchers' motion capture data.

5:40

**4pMUa8. Perception, verbal description and gestural control of piano timbre.** Caroline Traube (Laboratoire informatique, acoustique et musique, Faculté de musique, Université de Montréal, C.P. 6128, succursale Centre-Ville, Montréal, QC H3C 3J7, Canada, caroline.traube@umontreal.ca), Michel Bernays (Laboratoire informatique, acoustique et musique, Faculté de musique, Université de Montréal, C.P. 6128, succursale Centre-Ville, Montréal, QC H3C 3J7, Canada, michel.bernays@gmail.com), Madeleine Bellemare (Laboratoire informatique, acoustique et musique, Faculté de musique, Université de Montréal, C.P. 6128, succursale Centre-Ville, Montréal, QC H3C 3J7, Canada, madeleine.bellemare@umontreal.ca)

Musical expressivity in virtuosic pianistic performance relies heavily on timbre and performers call upon a vast vocabulary to describe the nature of their sound; examples of adjectives include velvety, metallic, bright, round and dark. The present study aims to determine whether this vocabulary, its perceptual meaning and the gesture applied to obtain the sounds it describes, stand as consensual among pianists. The relations between timbre, articulation, register and dynamics are also examined. Nearly 100 verbal descriptors were collected as well as the description of the associated gestures. Some timbres are specific to certain dynamic levels and others are the result of a combination of at least two sonic elements into one resulting sound object, where articulation, - the relation between one note to the next - plays a crucial role. A subset of these adjectives has been selected for further study. A professional pianist performed, on a computer-controlled recording acoustic piano, three short project-designed pieces with several timbres, as designated by adjectives. The excerpts were also captured with microphones to serve as stimuli for a timbre recognition task, both in free form and by selection, to which a group of 17 pianists performed with great accuracy.

6:00

**4pMUa9. Study of brass performer gestures.** René E. Causse (IRCAM, 1 Place Igor Stravinsky, 75004 Paris, France, [causse@ircam.fr](mailto:causse@ircam.fr)), Vincent Freour (IRCAM-CNRS (UMR 9912 STMS), 1, Place Igor Stravinsky, 75004 Paris, France, [Vincent.Freour@ircam.fr](mailto:Vincent.Freour@ircam.fr))

Brass instrument playing requires the musician to control his respiratory gesture and the elastic properties of his lips. This raises the question of musician gesture optimisation and strategy in order to complete a musical exercise. It also makes gesture characterization very hard to conduct in a non-invasive way. On the other hand, it is possible to measure some control parameters (linked to the respiratory and lip-adjustment gesture) like lip force applied on the mouthpiece and mouth air pressure. These parameters

measurements and also the specific mouthpiece receiver developed to measure lip force with a minimum interference to the player are presented. Respiratory flows during live playing are also evaluated thanks to the calibration of respiratory belts used on the thoracic and abdominal regions of musicians. Details of the method for carrying out this type of measurement and preliminary results are reported. During this measurement sound recording and analysis are also conducted thanks to a set of audio descriptors. Links between control parameters measurements and sound characterisation are examined. Parallel development of an automated artificial mouth, used for experimental validations, is also outlined. [Work, within the CONSONNES project, is lead with the support of the French Research National Agency ANR].

*Invited Papers*

6:20

**4pMUa10. Refining mapping strategies to improve the sound quality of physically controlled synthesis.** Vincent Verfaillie (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, [vincent@music.mcgill.ca](mailto:vincent@music.mcgill.ca)), Arnaud Rebillout (4 le Cabut, 33390 Cars, France, [arnaud.rebillout@gmail.com](mailto:arnaud.rebillout@gmail.com)), Philippe Guillemain (Laboratoire de Mécanique et d'Acoustique CNRS UPR-7051, 31, Chemin Joseph Aiguier, 13402 Marseille Cedex 20, France, [guillemain@lma.cnrs-mrs.fr](mailto:guillemain@lma.cnrs-mrs.fr)), Marcelo M. Wanderley (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, [marcelo.wanderley@mcgill.ca](mailto:marcelo.wanderley@mcgill.ca))

A new technique, called 'physically controlled synthesis', is being developed to improve the controllability and sound quality of digital sound synthesis. It can be seen as a mapping strategy that combines a synthesis model based on a physical model (controllability) with a signal model based on additive synthesis (sound quality), and a database of pre-analyzed natural instrumental sounds. A key point is the computation of perceptually relevant timbre descriptors that interface the two synthesizers to provide additive synthesis data by navigating the database and selecting neighbour additive frames to morph. To limit the latency introduced, we developed specific algorithms to extract sound features from the physical model; the database size was increased offline by using a higher control sampling rate. In addition to previous works by the authors, devoted to the permanent regime and using a 2D indexation of the additive database, transients handling is achieved through a 3D search in a frame-by-frame basis that ignores the natural time unfolding, hence allowing to choose the best frame with the proper pitch in addition to the two other sound descriptors.

6:40

**4pMUa11. Inverse problem: Recovering the full-state of a simplified model of a trumpet-like instrument from the radiated pressure.** Thomas Hélie (IRCAM, 1, place Igor Stravinsky, 75004 Paris, France, [thomas.helie@ircam.fr](mailto:thomas.helie@ircam.fr)), Brigitte D'Andréa-Novel (Ecole des Mines de Paris, 60 Bd. Saint-Michel, 75006 Paris, France, [brigitte.dandrea-novel@ensmp.fr](mailto:brigitte.dandrea-novel@ensmp.fr))

We consider a simplified model of a trumpet-like instrument composed of a valve (including the mechanics of the lips), a jet (coupled with the valve dynamics), and an acoustic pipe excited by the jet and radiating in the air. A special care is devoted to the energy balance of the whole system and its dissipative property. This leads us to introduce a model of a non-stationary jet. In a second step, the problem of the observation of the full-state (that is, the position and the velocity of the lips, the flow and the pressure in the jet and in the acoustic pipe) from the radiated pressure is analyzed. This problem can be recasted as a problem of control engineering, using a so-called neutral system (differential system including the delayed state and its time derivative). We show how the energy balance can help to solve this problem and define a naturally well-posed observator. As a last step, this work is recasted in the context of a more general inverse problem: What control input (pressure in the mouth, parameter of the lips, etc) must be used to feed the model in order to recover a target sound?

## Session 4pMUB

## Musical Acoustics: Bowed and Keyboard Stringed Instruments II (Poster Session)

Seiji Adachi, Cochair

*Fraunhofer Institute for Building Physics*

Simon Félix, Cochair

*Laboratoire d'Acoustique de l'Université du Maine*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pMUB1. Experimental modal analysis of bows.** Enrico Ravina (University of Genoa - Centre of Research on Choral and Instrumental Music (MUSICOS), Via Opera Pia 15 A, 16145 Genova, Italy, enrico.ravina@unige.it), Paolo Silvestri (Univ. of Genoa - DIMEC, Via Opera Pia 15 A, 16145 Genova, Italy, p.silvestri@unige.it), Antonino Airenti (Bow Maker, Via Marussig 13, 16100 Genova, Italy, inf@baroquebows.net)

The vibration performances of bow instruments are usually studied developing numerical and experimental modal analyses of the body of the instrument or of their parts (tailpiece, bridge, fingerboard, neck). The dynamic contribution of the bow is less considered, but the mutual actions generated between bow and strings are conditioned by the mechanical features of the bow. The paper analyzes the dynamic behaviour of different kind of bows (clip-in frog and screw-driven frog) and different geometries (baroque, modern,...) through experimental modal analyses. Bows are instrumented with micro-accelerometers and excited by a micro-hammer. Frequency Response Functions up to 2500 Hz allow a good characterization of the bow and show significant differences about the modal shapes. The study is integrated with the experimental strain analysis, based on micro strain gauges glued on the body; the very small dimensions of the transducers (2.5 mm) allow, from one side, a not intrusive analysis but, from another side, require specific contrivances of mounting. Details on the integrated experimentations are focussed and discussed.

**4pMUB2. Spherical mapping of violins.** Enrico Ravina (University of Genoa - Centre of Research on Choral and Instrumental Music (MUSICOS), Via Opera Pia 15 A, 16145 Genova, Italy, enrico.ravina@unige.it), Paolo Silvestri (Univ. of Genoa - DIMEC, Via Opera Pia 15 A, 16145 Genova, Italy, p.silvestri@unige.it), Pio Montanari (Master Violinmaker, Via delle Compere 2, 16100 Genova, Italy, montanaripio@libero.it), Guido De Vecchi (Musician, Via S. Maria di Castello 37, 16100 Genova, Italy, ensemble@ilfalcone.com)

An original experimental approach oriented to the evaluation of the acoustic performances of violins is described. Starting from 14th Century the violins family have passed through significant changing, strongly influencing their sound. The "instrumental music" requires different parts for different voices: the violin family changes its mechanical structure following this requirement. The structural elements are modified in order to adequate the sound to the aesthetic taste of the historical period. Mechanical modifications involve geometry, relative positions and structural characteristics of fixed and mobile parts. The consequence is a significant alteration of vibration and acoustic responses. The paper describes a systematic approach oriented to evaluate the acoustic performances of differently mounted violins (baroque, classical, modern) by means a spherical mapping of the generated sound. A workbench based on a semicircular structure carrying an array of

10 microphones interfaced to a portable acquisition unit, has been designed and realized. The violin, played by the musician in anechoic chamber or in representation room, is located at the centre of this semicircle: changing the relative angular position between the violin and the array acoustic spherical maps describing the actual sound emission are generated. A systematic comparison among differently mounted violins is shown and discussed.

**4pMUB3. On the generation of axial modes in the nonlinear vibration of strings.** Alexandre Watzky (Laboratoire de Mécanique Physique, Université Paris 12 - Fac. des Sciences & Technologie, 61 av. du Gal. de Gaulle, 94010 Créteil, France, watzky@univ-paris12.fr)

Vibrating strings are known to be nonlinear. Transverse vibrations induce axial motion as well as a modulation of the string's tension. An overview of the nonlinear models of the vibrating string has been presented at the 150th Meeting of the ASA. It has been shown that if temporal considerations are prevalent from an acoustical point of view, the involved coupling mechanisms are mainly ruled by the spatial shape of the modes. The purpose of this communication is to investigate the specific case of transverse-axial interactions and to examine the possible generation of axial modes through transverse vibrations. It also gives the opportunity to discuss the common hypotheses. Despite the complex modal couplings that can occur, this results enable to avoid or favor axial modes in the design of the string itself or of the boundary conditions i.e. the bridge.

**4pMUB4. The application of thin plate theory to the time evolution of musical instruments.** Christopher Gorman (Rollins College, Department of Physics, Winter Park, FL 32789, USA, cgorman@rollins.edu), David Parker (Rollins College, Department of Physics, Winter Park, FL 32789, USA, dparker@rollins.edu), Connor Ballance (Rollins College, Department of Physics, Winter Park, FL 32789, USA, ballance@vanadium.rollins.edu), Donald Griffin (Rollins College, Department of Physics, Winter Park, FL 32789, USA, griffin@vanadium.rollins.edu), Thomas Moore (Rollins College, Department of Physics, Winter Park, FL 32789, USA, tmoore@rollins.edu)

The steady state dynamics of many musical instruments can be modeled using thin-plate theory. It has been shown that the normal-mode frequencies of systems as diverse as the orchestral crotale and piano soundboard can be accurately calculated within this approximation, and it is therefore reasonable to ask if thin-plate theory can be applied to model the time evolution of these instruments. To answer this question we have modeled a struck flat plate using finite differences and compared the decay of the eigenmodes to experimental results. We find that the time evolution of the motion of a struck thin plate is not well described under the thin-plate approximation even when the modal frequencies are predicted accurately. We propose that



mode coupling between longitudinal and transverse modes requires that a full three dimensional model be used to predict the time evolution even when the plate is thin.

**4pMUb5. Considering the effect of hammer shank flexibility using a multibody dynamic simulation model of a piano action mechanism with string contact.** Chandrika P. Vyasaryani (University of Waterloo, Department of Systems Design Engineering, Waterloo, ON N2L 3G1, Canada, cpvyasar@engmail.uwaterloo.ca), Stephen Birkett (University of Waterloo, Department of Systems Design Engineering, Waterloo, ON N2L 3G1, Canada, sbirkett@real.uwaterloo.ca), John McPhee (University of Waterloo, Department of Systems Design Engineering, Waterloo, ON N2L 3G1, Canada, mcphoe@real.uwaterloo.ca)

A piano action mechanism converts a pianist's mechanical input into acceleration of the piano hammer, which impacts the string for tone generation. We present a multibody dynamic model of the mechanism, considering the differences when hammer shank flexibility is included as compared to a rigid shank. The model is developed using the graph theoretic approach and includes the hammer-string interaction. A Rayleigh beam model including complete second order deformation field is used for simulating hammer shank flexibility. The governing partial differential equation is discretized using Ritz approach considering Taylor monomials as assumed modes. A convergence study confirms that two bending modes and one axial mode are sufficient to represent the hammer shank deformation. The vibrating string is modeled using a standard modal analysis procedure. The many contacts between components of the mechanism include significant sliding during contact; for these contacts a modified Hunt-Crossley law is used to represent the normal force, and interface friction is handled using a Cull and Tucker friction model. The results of parametric studies show the effect of hammer head friction on the dynamics of the mechanism during string impact, as well as the influence of hammer shank flexibility on the frequency response of the string.

**4pMUb6. A complex model for piano action.** José Lozada (Laboratoire de Mécanique des Solides, Ecole Polytechnique, 91128 Palaiseau Cedex, France, jose.lozada@cea.fr), Xavier Boutillon (Laboratoire de Mécanique des Solides, Ecole Polytechnique, 91128 Palaiseau Cedex, France, boutillon@lms.polytechnique.fr), Moustapha Hafez (Laboratoire des Interfaces Sensorielles, CEA-LIST, 18 route du Panorama, BP 6, 92265 Fontenay-aux-Roses Cedex, France, moustapha.hafez@cea.fr)

Available mechanical models of piano action are extremely simplified in view of the complexity of an actual mechanism. Several arguments will be presented which indicate that more dynamical complexities than those of-

fered by the current models are considered as intrinsic properties of good piano actions. A more complete model of a grand piano action during the attack of a note will be presented which includes the six degrees of freedom of the action, all considered as rigid bodies: the key, the damper, the whippen, the repetition lever, the jack, and the hammer. In this model, coupling features are represented by lumped elements: Coulomb friction, linear and nonlinear springs. Methods for characterizing each element of the model will be presented. The final result will be given in terms of dynamical equations, coupling equations, and geometrical constraints.

**4pMUb7. Feature exaggeration in scale performance on the piano.** Shinya Morita (Graduate School of Science and technology, Ryukoku University, 1-5, Yokotani, Oe-cho, Seta, 520-2194 Otsu, Shiga, Japan, t040550@mail.ryukoku.ac.jp), Norio Emura (Faculty of Engineering, Doshisha University, 1-3, Tataramiyakodani, 610-0321 Kyo-tanabe, Kyoto, Japan, etf1702@mail4.doshisha.ac.jp), Masanobu Miura (Graduate School of Science and technology, Ryukoku University, 1-5, Yokotani, Oe-cho, Seta, 520-2194 Otsu, Shiga, Japan, miura@rins.ryukoku.ac.jp), Seiko Akinaga (Department of Education, Shukugawa Gakuin College, 6-58, Kosikiiwa-cho, 662-8555 Nishinomiya, Hyogo, Japan, akinaga@shukugawa-c.ac.jp), Masuzo Yanagida (Faculty of Engineering, Doshisha University, 1-3, Tataramiyakodani, 610-0321 Kyo-tanabe, Kyoto, Japan, myanagid@mail.doshisha.ac.jp)

This paper proposes a set of parameters for describing features of scale-playing on the piano. The parameter set consists of 15 parameters, among which 12 are three sets of four parameters  $\pi_{i1}$  through  $\pi_{i4}$  where  $i = \{1, 2, 3\}$  distinguishes three basic features; onset time, velocity, and duration. Each of these basic features is modeled as the sum of a global curve and the deviation from it, where the spline interpolation is employed using locally averaged points, or representative points, as the points to be passed. The local average is calculated for each sequence of notes played without finger crossing. The suffix  $j$  in  $\pi_{ij}$  distinguishes the standard deviations ( $j=0$ ), the rms deviation from the spline curve ( $j=1$ ), the range of the curve ( $j=2$ ), the rms difference between successive notes ( $j=3$ ), and the rms of the spline curve from the metronomic line ( $j=4$ ). All parameters are made controllable with slider bars from 0% to 200% for synthesizing suppressed performance or exaggerated performance, where 100% represents the original performance. Proposed parameter set is expected to be useful in self-training of piano, as it can indicate the features and undesirable habits of the player by setting values above 100% in exaggerated form.

## Session 4pMUc

## Musical Acoustics: Plucked Stringed Instruments II (Poster Session)

François Gautier, Cochair

*Laboratoire d'Acoustique de l'Université du Maine*

Chris Waltham, Cochair

*University of British Columbia*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pMUc1. The physics of the American five-string banjo.** Laurie Stephey (Rollins College, Department of Physics, Winter Park, FL 32789, USA, lstephey@rollins.edu), Thomas Moore (Rollins College, Department of Physics, Winter Park, FL 32789, USA, tmoore@rollins.edu)

We report on a systematic study of the American five-string banjo, which was undertaken in an effort to understand the dynamics of this complex instrument. The deflection shapes of the membranous head were studied and catalogued using time-averaged electronic speckle pattern interferometry. The impedance presented to the strings was measured using laser Doppler vibrometry coupled with an integrated force sensor and harmonic driver. Additionally, time resolved spectral analysis of the plucked strings was used to quantify the characteristic decay of the coupled string/membrane system, while time-resolved interferometric studies of the membrane have led to a better understanding of the motion after a string is plucked. All of these investigations help to reveal the importance of the various parameters that affect the sound of this unusual instrument.

**4pMUc2. The interaction between the strings and soundboard of a harp.** Chris Waltham (University of British Columbia, Department of Physics & Astronomy, Vancouver, BC V6T 1Z1, Canada, cew@phas.ubc.ca)

The harp is an instrument with a set of plucked strings that excite the sound board directly, without the medium of a bridge. The strings are positioned at an acute angle to the plane of the sound board. The quality of the sound produced depends on the motion of the string, which is non-planar, and its interaction with the resonances of the sound board. The interaction is intrinsically non-linear as the soundboard responds to changes in both the angle and the tension of the string. To avoid the difficulties of string-string interactions on a real harp, a small test "instrument" has been constructed with a single string and a variable-angle sound board. The string and sound board motions have been measured simultaneously. Preliminary results will be presented.

**4pMUc3. Categorization of guitars from bridge admittance measurements.** François Gautier (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, francois.gautier@univ-lemans.fr), Jean-Loïc Le Carrou (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le

Mans, France, jean-loic.le\_carrou@univ-lemans.fr), Maarten Hol (LAUM, CNRS, Université du Maine, Lab. d'Acoustique Université du Maine, UMR CNRS 6613, 72085 Le Mans Cedex 9, France, M.Hol@student.tue.nl), Vincent Doutaut (Institut Technologique Européen des Métiers de la Musique, 71 Av. O. Messiaen, 72000 Le Mans, France, Vincent.Doutaut@itemm.fr)

The acoustical characteristics of the guitar depend on the instrument makers' choices concerning the geometry, the material and the assembly techniques. The aim of this paper is to define criteria permitting the discrimination of guitars according to their acoustic characteristics. These criteria can be used, for example, by instrument makers to test the repeatability of their making process. Evaluation of the guitars' quality from this categorization is beyond the scope of this paper. A low-cost portable system allowing bridge admittance measurements has been designed and used on 4 groups of classical guitars, each being composed of about 10 similar instruments. A statistical study shows that the tested instruments can clearly be differentiated according to (1) the modal parameters associated to the first 2 vibroacoustic modes (air mode A0 and first soundboard mode T1), and to (2) a 'merit indicator', close to the one defined by C. Barlow (Proc.I.O.A., vol 19, Pt 5, 1997, 69-78). This 'merit indicator' is computed from the mean value of the bridge admittance and an estimation of the critical frequency of the soundboard.

**4pMUc4. Modal analysis and transient string response of solid body electric bass guitars with effects of the instrumentalist.** Dave Woolworth (Oxford Acoustics, Inc., 356 CR 102, Oxford, MS 38655, USA, dave@oxfordacoustics.com), Henry A. Scarton (Rensselaer Polytechnic Institute, Department of Mechanical Engineering, Troy, NY 12180, USA, scarton@rpi.edu)

This paper documents modal analysis of seven solid body electric basses. The basses are modeled in one dimension of motion normal to the face of the instrument in both free-free state and with boundary conditions imposed by a musician. The results are compared and analyzed in terms of bending and torsion. Analysis is done regarding open string harmonic content of the electromagnetic output over time versus the resonant frequencies of the instruments. Results of finite element analysis modeling of a simplified bass guitar structure will be presented, and significance of damping effects on harmonic output will be considered in terms of the listener.

## Session 4pMUD

## Musical Acoustics: Wind Instruments II (Poster Session)

Seiji Adachi, Cochair

*Fraunhofer Institute for Building Physics*

Simon Félix, Cochair

*Laboratoire d'Acoustique de l'Université du Maine*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pMUD1. Controllable pitch-bending effects in the accordion playing.**

R. Llanos-Vazquez (Dpto. Física Aplicada 1. Escuela Técnica Superior de Ingeniería, Alameda de Urquijo s/n, 48013 Bilbao, Spain, rucurdu@orange.es), M. J. Elejalde-García (Dpto. Física Aplicada 1. Escuela Técnica Superior de Ingeniería, Alameda de Urquijo s/n, 48013 Bilbao, Spain, mariajesus.elejalde@ehu.es), E. Macho-Stadler (Dpto. Física Aplicada 1. Escuela Técnica Superior de Ingeniería, Alameda de Urquijo s/n, 48013 Bilbao, Spain, erica.macho@ehu.es)

The accordion employs reeds in which the tongue is mounted outside the reed frame in such a way that sounding is normally possible only on one direction of airflow. Under normal operating conditions the reeds behave as blown-closed. Pitch bending technique allows players to make a controllable glide, non tempered glissando, from one pitch to another. Pitch-bending frequency shift, defined as the percentage of the ratio of the frequency change and the original frequency has been measured in a series of experiments. Some of the results involving the dependence of the function on pitch, direction of the bellows movement, cassoto possibilities and harmonic number are reported here. If the player qualifies, he/she can make controllable pitch-bending effects where the glissando may fall an exact semitone.

**4pMUD2. Study the mouthpiece of the txistu.**

A. Agos-Esparza (Dpto. Física Aplicada 1. Escuela Técnica Superior de Ingeniería, Alameda de Urquijo s/n, 48013 Bilbao, Spain, bckagesa@ikasle.ehu.es), M. J. Elejalde-García (Dpto. Física Aplicada 1. Escuela Técnica Superior de Ingeniería, Alameda de Urquijo s/n, 48013 Bilbao, Spain, mariajesus.elejalde@ehu.es), E. Macho-Stadler (Dpto. Física Aplicada 1. Escuela Técnica Superior de Ingeniería, Alameda de Urquijo s/n, 48013 Bilbao, Spain, erica.macho@ehu.es)

The txistu is a three finger holed recorder from the Basque Country. The evolution of the txistu involves different kinds of wood, bore length, finger hole positioning and it being made of two or three pieces. The txistu also has a unique mouthpiece, made up by a small metallic piece of the pipe and also a metal reed which was introduced centuries ago. The distance between the pipe and the reed can be adjusted to fit the user's preferences. By using a blowing machine and data acquisition software, experiments study the influence of the shape and inclination of the reed on the values of the harmonic content and the sonority of the final note.

**4pMUD3. Automated fingering services for woodwinds: development of a "virtual clarinet".**

Yakov Kulik (University of New South Wales, Music Acoustics, School of Physics, NSW 2052 Sydney, Australia,

ykulik@phys.unsw.edu.au), Andrew Botros (University of New South Wales, Music Acoustics, School of Physics, NSW 2052 Sydney, Australia, ABotros@phys.unsw.edu.au), John Smith (University of New South Wales, Music Acoustics, School of Physics, NSW 2052 Sydney, Australia, john.smith@unsw.edu.au)

The Virtual Flute is a popular web service that recommends alternative fingerings for difficult passages, timbre variations, intonations or multiphonics. Its database was generated by a machine-learned expert system analysing waveguide models for all 39,744 fingerings. The relatively simple geometry of the flute and its tone holes allowed a simple yet accurate model. The development of similar systems for other woodwinds faces greater modelling and computational challenges. For example, the clarinet has a more complex geometry, with tone holes whose radius and length vary by factors of 4.2 and 2.8. Further, it has several million different fingerings. To achieve the required accuracy, individual measurements of each hole separately and of mouthpiece and bell, as well as several dozen fingering examples, were used to determine parameters of a still simple waveguide model. The model uses conical and cylindrical segments with parallel and shunt impedances at junctions, representing tone holes. This approach of incrementally enhancing our waveguide model allows computational advantages: an efficient, woodwind-generic software framework is built that can adapt to the instrument of interest. We report interim results with such a system, with further potential applications in the design of woodwind instruments and other acoustic duct systems.

**4pMUD4. Source-resonator modeling: a rough paradox.**

Laurent P. Millot (IDEAT (University Paris 1), ENS Louis-Lumière, 7 allée du Promontoire, 93161 Noisy-le-grand, France, l.millot@ens-louis-lumiere.fr)

Using the example of a short pipe loaded by a free reed located upwards, we will underline the fact that a reflection function based modeling will not give the same results (and synthesized sounds) as an unsteady incompressible and lossless Bernoulli description while this model is commonly assumed to be a low-frequency limit case of the first one, and that the second description is the only valid one. This paradox will be explained and it will be underlined why the so-called electro-acoustical analogies do not constitute a low-frequency limit case of reflection function or input impedance descriptions, based on a waves paradigm. Extension to striking reeds and/or longer pipes, as found in saxophone, clarinet or sheng for instance, will be discussed and a strange coupling between a priori incompatible models, commonly used in Musical Acoustics notably, will be described.

**4pMUd5. Sound quality of the 16'' french bagpipe: wood influence.** Mathieu Paquier (LISyC EA 3883, 6 avenue Victor Le Gorgeu, CS 93837, 29238 Brest Cedex 3, France, mathieu.paquier@univ-brest.fr), Raphaël Jeannin (LISyC EA 3883, 6 avenue Victor Le Gorgeu, CS 93837, 29238 Brest Cedex 3, France, jeanninraphael@yahoo.fr)

The most played among the bagpipes from the centre of France is the 16-inch musette, called in this way because of the length of the melodic pipe (oboe). Though these instruments are less known than the binou from Brittany or the Great Highland Bagpipe, the number of players and makers is, nowadays, in increase because of their easy play and quasi-chromatic scale.

Whereas the Breton and Scottish bagpipes are always made of very hard woods, some 16'' musettes are fabricated with softer woods. This difference is certainly related to the flexibility of the 16'' oboe double-reed. At first, we recorded some short musical sequences played on 16'' musettes made of 5 different woods (African Ebony, Santos Rosewood, Boxwood, African Blackwood and Service Tree), then some listeners (specialist and naïve) were asked to give their feedback about the quality of the recorded sounds. In a second set of experiments, we recorded some single notes played with these various bagpipes and analyzed the acoustic features of the recorded signals. At last, the perceptual results were compared with the physical parameters obtained in the second experiment.

THURSDAY AFTERNOON, 3 JULY 2008

P2-D, LEVEL 2, 3:40 TO 5:20 P.M.

## Session 4pMUe

### Musical Acoustics: Control of Natural and Synthetic Musical Sounds II (Poster Session)

Marcelo Wanderley, Cochair

*Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ.*

Jean Kergomard, Cochair

*Laboratoire de Mécanique et d'Acoustique CNRS UPR-7051*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

#### 4pMUe1. Exploration of timbre variations in music performance.

Mathieu Barthet (CNRS-Laboratoire de Mécanique et d'Acoustique, 31 Chemin Joseph Aiguier, 13402 Marseille, France, barthet@lma.cnrs-mrs.fr), Philippe Guillemain (Laboratoire de Mécanique et d'Acoustique CNRS UPR-7051, 31, Chemin Joseph Aiguier, 13402 Marseille Cedex 20, France, guillemain@lma.cnrs-mrs.fr), Richard Kronland-Martinet (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, kronland@lma.cnrs-mrs.fr), Solvi Ystad (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, ystad@lma.cnrs-mrs.fr)

Timbre has seen during the twentieth century a growing attention from both musicians and scientists, the former to expand a traditional music system until now governed by the structures of pitches, the latter to better understand how timbre is produced and how it is processed by the perceptive and cognitive systems. In the music performance context, many studies deal with the role of timing and dynamics, but much fewer are dedicated to the one of timbre. The works presented here aim at showing the importance of timbre variations from two points of view: the production of sounds and their perception. The relations between the control parameters (mouth pressure and reed aperture) of a simplified physics-based clarinet synthesis model and the generated timbres have been investigated. Experiments have further been done to measure the within-individual consistency of timbre variations of a clarinet player repeating several instances of a musical excerpt from a Bach piece while keeping the same musical intention. Brightness variations characterized by the time-varying Spectral Centroid showed a strong consistency across the repetitions. The influence of such brightness variations on the perceived musical quality of a performance was then assessed thanks to an analysis-transformation-synthesis paradigm.

#### 4pMUe2. Active Sound Design of a Bassoon.

Frederic Konkel (TU Berlin, Institute of Fluid Mechanics and Engineering Acoustics, Einsteinufer 25, Sekr. TA 7, D-10587 Berlin, Germany, f.konkel@advacoustics.de), Andre Jakob (TU Berlin, Institute of Fluid Mechanics and Engineering Acoustics, Einsteinufer 25, Sekr. TA 7, D-10587 Berlin, Germany, kontakt@advacoustics.de), Frank Heintze (Staatskapelle Berlin, Staatsoper Unter den Linden, Unter den Linden 7, D-10117 Berlin, Germany, f.heintze@advacoustics.de), Michael Möser (TU Berlin, Institute of Fluid Mechanics and Engineering Acoustics, Einsteinufer 25, Sekr. TA 7, D-10587 Berlin, Germany, moes0338@mailbox.tu-berlin.de)

Possibilities of influencing the sound characteristics of a woodwind by means of active noise control were investigated. The woodwind used in this investigation was a bassoon. The first step of the investigation consisted of a set of measurements of the sound spectra of different tones and volumes by means of measurement microphones inside and outside the instrument. Additionally measurements of the radiation characteristics of the instrument were performed by means of the acoustic camera. An experimental setup was designed with the instrument driven mechanically by compressed air. The aim of the work was to change the sound characteristics of the bassoon by changing individual harmonics of the tones by means of a loudspeaker attached to the bassoon. The loudspeaker was attached to the bassoon via a tube. Suitable positions for the connection of the tube to the instrument were examined as well as suitable possibilities for the generation of the reference signal necessary for the feedforward control scheme. The experimental setup will be presented here as well as results of the measurements with and without active sound design with an analog controller.

4p THU. PM



**4pMUe3. The influence of control parameters and physical parameters on reed instruments playing.** Jean Kergomard (Laboratoire de Mécanique et d'Acoustique CNRS UPR-7051, 31, Chemin Joseph Aiguier, 13402 Marseille Cedex 20, France, kergomard@lma.cnrs-mrs.fr), Philippe Guillemain (Laboratoire de Mécanique et d'Acoustique CNRS UPR-7051, 31, Chemin Joseph Aiguier, 13402 Marseille Cedex 20, France, guillemain@lma.cnrs-mrs.fr), Fabrice Silva (CNRS-Laboratoire de Mécanique et d'Acoustique, 31 Chemin Joseph Aiguier, 13402 Marseille, France, silva@lma.cnrs-mrs.fr), Christophe Vergez (CNRS-Laboratoire de Mécanique et d'Acoustique, 31 Chemin Joseph Aiguier, 13402 Marseille, France, vergez@lma.cnrs-mrs.fr)

An attempt to summarize the influence of both control and physical parameters on production and radiation of sound of reed instruments is given. Some parameters, such as the shape of the resonator, are fixed by the instrument maker, or chosen by the instrumentalist itself: the reed, and the fixation of the reed on the mouthpiece. These parameters, named physical parameters, are fixed when playing. The second kind of parameters can be totally or partially controlled when playing: the fingering is an obvious one, then the way the reed is pinched by the lip, with an effect on both the reed opening and the reed dynamics, finally the mouth pressure and the shape of the vocal tract. The influence of these parameters is discussed with respect to several attributes of the sound: the various thresholds (normal sound at pianissimo level, extinction at fortissimo level), the control of different regimes, the shape of transients, the playing frequency, the spectrum. The state of present knowledge is given, with emphasis of further research that should

be done, and a comparison with the cases of sound controllers and artificial mouth is discussed.

**4pMUe4. A setup for measurement of bowing parameters in bowed-string instrument performance.** Erwin Schoonderwaldt (Dept. of Speech, Music and Hearing, Royal Institute of Technology (KTH), Lindstedtsvägen 24, SE-100 44 Stockholm, Sweden, schoondw@kth.se), Matthias Demoucron (IRCAM, 1 Place Igor Stravinsky, 75004 Paris, France, demoucron@ircam.fr), Nicolas Rasamimanana (IRCAM, 1 Place Igor Stravinsky, 75004 Paris, France, rasamimanana@ircam.fr)

An accurate measurement of instrumentalists' actions in playing situations is the basis for several research topics such as musical performance analysis, control of sound synthesis algorithms or effective validation of theoretical results in musical acoustics. We present a setup for a complete and accurate measurement of bowing parameters in bowed-string instrument performance, with minimum interference for the player. The method is based on the combined use of motion capture and sensors attached to the bow. An optical motion capture system was used to track the position and orientation (6 DOF) of the bow and the instrument. In addition, sensors were attached to the frog of the bow for measuring bow acceleration in different axes and bow force exerted on the strings. Both sensors can be easily attached to any bow. We will describe how the data from the different systems are combined for an accurate calculation of bowing parameters, such as bow-bridge distance, bow velocity, bow acceleration, bow angles and bow pressure. Finally, we will present examples of measurements and briefly discuss some potential applications.

THURSDAY AFTERNOON, 3 JULY 2008

P2-D, LEVEL 2, 3:40 TO 5:20 P.M.

## Session 4pMUf

### Musical Acoustics: Virtual Musical Instruments II (Poster Session)

Julius Smith, Cochair  
*Stanford Univ.*

Antoine Chaigne, Cochair  
*ENSTA*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pMUf1. Simulation of Acoustic Pressure and Flow Velocity in Human Glottis.** Petr Sidlof (Academy of Sciences of the Czech Republic, Narodni 3, 11720 Prague, Czech Republic, sidlof@it.cas.cz), Antoine Chaigne (ENSTA, Chemin de la Hunière, 91761 Palaiseau, France, antoine.chaigne@ensta.fr), Olivier Doare (ENSTA, Chemin de la Hunière, 91761 Palaiseau, France, olivier.doare@ensta.fr), Olivier Cadot (ENSTA, Chemin de la Hunière, 91761 Palaiseau, France, olivier.cadot@ensta.fr)

The pressure and velocity fields in coronal plane along the vibrating vocal folds were studied using a finite element mathematical model. The shapes of the vocal folds were specified according to data measured on excised human larynges in phonation position. The mathematical model of the

flow is based on 2D incompressible Navier-Stokes equations adapted to deal with the time-variable shape of the domain, caused by vocal fold vibration. The numerical simulations allow to observe closely various flow features related to phonation - flow separation in the glottis, Coanda effect or vortex shedding. The numerical results were verified experimentally by Particle Image velocimetry (PIV) on a physical vocal fold model. In addition to acoustic, subglottal pressure and impact intensity measurements, flow velocity fields were recorded in the domain immediately above glottis. Analysis of the PIV images gives good insight into the dynamics of the supraglottal flow. Among other applications, the results should be used in the future for singing voice simulations based on physical models.

**4pMUf2. A synthesis and analysis framework for wind instrument sounds based on the digital pulse forming principle.** Michael Oehler (Institute for applied Musicology and Psychology, Saarstrasse 1A, 50677 Koeln, Germany, michael.oehler@iamp.info), Christoph Reuter (University of Cologne - Musicological Institute, Beethovenstrasse 4, 50674 Koeln, Germany, info@chr-reuter.de)

A digital real-time-capable analysis- and synthesis-system for wind instrument sounds, based on the pulse forming theory, has been developed. The rediscovered model for the sound generating process of wind instruments rests upon the idea, that wind instrument sounds can basically be put down to its excitation impulses, which independently of the fundamental always behave according to the same principles. First realised in the analogue wind instrument synthesizer Realton Variophon (1975), the sound synthesis method has currently been transferred onto a digital platform [supported by the German Research Foundation]. Instrument specific algorithms control the pulse width and shape according to the applied pitch and dynamic values. That way subtle sound nuances that can be produced on acoustic wind instruments as well as real timbre modulation may be synthesized by just modifying a single parameter (i.e. breath pressure). In order to validate the performance of the developed framework, several perception experiments were conducted subsequently.

**4pMUf3. Perceptual effects of radiation control with a multi-loudspeaker device.** Nicolas Misdariis (IRCAM - UMR CNRS 9912, Equipe Perception et Design Sonores, 1, place Igor Stravinsky, 75004 Paris, France, misdariis@ircam.fr), Alexandre Lang (Université de Technologie de Compiègne, E.A. Costech - Groupe de Suppléance Perceptive, BP 60319, 60206 Compiègne Cedex, France, lang.alexandre@gmail.com), Brian F. Katz (LIMSI-CNRS, B.P. 133, 91403 Orsay, France, brian.katz@limsi.fr), Patrick Susini (IRCAM - UMR CNRS 9912, Equipe Perception et Design Sonores, 1, place Igor Stravinsky, 75004 Paris, France, susini@ircam.fr)

This study investigates the perceptual issue of acoustical radiation control with the following hypothesis: radiation control can reduce the perceptual gap between a sound coming from an acoustical source (e.g., an instrument) and a sound coming from an electro-acoustical device (e.g., the recording of an instrument played by a loudspeaker). The work is technically supported by a generic multi-loudspeaker device that allows sound reproduction with controlled directivity patterns in a given number of spatial dimensions. The conducted experiment involves two distinct sound corpus: speaking/singing voice (recorded and spatially measured) and a struck plate (synthesized and spatially computed with modeled directivity). Incremental levels of radiation pattern control, in terms of precision of the reproduction, are also considered by combining several diffusion systems and different directivity patterns. Participants are asked to evaluate the stimuli on three semantic differential scales: source width (largeur), distance (distance) and realism (réalisme). The results show a global significant effect of radiation on sound perception: the acoustical/electro-acoustical gap is significantly reduced through radiation control, especially when considering the scales of width and distance. Theoretical principles, technical elements, experimental set-up and overall results will be presented, detailed and discussed during the presentation.

**4pMUf4. Measuring and Modeling Violin Sound Radiation for Sound Equalization.** Alfonso Perez Carrillo (Pompeu Fabra University, Ocata 1,3°, 08003 Barcelona, Spain, aperez@iua.upf.edu), Jordi Bonada (Pompeu Fabra University, Ocata 1,3°, 08003 Barcelona, Spain, jbonada@iua.upf.edu), Esteban Maestre (Pompeu Fabra University, Ocata 1,3°, 08003 Barcelona, Spain, emaestre@iua.upf.edu), Enric Gueaus (Pompeu Fabra University, Ocata 1,3°, 08003 Barcelona, Spain, egueaus@iua.upf.edu), Merlijn Blaauw (Pompeu Fabra University, Ocata 1,3°, 08003 Barcelona, Spain, mblaauw@iua.upf.edu)

During a performance a violin is put into vibration and this vibration is radiated to the air. Depending on the distance and direction of the listener, he perceives a slightly different sound. Sound radiation at each point in the space is determined by the impulse response or transfer function between vibration of the violin and sound pressure at that point. There is variety of methodologies to obtain acoustic impulse responses of violins. They differ

mainly in the way of exciting the violin, the point of excitation, the position where the acoustic response is measured and how excitation and response signals are deconvolved. In this work we measure violin vibration with a bridge pickup and we 1) propose a method to calculate the transfer function between signals from the bridge pickup and a microphone, 2) use a 3D motion tracker to get the position of the microphone respect to the violin so that we can calculate the response for different directions and distances and 3) we model sound radiation as a "reference" impulse response and a parametric filter whose response depends on distance and orientation with respect to the "reference" position.

**4pMUf5. Virtual electric guitars and associated audio effects in Faust and C++.** Julius O. Smith (Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180, USA, jos@ccrma.stanford.edu)

Advances in computing technology, both in hardware and software, are enabling new levels of performance in real-time for virtual musical instruments based on acoustic principles. This paper is concerned with software implementation technology for such instruments, written in high-level languages that compile down to efficient low-level implementations on a wide variety of platforms. Specific results will be presented for the case of virtual electric guitars and associated digital audio effects expressed in the Faust and C++ languages, and compiled to become plugins for VST, Pure Data, and other real-time performance environments. This work builds upon a previous paper at the Linux Audio Conference (LAC-2008), and laboratory module written for the RealSimple project at CCRMA (<http://ccrma.stanford.edu/realsimple/>).

**4pMUf6. Fine tuning of guitar sounds with changed top plate, back plate and rim geometry using a whole body 3D Finite-Difference model.** Rolf Bader (University of Hamburg, Institute of Musicology, Neue Rabenstr. 13, 20354 Hamburg, Germany, R\_Bader@t-online.de)

The change of radiated sounds from guitars is investigated in terms of the fine tuning of these sounds applying slight changes to the guitar body, instrument builders would do with their guitars. As changes of these geometries of real guitars can only be perceived when the instrument is put together, the question is mostly open how different the whole instrument would have sounded with slight changes. So here the virtual guitar can be of great use. In the model, the top plate thickness, fan bracing, rim thickness and back plate geometry were changed in a linear way. Then these results could be associated directly to a change in only one geometry parameter. It is not assumed, that the sounds fit one existing guitar perfectly. Rather the overall influence of these changes on the sounds is investigated to show an overall behaviour. As the eigenfrequencies of the guitar body are that many, that a more or less closed resonance curve is achieved (without so-called 'dead spots'), the changes in the sounds are measured in the following terms: changes in the initial transient of the sounds, its brightness using the spectral centroid, its density using a spectral entropy measurement and amplitude fluctuations of the different sound partials.

**4pMUf7. Nonlinear vibrations of impacted rectangular plates. Comparison between numerical simulation and experiments.** Cédric Camier (ENSTA, Chemin de la Hunière, 91761 Palaiseau cedex, France, cedric.camier@ensta.fr), Kevin Arcas (ENSTA, Chemin de la Hunière, 91761 Palaiseau, France, arcas@ensta.fr), Stefan Bilbao (University of Edinburgh, Room 7306B, JCMB, King's Bldgs., Mayfield Rd., EH9 3JZ Edinburgh, UK, sbilbao@staffmail.ed.ac.uk)

Large amplitude vibrations of free-edge rectangular plates, subjected to an impulse excitation is addressed. In particular, this work is devoted to the analysis and simulation of a nonlinear von Kármán plate equations and by associate experimental investigations. Time domain simulations are achieved using implicit finite differences (FD) schemes recently developed by Bilbao. These energy-conserving methods guarantee the stability of the algorithm. To compare with simulations results, an experimental setup which allows reproducible impulse excitation and measurements by laser vibrometry has been developed. The time history of the force applied to a rectangular steel plate is recorded and this signal is used as excitation term in the simulations. We perform a parametric study, with both experimental and

numerical approaches, by increasing gradually the amplitude of forcing. Non-linear phenomena, such as pitch glide and chaotic behaviour are observed and discussed.

**4pMUf8. Active control of a vibrating string.** Edgar Berdahl (Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180, USA, eberdahl@ccrma.stanford.edu), Guenter Niemeyer (Stanford Univ., Mech. Eng., Bldg. 530, Stanford, CA 94305, USA, gunter.niemeyer@stanford.edu), Julius O. Smith (Stanford Univ., Center for Computer Research in Music and Acoustics (CCRMA), Dept. of Music, Stanford, CA 94305-8180, USA, jos@ccrma.stanford.edu)

We discuss the specifics of applying active feedback control to a vibrating string. Using sensors, actuators, and digital controller hardware, we make the acoustics of the string programmable, yet the string retains its tangible qualities. As a consequence, fretting, bowing, and plucking controlled and uncontrolled strings have similar physical consequences. Consider that any controller emulating a network of springs, masses, and dashpots attached to the string is a passive controller. To allow the string's acoustics to be programmable over a wide range, we should be able to implement passive controllers. This means that there must be at least one linear and collocated sensor/actuator pair. We explain how to construct such a pair in the laboratory. Finally, we explain one controller particular to one-dimensional systems such as vibrating strings. Whenever the sensor detects a pulse arriving, the actuator emits a new pulse. The output spectrum consists of a harmonic series proportional to the sampling of the product of the pulse's Fourier transform and the transfer function from the actuator to the sound recording device. Sound examples are presented. Finally, we discuss an

open source environment we have created for adjusting controller parameters in real time from standard computer music software.

**4pMUf9. Interfacing Digital Waveguide with CORDIS ANIMA networks.** Alexandros Kontogeorgakopoulos (ACROE-ICA (Institut National Polytechnique de Grenoble), 46 av. Félix Viallet, 38001 Grenoble, France, alexandros.kontogeorgakopoulos@imag.fr), Claude Cadoz (ACROE-ICA (Institut National Polytechnique de Grenoble), 46 av. Félix Viallet, 38001 Grenoble, France, claudc.cadoz@imag.fr)

The aim of this article is mainly to offer a link between the Digital Waveguide and the CORDIS-ANIMA physical modeling formalism. The first one is highly a modular lumped physical modeling and simulation system based on the mass-interaction paradigm while the second one offers accurate and efficient discrete time distributed models synthesized typically by delay lines and scattering junctions, in combination with digital filters. Both of them are widely developed and used in the domain of computer music field by scientists and artists. Although Digital Waveguide models have already been combined with Wave Digital Filters, they have never been exploited and integrated with CORDIS ANIMA networks. Wave Digital Filters are lumped models which are based on a scattering theoretic formulation which simplifies interfacing to Digital Waveguide models in contrast with the CORDIS ANIMA models. This research investigates the similarities of those formalisms, as well as focuses on the advantages of each modeling technique and proposes a real time computable interface between them. Moreover it results as well in a common convenient structural representation of their computational algorithms using signal processing block diagrams. These hybrid models were designed directly by their block diagrams, simulated and run in differ time using the Simulink software package of the matlab technical computing language.

THURSDAY AFTERNOON, 3 JULY 2008

ROOM 250A, 2:00 TO 3:40 P.M.

## Session 4pNSa

### Noise and EURONOISE: Vibration Perception I

Patricia Davies, Cochair

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

Michael J. Griffin, Cochair

*Institute of Sound and Vibration Research, University of Southampton, Human Factors Research Unit, Southampton, SO17 1BJ, UK*

### Invited Papers

2:00

**4pNSa1. Recent research activities for the assessment of vibration in living environment with respect to human perception in Japan.** Yasunao Matsumoto (Saitama University, 255 Shimo-Ohkubo, Sakura, 338-8570 Saitama, Japan, ymatsum@mail.saitama-u.ac.jp), Sunao Kunimatsu (National Institute of Advanced Industrial Science and Technology, Tsukuba Central 7, 1-1-1 Higashi, 305-8567 Tsukuba, Japan, s.kunimatsu@aist.go.jp), Takashige Ishikawa (Japan Women's University, 2-8-1 Mejiro-dai, Bunkyo, 112-8681 Tokyo, Japan, ishikawa@fc.jwu.ac.jp), Yoshihiro Hirao (Kobayashi Institute of Physical Research, 3-20-41 Higashi-Motomachi, 185-0022 Tokyo, Japan, hirao@kobayashi-riken.or.jp)

The number of complaints against vibration has been increasing gradually in recent years in Japan. This fact may imply problems of the Vibration Regulation Law that was implemented almost 30 years ago so as to regulate vibrations caused by factories and construction works and to mitigate vibration problems caused by road traffic. A group of experts has been investigating an assessment method that may be able to assess recent vibration problems reasonably since 2004. This paper presents a part of group activity that aims at improving the understanding of human vibration perception and applying it in vibration assessment in living environment. Results obtained from experiments involving human subjects to determine perception thresholds of whole-body vibration are summarised.

**4pNSa2. Development of a social survey questionnaire of reactions to vibration in residential buildings.** Henrietta Howarth (Institute of Sound and Vibration Research, University of Southampton, Human Factors Research Unit, SO17 1BJ Southampton, UK, h.howarth@soton.ac.uk), Michael J. Griffin (Institute of Sound and Vibration Research, University of Southampton, Human Factors Research Unit, SO17 1BJ Southampton, UK, M.J.Griffin@soton.ac.uk)

A social survey questionnaire has been developed to determine human responses to vibration in residential environments. The overall aim was to produce a robust methodology for obtaining responses that could be combined with vibration measurements so as to investigate dose-response relationships for vibration in residential buildings. The vibration considered included that from sources outside the control of residents (e.g., road, rail, industrial, construction). This paper describes the development of the questionnaire and explains its structure and content. A review of social surveys and field and laboratory studies of vibration and noise is included. Methods of analysing responses obtained to the questionnaire are summarised.

**4pNSa3. Cross-modality matching of loudness and whole-body vibration strength.** Stephan Töpken (Oldenburg University, Institute of Physics - Acoustics, Carl-von-Ossietzky Str. 9-11, 26111 Oldenburg, Germany, stephan.toepken@uni-oldenburg.de), Michael Bellmann (ITAP GmbH, Marie-Curie Str. 8, 26129 Oldenburg, Germany, Bellmann@itap.de), Reinhard Weber (Oldenburg University, Institute of Physics - Acoustics, Carl-von-Ossietzky Str. 9-11, 26111 Oldenburg, Germany, Reinhard.Weber@uni-oldenburg.de)

In every day live humans are often exposed to noise and vibration simultaneously. Regarding comfort issues inside a car, it is of interest to know, whether the simultaneous perceptions of noise and vibration interfere. Laboratory tests have been carried out with temporally overlapping, partially overlapping and non overlapping presentations of acoustic and whole-body vibration stimuli. Sitting on a rigid chair on a vibration test bench, participants are asked to judge the strength of the excited whole-body vibrations in comparison to the loudness of noise presented via closed headphones. An adaptive method is employed to determine the subjective point of equality (PSE) between both sensory channels. The acoustic stimulus is a 1/2-octave band-pass noise centred at 100 Hz with a fixed noise level, the vibration stimulus is a narrowband noise, also 1/2-octave broad with a centre frequency of 31,5 Hz. The signals have an equal duration of one second. The signal parameters are chosen with respect to specific situations in a car. The PSE's of the loudness and the vibration strength as a function of the temporal order of the acoustic and vibration stimuli exhibit no dependency on the degree of temporal overlap of the stimulus presentation.

**4pNSa4. Thresholds for the perception of fore-and-aft, lateral and vertical vibration by seated persons.** Miyuki Morioka (Institute of Sound and Vibration Research, University of Southampton, Human Factors Research Unit, SO17 1BJ Southampton, UK, M.Morioka@soton.ac.uk), Michael J. Griffin (Institute of Sound and Vibration Research, University of Southampton, Human Factors Research Unit, SO17 1BJ Southampton, UK, M.J.Griffin@soton.ac.uk)

Vibration experienced in transport and in buildings can yield discomfort or annoyance if the vibration exceeds the threshold for vibration perception. Knowledge of thresholds makes it possible to determine which frequencies and directions of low magnitude vibration give rise to perception. The effect vibration frequency (2 to 315 Hz) on absolute thresholds for the perception of whole-body vibration has been determined experimentally with 12 seated persons for each of the three axes of excitation (fore-and-aft, lateral and vertical). The frequency-dependence of the thresholds differed between the three axes. At frequencies, greater than 10 Hz, sensitivity was greatest for vertical vibration. At frequencies less than 3.15 Hz, sensitivity was greatest to fore-and-aft vibration. In all three axes, the acceleration threshold contours at frequencies greater than 80 Hz were U-shaped, suggesting the same psychophysical channel mediated high frequency thresholds for fore-and-aft, lateral and vertical vibration. It is shown that the frequency-dependence of absolute thresholds for the perception of whole-body vibration are not consistent with the frequency weightings used in current standards.

### *Contributed Paper*

**4pNSa5. Is there a perceptive signature of vehicles vibrations?** Maël Amari (PSA Peugeot Citroën, Centre Technique de Vélizy, Route de Gisy, 78943 Vélizy-Villacoublay, France, mael.amari@mpsa.com), Etienne Parizet (Laboratoire Vibrations Acoustique, Insa Lyon, 25 bis, av. J. Capelle, 69621 Villeurbanne Cedex, France, etienne.parizet@insa-lyon.fr), Vincent Roussarie (PSA Peugeot Citroën, Centre Technique de Vélizy, Route de Gisy, 78943 Vélizy-Villacoublay, France, vincent.roussarie@mpsa.com)

The vibro-acoustic comfort of vehicles running at low speed has been studied for several years by car manufacturers. Even if car passengers are exposed to a complex environment involving sight, hearing and touch, it is generally agreed that vibrations transmitted through the seat is a very sig-

nificant parameter in such situations. Previous laboratory experimentations revealed that vehicles were strongly discriminated even when subjects were submitted to seat vibrations only. The ranking of vehicles was also identical for different tested roads. All these observations raised the question of the existence of an identifiable signature of vehicles, independent of the road type. A perceptive experiment designed to evaluate the influence of such phenomenon was conducted. Subjects were submitted to a free sorting test. The categorisation task consisted in grouping vibrations stimuli recorded in 9 cars running on 3 different roads according to their similarity. The RMS level of stimuli was normalised so that its effect was not predominant during the experiment. Results showed that stimuli groups corresponded to the 3 tested roads, and were correlated to time envelopes of vibrations. Vehicles were not significantly discriminated.



**Session 4pNSb****Noise, Architectural Acoustics, and EURONOISE: Noise, Vibration and Acoustics for Medical and Research Facilities and Their Occupants I**

James West, Cochair

*Johns Hopkins University, Department of Electrical Engineering, 3400 North Charles Street, Baltimore, MD 21218, USA*

Jack B. Evans, Cochair

*JEAcoustics, 1705 West Koenig Lane, Austin, TX 78756, USA*

Marc Asselineau, Cochair

*Peutz & Associés, 10 rue des Messageries, Paris, F75010, France*

Erica Ryherd, Cochair

*Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405, USA***Invited Papers****2:00**

**4pNSb1. Toward a comprehensive hospital noise reduction research program.** Ilene Busch-Vishniac (McMaster University, 1280 Main Street West, Hamilton, ON L9G 4X6, Canada, provost@mcmaster.ca), James West (Johns Hopkins University, Department of Electrical Engineering, 3400 North Charles Street, Baltimore, MD 21218, USA, jimwest@jhu.edu)

Over the last few years there has been a growing interest in the control of hospital noise. This is prompted by a number of drivers: the recognition that hospital noise is a top complaint of patients, staff, and visitors; the move to a digital hospital and the impediment of speech recognition in very noisy environments; the implications of HIPAA and speech privacy on hospitals; and concerns about safety when communication is compromised by noise. Although there are now a number of studies of various issues associated with hospital noise, each views only a small piece of the very complicated problem. This presentation outlines a comprehensive noise research program designed to link acoustical measures with noise interventions and with staff and patient outcomes. Only through such an approach will it be possible to answer important questions such as which noise interventions are most effective and what hospital design strategies are most effective in terms of noise mitigation.

**2:20**

**4pNSb2. Describing the sound environment of the neurological intensive care unit and personnel response.** Erica Ryherd (Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405, USA, erica.ryherd@me.gatech.edu), Kerstin Persson Waye (Dept. of Environ. Medicine, The Sahlgrenska Acad. of Gothenburg Univ., Box 414, 405 30 Gothenburg, Sweden, kerstin.persson-waye@amm.gu.se)

The hospital soundscape is populated with a number of noise sources that may be detrimental to the occupational environment, including medical equipment, alarms, portable carts, activities, paging/communication, and ventilation systems. This paper describes a study in which sound measurements were made over five days and occupant evaluations were conducted in a neurological intensive care unit (ICU). Forty-seven nursing staff members completed questionnaires regarding general reactions to the ICU environment as well as perceived psychological and physiological reactions to the noise. Acoustical characteristics such as level distributions, restorative periods, and spectral content were explored in addition to overall equivalent, minimum, maximum, and peak sound pressure levels. Results showed the mean length of restorative periods (LAeq below 50 dB for more than 5 min) was 9 and 13 minutes for day and night, respectively. Sound measurements near the patients showed LAeq values of 53 - 58 dB, and dosimeters worn by the personnel revealed higher noise levels. Questionnaire results indicated that 91% of those surveyed felt that the noise negatively affected them in their daily work environment. They perceived the noise as contributing to stress symptoms such as irritation, fatigue, tension headaches, and difficulties concentrating. [Work supported by ASA and Swedish FAS].

**2:40**

**4pNSb3. Indoor environment and acoustic conditions in two Finnish hospital wards.** Annu Haapakangas (Finnish Institute of Occupational Health, Lemminkäisenkatu 14-18 B, 20520 Turku, Finland, annu.haapakangas@ttl.fi), Riikka Helenius (Finnish Institute of Occupational Health, Lemminkäisenkatu 14-18 B, 20520 Turku, Finland, riikka.helenius@ttl.fi), Valtteri O. Hongisto (Finnish Institute of Occupational Health, Lemminkäisenkatu 14-18 B, 20520 Turku, Finland, valtteri.hongisto@ttl.fi)

The aim of this study was to determine the acoustic conditions of two typical Finnish wards. The methods included long-term noise measurements, building acoustical measurements and questionnaires for patients (N=33) and nurses (N=27). The average sound pressure levels were within 49 and 58 dBA in the corridors, office and patient rooms. Personal noise exposure levels among nurses were within 56 and 70 dBA. The noise was mainly caused by people. Ventilation noise level was 28 dBA. Building acoustic measurements

showed reasonable agreement with national directions. Indoor environment was not a problem for most patients. Some disturbance from thermal conditions, dry air and noise was experienced. From daytime noise sources, other patients' snore and groans caused annoyance in some patients. Night-time sleep was disturbed most by anxiety, pain, noise, thermal conditions and an uncomfortable bed. Patients were, on average, quite satisfied with the overall room conditions. Nurses were more bothered by environmental factors than patients, giving highest annoyance ratings to thermal conditions, air quality and noise. Noises were experienced to cause some distraction when concentrating to work tasks. Sounds of phones ringing were experienced as particularly detrimental. Lack of privacy was the most obvious problem in both staff and patient evaluations.

3:00

**4pNSb4. Eliminating environmental stressors in hospitals: managing noise through different interventions.** Michael Phiri (The University of Sheffield, School of Architecture, Arts Tower, Western Bank, S10 2TN Sheffield, UK, m.phiri@sheffield.ac.uk)

There is growing body of knowledge of over 1,000 scientific studies which provide evidence that patients experience positive health outcomes in an environment that incorporates natural light, elements of nature, soothing colours, meaningful and varying stimuli, peaceful sounds, pleasant views and a sense of beauty. This paper reviews the research, its practical applications in order to enhance acoustics comfort and quality of care in healthcare settings. Healthcare planners can carry out small-scale, medium- and large-scale evidence-based design interventions. Small-scale interventions which minimise cost maximise impact on acoustic comfort can be implemented relatively quickly and easily e.g. provision of rubbish bins with a damping system to close lids slowly, dimming lights on the wards etc. Control measures include strategies to reduce noise levels in healthcare facilities with or without physical environmental alterations including specification of appropriate absorbent materials (e.g. acoustic ceiling tiles). Medium-scale interventions e.g. inpatient or nursing unit design involve consideration of the shape, geometry and other characteristics of a room in order to deal with background noise, speech intelligibility and other elements which promote acoustic comfort. Large-scale interventions involve large capital works notably entire hospital complexes, assemblies of departments/specialties but have evidence extrapolated from the small- and medium-scale interventions.

3:20

**4pNSb5. Improved perception of sound environments through room acoustic interventions.** Marc R. Janssen (Saint-Gobain Ecophon AB, Box 500, 26061 Hyllinge, Sweden, marc.janssen@ecophon.se)

Noise in hospitals is apparently among the top complaints of patients, visitors, and staff. An increasing amount of evidence shows the negative effects of sound within health care environments, affecting the quality of care in the end. Very few hospitals have actually been able to characterize hospital noise and to find effective noise control approaches. This paper will show international results of room acoustic interventions on the patients and staff perception of the sound environment. Next to that the suggestions will be presented on the use of relevant acoustical parameters to support the performance and well-being of staff and patients. One specific example will deal with a case study at the Thorax ICU at the Karolinska University Hospital in Stockholm. The purpose of the study has been to clarify how room acoustics in a patient ward affect the staff's perception of the noise situation in that ward. Another purpose has been to develop a better understanding of how a set of requirements for room acoustics should be formulated in order to obtain a supportive health care environment. It is possible to conclude that, by adding more absorption to the room, the noise situation has been positively influenced and perceived.

### 3:40-5:20 Posters

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

### Contributed Paper

5:20

**4pNSb6. Aural Connectivity: Enhancing sound environments in critical care settings for effective nurse auditory monitoring.** Selen Okcu (Georgia Institute of Technology, College of Architecture, Atlanta, GA 30332-0155, USA, okcuse@yahoo.com), Craig Zimring (Georgia Institute of Technology, College of Architecture, Atlanta, GA 30332-0155, USA, craig.zimring@coa.gatech.edu), Erica Ryherd (Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405, USA, erica.ryherd@me.gatech.edu)

In intensive care unit (ICU) settings, the sound environment is critically important to nurses accomplishing their tasks. In earlier studies by the authors, it was found that non-amplified environmental sounds such as patient

bodily sounds, patient threatening/unusual sounds, and help calls from patients and other caregivers are critically important auditory cues that nurses must listen for and respond to immediately. These sounds do not exist in isolation but matter as a pattern of aural connectivity that can support a nurse's critical monitoring abilities as she moves through her workplace. Aural connectivity is a network measure that reflects the overall pattern of where users can hear and respond to different key sounds within a setting. This paper describes the sound environments of two ICU hospital settings with similar patient acuity levels but differing layout designs. Preliminary results regarding the patterns of aural connectivity and the role that layout design might play in those patterns are discussed and potential implications for layout design are proposed.

4p THU. PM

5:40

**4pNSb7. Vibration Effects on Laboratory Mice during Building Construction.** Richard A. Carman (Wilson, Ihrig & Associates, 5776 Broadway, Oakland, CA 94618, USA, rcarman@wiai.com), Deborah A. Jue (Wilson, Ihrig & Associates, 5776 Broadway, Oakland, CA 94618, USA, djue@wiai.com), Gary M. Glickman (Wilson, Ihrig & Associates, 65 Broadway, Suite 401, New York, NY 10006, USA, gglickman@wiai.com)

Laboratory animals, in particular mice, are an integral part of medical and scientific research. Genetic research involving mice can be substantially affected by disruptions to the animals' environment. A new research facility is being built in close proximity to an existing one and the work will involve both demolition and new construction. Prior to construction, a study was conducted at the research facility to establish acceptable vibration levels in the vivaria areas. The study involved an experiment using an electrodynamic shaker to determine the effect of whole-body vibration on pregnant mice. The results of that study have been published. During construction, continuous vibration monitoring was conducted and the program and instrumentation used for monitoring are described. Results are presented, which include the researchers' data on the observed effects on the mice as well as the measured vibration levels during the construction.

6:00

**4pNSb8. "Vibration Kills" and other lessons from the trenches.** Hal Amick (Colin Gordon & Associates, P. O. Box 39, San Bruno, CA 94066, USA, hal.amick@colingordon.com), Michael L. Gendreau (Colin Gordon & Associates, P. O. Box 39, San Bruno, CA 94066, USA, michael.gendreau@colingordon.com)

In many areas of acoustics and vibration design, criteria are based upon parameters such as comfort, privacy, intelligibility, productivity, or machine precision. One can approach these criteria with a somewhat dispassionate attitude. However, when working with the biological research communities, one is periodically reminded by the researchers and medical practitioners that much of their work ultimately deals with life-and-death issues, either for patients for whom a drug or medical device is intended, or for organisms used in test protocols. The authors will share a collection of case studies in which these issues are illustrated. These include: Death of cells due to excessive vibration during electrophysiology. Vibration-induced nausea experienced by a surgeon using a surgical microscope for spine surgery. A life-saving medical technology resulting from a serendipitous discovery in a low-vibration lab environment. These case studies highlight the need for care in the specification, design, and construction of biological and biomedical research and healthcare facilities.

6:20

**4pNSb9. Don't forget the quench pipe when installing an MRI.** Carel Ostendorf (Cauberg-Huygen R.I. BV, Postbus 480, 6200 AL Maastricht, Netherlands, c.ostendorf@chri.nl)

An MRI is a useful medical device but it makes a lot of noise. A sound level of 90 dB(A) or more in the MRI room is not unusual. Placing an MRI in an existing hospital means that extra care has to be taken to prevent the noise from the MRI causing nuisance in adjacent rooms. In this paper the situation is discussed in which complaints appeared after the installation of the MRI. First sound measurements were done to see if the sound level in the office above the MRI fulfils the noise ratings. This way it would be clearer if the acoustic measures were doing their job. Whatever the outcome, the hospital wanted to put an end to the complaints. So, more sound measurements were done to establish the cause of the nuisance and to point out what extra measures have to be taken to solve the problem. It turned out that the quench pipe played an important role in this situation.

6:40

**4pNSb10. Generic noise criterion curves for sensitive equipment.** Michael L. Gendreau (Colin Gordon & Associates, P. O. Box 39, San Bruno, CA 94066, USA, michael.gendreau@colingordon.com)

Electron beam-based instruments are sensitive to the environment in which they operate. Adverse environments may limit their achievable resolution. Many equipment manufacturers provide specifications for the acceptable level of various environmental conditions, such as vibration, EMI, and acoustic noise. However, the quality of the specifications vary significantly, from well-defined to conjectural. Additionally, during the design of a facility, the specific instruments that will be used may not yet be known. Thus, it is useful to have "generic" criteria, intended to represent entire classes of instruments, to use in the design of facilities. Generic vibration criteria exist to aid in the design of laboratories, though there have been no such instrument-based generic criteria available for acoustic noise. The generic noise criteria that are currently used in lab design (NC, NR, dBA, etc.) were established to address the effect of noise on human beings. Using noise specifications for a significant number of instruments with varying resolving powers, correlation of resolution with environmental noise is demonstrated. Based on the data reviewed, generic noise criterion curves have been developed and presented for use in the design of facilities that contain noise-sensitive equipment. These are applicable when other well-defined and specific criteria are not available.

## Session 4pNSc

## Noise and EURONOISE: Potential to Reduce Tire/Road Noise I

J. Stuart Bolton, Cochair

Ray W. Herrick Labs., School of Mech. Eng., Purdue University, 140 S. Martin Jischke Drive, West Lafayette, IN 47907-2031,  
USA

Ernst-Ulrich Saemann, Cochair

Continental AG, Jaedekamp 30, Hannover, 30419, Germany

*Invited Papers*

2:00

**4pNSc1. The influence of tyre design on tyre/road noise - some fundamental thoughts.** Wolfgang Kropp (Chalmers University of Technology, Division of Applied Acoustics, SE-41296 Gothenburg, Sweden, wolfgang.kropp@chalmers.se), Patrick Sabiniarz (Chalmers University of Technology, Division of Applied Acoustics, SE-41296 Gothenburg, Sweden, patrick.sabiniarz@chalmers.se)

Tyre/road noise generation mechanisms are divided into two categories, tyre vibrations (due to time varying contact forces) and airflow related processes (e.g. air-pumping) in the contact between tyre and road. The paper only focuses on tyre vibrations. An existing model for the simulation of tyre/road interaction is used to investigate the influence of tyre design on the vibrational energy stored in the tyre structure during rolling. It can be shown that, although design is changed substantially, very little changes can be observed with respect to input power through the contact into the tyre structure. Changes in driving point mobilities of tyres are not directly related to changes in noise generation. Geometry changes leading to different contact geometry and in this way to different modal composition of the vibrational field as well as design changes leading to changes in wave speed are of higher importance. These changes have a strong influence on the radiation efficiency of tyres.

2:20

**4pNSc2. Development of low noise tyres in EC project SILENCE.** Ernst-Ulrich Saemann (Continental AG, Jaedekamp 30, 30419 Hannover, Germany, Ernst-Ulrich.Saemann@conti.de)

A lot of efforts were made in the last two decades to lower the tire/road noise. The tire industry has optimized the tread pattern as the main influence parameter so that nowadays the radiated sound pressure of a modern tire in the far field is very close to that of a blank tire with the same construction. Because the tread pattern is needed to achieve the required safety level further noise reduction has to be addressed mainly by tire construction. For many years tire manufacturers have been searching for a construction, which fulfils the targets of the automotive industry and generates less noise. The research was done not only in house but also with public projects. In the EC Project SILENCE a subproject has provided design solutions and hardware solutions for noise reduction, with respect to vehicle/tyre/road integration, under typical urban and suburban traffic conditions. This improvement is based on an increased understanding of noise generation and radiation mechanisms gained by the further development of experimental and simulation techniques. A series of low noise tyres (prototypes) has been developed and tested on a selection of appropriate low noise road surfaces.

2:40

**4pNSc3. The ranking of rolling noise from passenger car tyres - a comparison between measurements and modelling results.** Truls Berge (SINTEF ICT Dept. of Acoustics, O.S. Bragstads pl., NO-7465 Trondheim, Norway, truls.berge@sintef.no)

Tyres are type approved with regard to rolling noise on an ISO-test track, according to the EU-directive 2001/43/EC. The test track is basically a rather smooth road surface with maximum chipping size of 8 mm. However, most surfaces normally used on roads, especially in the northern European countries, are rougher surfaces, typically Stone Mastic Asphalt (SMA) with 11 to 16 mm stones. A project has been initiated to compare the noise levels of a selection of highly used after market summer tyres (in Norway). Noise measurements of 12 tyres have been performed on a selection of new and old SMA-road surfaces. The 3D texture of the same road surfaces has been measured with laser profile equipment. In addition, the point mobility and other design features of the tyres have been measured to be used as input data to the SPERoN tyre/road noise model. Then, comparison will be made between measurements and modelling results. Preliminary results from the noise measurements show a difference of 2.5-3 dB(A) on SMA-surfaces, between the tyres. The project is a co-operative between SINTEF (Norway), MüllerBBM (Germany), and Chalmers University (Sweden).



3:00

**4pNSc4. Sound Radiation Modes of a Tire on a Reflecting Surface.** J. Stuart Bolton (Ray W. Herrick Labs., School of Mech. Eng., Purdue University, 140 S. Martin Jischke Drive, West Lafayette, IN 47907-2031, USA, bolton@purdue.edu), Kiho Yum (Hyundai Motor Company, Chassis & Powertrain Engineering Team 3, Hyundai-Kia Motors Corporate R&D Division, Hwasung-Si, Gyeonggi-Do, 772-1, 445-706 Jangduk-Dong, Republic of Korea, kihoyum@hyundai-motor.com)

Wave number decomposition of a tire's radial vibration can be used to reveal the wave propagation characteristics of tires. In this paper, the relationship between the structural wave propagation characteristics of a tire excited at one point and its sound radiation is considered. The sound radiation resulting from structural vibration of a tire in contact with the ground was investigated by using boundary element analysis. In particular, the orthogonal radiation modes of a tire in the presence of a reflecting surface, along with their radiation efficiency characteristics, were calculated by applying an eigenvector analysis to the tire's radiation resistance matrix. The latter analysis made use of acoustic transfer vectors and a recovery surface appropriate for a pass-by noise test. The radiation mode results reveal that it is the vibration in the region close to the contact patch that primarily controls sound radiation. In particular, to reduce pass-by noise levels, it is necessary to mismatch the tire's structural ring mode and the radiation modes with high radiation efficiencies. It has also been found that the radiation from a tire is controlled by a relatively small number of radiation modes (although the number of contributing modes increases with frequency).

3:20

**4pNSc5. Analysis of truck tire rolling noise and structural vibrations.** Samuel Blanc (Goodyear, Technical Center, L - 7750 Colmar-Berg, Luxembourg, samuel\_blanc@goodyear.com), Jan Leyssens (Goodyear, Technical Center, L - 7750 Colmar-Berg, Luxembourg, jan.leyssens@goodyear.com)

Legislators are increasingly demanding towards the vehicle and tire industry for further reduction of tire/road noise emissions. With the need to develop quieter tires, noise has become one of the driving performance criteria in the design process, together with safety, wear and rolling resistance. Next generation truck tires will have to meet more severe legal requirements. In this context, noise has to be improved both at constant speed for the tire regulation and under acceleration for the vehicle legislation. Field test results of different tire tread designs under accelerated and constant speed conditions are given in this presentation. Noise mechanisms and effects of rolling conditions are discussed based on pass-by and near-field microphones. In addition, the dynamic vibration of a truck tire casing is analysed using both detailed FEA modelling and experimental results.

3:40-5:20 Posters

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

### *Invited Paper*

5:20

**4pNSc6. Simulation of tire tread block dynamics with respect to complex contact phenomena.** Gunnar Gäbel (Institute of Dynamics and Vibration Research, Leibniz University Hannover, Appelstr. 11, 30167 Hannover, Germany, gaebel@ids.uni-hannover.de), Patrick Moldenhauer (Institute of Dynamics and Vibration Research, Leibniz University Hannover, Appelstr. 11, 30167 Hannover, Germany, moldenhauer@ids.uni-hannover.de), Matthias Kröger (Institute of Dynamics and Vibration Research, Leibniz University Hannover, Appelstr. 11, 30167 Hannover, Germany, kroeger@ids.uni-hannover.de)

Vibrations of the tire structure are caused by the interaction between the rolling tire and the road surface. The tread block is the only tire component which is directly in contact with the cleft road surface texture and therefore of special interest. The contact due to the rough surface leads to complex contact phenomena e.g. friction characteristics depending on normal contact pressure and relative velocity, non-linear contact stiffness and wear effects changing the tread block geometry and in hence the local contact forces. These contact phenomena strongly influence the dynamical behaviour of the tread block and the whole tire. A modular model based on a modal reduction method will be presented to investigate the tread block dynamics under consideration of the local friction characteristic, the non-linear contact stiffness and wear which are also analyzed experimentally on a concrete road surface to identify the model parameters.

### *Contributed Papers*

5:40

**4pNSc7. Transmissibility of a deformed rotating tyre.** Ruud Van Doorn (Eindhoven University of Technology, Dept. of Mechanical Engineering, Dynamics & Control, P.O. Box 513, 5600MB Eindhoven, Netherlands, r.r.j.v.doorn@student.tue.nl), Ines Lopez (Eindhoven University of Technology, Dept. of Mechanical Engineering, Dynamics & Control, P.O. Box 513, 5600MB Eindhoven, Netherlands, i.lopez@tue.nl), René Van Der Steen (Eindhoven University of Technology, Dept. of Mechanical Engineering, Dynamics & Control, P.O. Box 513, 5600MB Eindhoven, Netherlands, r.v.d.steen@tue.nl), N.b. Roozen (Eindhoven University of Technology, Dept. of Mechanical Engineering, Dynamics & Control, P.O. Box 513, 5600MB Eindhoven, Netherlands, n.b.roozen@tue.nl), Henk

Nijmeijer (Eindhoven University of Technology, Dept. of Mechanical Engineering, Dynamics & Control, P.O. Box 513, 5600MB Eindhoven, Netherlands, h.nijmeijer@tue.nl)

The major source of environmental noise exposure is road traffic noise. Of all noise sources, tyre rolling noise is dominant for speeds above 30 km/h for passenger cars. Tyre rolling noise can be subdivided into interior and exterior noise. For the interior noise to which the passengers are exposed to, the tyre transmissibility is essential since it relates the contact forces with the axle forces. These axle forces are responsible for the structure borne interior noise. Here, a Finite Element tyre model, including a fully coupled air column, is used to examine the transmissibility in the frequency domain 0-300 Hz. It is shown that three aspects are essential in mod-

eling the axle forces resulting from tyre-road interaction: 1) the tyre deformation since it leads to a set of non-axisymmetric eigenmodes, 2) the relatively low-damped non-axisymmetric acoustic resonance, and 3) rotation. A methodology using substructuring techniques is presented to include rotational effects both in the case of an undeformed and deformed tyre. These effects of rotation on the transmissibility differ in the deformed and undeformed case: frequency loci veering occurs in the deformed case, while in the undeformed case rotation results in a pure split of the eigenfrequencies.

6:00

**4pNSc8. Simulation and analysis of tire road noise, finite element results and validation.** Maik Brinkmeier (Institut für Baumechanik und Numerische Mechanik, Appelstraße 9A, 30167 Hannover, Germany, brinkmeier@ibnm.uni-hannover.de), Udo Nackenhorst (Institut für Baumechanik und Numerische Mechanik, Appelstraße 9A, 30167 Hannover, Germany, nackenhorst@ibnm.uni-hannover.de), Jan Biermann (Institut für Modellierung und Berechnung, Denickestraße 17, 21073 Hamburg, Germany, biermann@tu-harburg.de), Otto Von Estorff (Institut für Modellierung und Berechnung, Denickestraße 17, 21073 Hamburg, Germany, estorff@tu-harburg.de)

This presentation shows the methods and results of the German research project "Silent Traffic". The main topics are the simulation of tire road noise as well as the validation of the methods and finite element models. The target is to understand the mechanisms of sound generation and to get suggestions to reduce the traffic noise resulting from the virtual system. The investigations are based on a physical modeling of the tire road system rather than on the processing of statistical data. The simulation procedure can be decomposed into four steps: The computation of non-linear stationary rolling, the eigenvalue analysis in the deformed state, the analysis of road surface textures, and the calculation of the noise radiation including a modal super-

position approach with an excitation by deterministic functions. Thereby, the numerical model enables for a detailed analysis of certain effects of the sound generation and radiation that contribute to the overall tire road noise. The simulation results are compared to measurements, both for structural dynamics and acoustics, to show the quality of the model and to indicate possible improvements for further development.

6:20

**4pNSc9. A contact solver suitable for tyre/road noise analysis.** Arjan Schutte (University of Twente, Dept. Mechanical Engineering, P.O. Box 217, 7500 AE Enschede, Netherlands, j.h.schutte@utwente.nl), Ysbrand H. Wijnant (University of Twente, Dept. Mechanical Engineering, P.O. Box 217, 7500 AE Enschede, Netherlands, Y.H.Wijnant@ctw.utwente.nl), André De Boer (University of Twente, Dept. Mechanical Engineering, P.O. Box 217, 7500 AE Enschede, Netherlands, a.deboer@utwente.nl)

Road traffic noise is a major environmental problem in modern society. The interaction between tyre and road surface, the major noise source, is non-linear and is best described in the time domain. The currently used contact models for acoustic analyses have problems with either accuracy or calculation speed. At the Structural Dynamics and Acoustics group of the University of Twente an alternative contact algorithm has been developed. The characteristic feature of this algorithm is that, while solving the set of equations, the contact condition, i.e. the condition stating that there is no overlap between the bodies, is satisfied exactly. Hence, there is no need for contact elements or contact parameters. The possibility to optimize and speed up the algorithm, using multigrid is the major advantage of the new approach. In this paper the contact algorithm is applied to a two-dimensional finite element model. Coulomb friction is taken into account. Some test simulations illustrate the ease of the algorithm. The model will be extended and non-linear material behaviour will be added. Multigrid and multilevel will be used to speed up the calculation. The goal is to compute the vibrations and radiated noise pattern of a profiled tyre rolling on a road.

### Invited Paper

6:40

**4pNSc10. The effect of porous road surfaces on radiation and propagation of tyre noise.** Bert Peeters (M+P - consulting engineers, PO Box 2094, 5260 CB Vught, Netherlands, BertPeeters@mp.nl), Ard Kuijpers (M+P - consulting engineers, PO Box 2094, 5260 CB Vught, Netherlands, ArdKuijpers@mp.nl)

It is well-known that porous road surfaces are very effective for the abatement of tyre/road noise. However, the physical principles behind the noise-reducing properties of these surfaces are not well understood and often even misinterpreted. Lack of understanding becomes a problem when developing a prediction model for tyre/road noise to be able to optimize the road surface for noise abatement. In the framework of the EU SILENCE project and the Dutch IPG program a model has been developed to predict the influence of road surface porosity on the radiation and propagation of tyre/road noise. First step in the model development was to gain physical insight from stationary and vehicle measurements with passenger car and truck tyres on both dense and porous surfaces. Next step was to qualitatively and quantitatively describe the observed physical phenomena in a mathematical model according to the KISS principle: avoid unnecessary complexity, both in input parameters and in the mathematical model. The end result is a mathematical model that describes the noise reduction potential for a standard tyre on various (porous) road surfaces, using a sound absorption spectrum as input. This model is available for a broad audience in the recently launched SPERoN acoustic optimization tool.

**Session 4pNSd****Noise and EURONOISE: Cultural Variations in Sound/Noise Assessment I (Poster Session)**

Catherine Guastavino, Cochair

*Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - School of Information Studies - McGill Univ.*

Danièle Dubois, Cochair

CNRS

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

**Contributed Papers**

**4pNSd1. Noise exposition in the daily life.** José Romero (Grup d'Acustica Arquitectònica, Ambiental i Industrial, E.T.S.I.I, Univ. Politècnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, romerof@fis.upv.es), Alicia Giménez (Grup d'Acustica Arquitectònica, Ambiental i Industrial, E.T.S.I.I, Univ. Politècnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, agimenez@fis.upv.es), Salvador Cerdá (Grup d'Acustica Arquitectònica, Ambiental i Industrial, E.T.S.I.I, Univ. Politècnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, salcerjo@mat.upv.es), Joaquin Navasquillo (Facultad de Ciencias, Univ. de Valencia, Blasco Ibañez, 15, 46010 Valencia, Spain, joaquin.navasquillo@uv.es), Radu Lacatis (Grup d'Acustica Arquitectònica, Ambiental i Industrial, E.T.S.I.I, Univ. Politècnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, rala1@doctor.upv.es), Arturo Barba Sevillano (Grup d'Acustica Arquitectònica, Ambiental i Industrial, E.T.S.I.I, Univ. Politècnica de Valencia, Camino de Vera, s/n, 46022 Valencia, Spain, arbarse@doctor.upv.es)

Nowadays, the man lives exposed in a world of sounds (pleasant - disagreeable, nonannoying - annoying) throughout the time, during the 24 hours a day. The man is exposed to the noise in his daily life, since he awakes, moves, works, entertains itself, rests, and to sleep. We have made a survey of levels of exhibition to the noise, having distinguished they are leisure, work or rest, in a amog young population of 20-35 years without hearing problems, and of both sexes. The results show us and agree with other authors, that the exhibition levels are highest in the activities chosen, like for example the leisure. In addition, the levels of exhibition, to the noise are related with the diversives activities of the population.

**4pNSd2. Noise(s) and sound(s): comparing various conceptualizations of acoustic phenomena across languages.** Danièle S. Dubois (CNRS, LCPE/LAM 11 rue de Lourmel, 75015 Paris, France, ddubois@ccr.jussieu.fr), Catherine Guastavino (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - School of Information Studies - McGill Univ., 3459 McTavish, Montreal, QC H3A1Y1, Canada, Catherine.Guastavino@mcgill.ca)

Musical listening focuses on perceptual attributes of the sound itself (e.g. pitch, loudness), whereas everyday listening focuses on events to gather relevant information about what happens in our environment (e.g. car approaching), that is, not about the sound itself but rather about noise as produced by sources and actions, and the effect of sound/noise on listeners. Previous linguistic analyses of discourses identified different conceptualizations of everyday/environmental noises and of musical sounds, as well as different conceptualizations for acousticians and non-acousticians. A more extensive psycholinguistic investigation was conducted to evaluate how different languages (not only consensual vocabularies) differently structure the semantic space of acoustic phenomenon. An open questionnaire was administered to expert listeners (acousticians) and naïve listeners from 5 different languages (French, Italian, Spanish, German and English) to collect linguistic resources available in each language and used in discourse to describe acoustic phenomena in their native languages. Presented in this paper are the first results of the comparative psycholinguistic analysis, in terms of linguistic devices and correlated conceptualizations, focusing on - the distinction between noise(s) and sound(s) across languages - the difference between scientific discourses and common sense discourses within and across languages.

## Session 4pNSf

## Noise and EURONOISE: Car Acoustics III (Poster Session)

Luc Mongeau, Cochair  
*McGill University*

Virginie Maillard, Cochair  
*RENAULT*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pNSf1. New vehicle noise emission values to update the French 'Guide du bruit'.** Sonia Doisy (Laboratoire des Ponts et Chaussées, 11, rue Jean Mentelin, BP 9, 67035 Strasbourg, France, sonia.doisy@equipement.gouv.fr), Joel Lelong (INRETS, 25 av. F. Mitterrand, case 24, 69675 Bron, France, lelong@inrets.fr), Jean-François Hamet (INRETS, 25 av. F. Mitterrand, case 24, 69675 Bron, France, hamet@inrets.fr)

French traffic noise prediction models are based on vehicle noise emission values defined by the "Guide du Bruit des Transports Terrestres", issued in 1980. A research programme was established in order to update these values and take into account developments in car and road technology. The methodology followed to output emission values has been developed by successive steps, as the processing performed on the collected data went along. After several attempts the shape adopted for the emission formulas considers the vehicle pass-by LAmax as the sum of two subsource contributions: the power unit contribution which varies with acceleration and the road gradient, and the tyre/road contribution which varies with speed and the road surface type. New emission values are now completely defined, and ready to be used in noise prediction models.

**4pNSf2. Acoustical Parameters of Automotive Interiors using Hybrid Fleeces basing on natural fibres.** Christian R. Koenig (University of Bremen, Badgasteiner Str. 1, FZB - Room 2140, 28359 Bremen, Germany, chkoenig@uni-bremen.de), Dieter H. Müller (University of Bremen, Badgasteiner Str. 1, FZB - Room 2140, 28359 Bremen, Germany, ml@biba.uni-bremen.de), K.-D. Thoben (University of Bremen, Badgasteiner Str. 1, FZB - Room 2140, 28359 Bremen, Germany, tho@biba.uni-bremen.de)

Hybrid fleeces are often used to produce composites and layered structures for car interiors. The fleeces consist of reinforcement fibres and polymeric fibres as matrix material. The utilization of natural fibres as a reinforcement for composite may be regarded as an emerging research area in polymer science. An important range of application can be found in the automotive industry. Despite of ecological gains like less environmental impact of the later product within the formation, usage and disposal period further technical and economical advantages result from this strategy. In addition to a reduction of the component's cost and net weight or an improvement of driving safety due to the crash behaviour of the composite material, natural fibre reinforced polymers offer increased recycling capabilities over conventional polymers used in that area. The presentation concentrates on the possibilities by using different materials, different ratios of thermoplastic and natural fibre material and different process parameters. Layers are especially investigated to demonstrate how the fleeces and the tech-

niques to bond the fleeces can influence the acoustical behaviour and mechanical properties. Based on the results of the measurements optimised, multi-layered sandwiches were developed and will be presented.

**4pNSf3. Statistical energy analysis limits for acoustic radiation car: an alternative approach.** Gérard Borello (InterAC, 10 impasse Borde-Basse, ZA. La Violette, 31240 L'Union, France, gerard.borello@interac.fr), Alex Borello (InterAC, 10 impasse Borde-Basse, ZA. La Violette, 31240 L'Union, France, info@interac.fr), Julien Primus (InterAC, 10 impasse Borde-Basse, ZA. La Violette, 31240 L'Union, France, info@interac.fr), Laurent Gagliardini (PSA Peugeot Citroën, Route de Gisy, 78943 Vélizy-Villacoublay Cedex, France, laurent.gagliardini@mpsa.com)

Due to a new pass-by noise regulation, Vehicle exterior noise will have to be reduced in the coming years. This may be achieved by optimizing underbody and underhood absorption and screening apertures. There is then a need for numerical techniques able to predict sound reduction related to acoustic absorption and transmission loss changes. Through a work supported by ADEME and headed by PSA, energy-based predictive techniques such as Analytical Statistical Energy Analysis (ASEA) and discretized Energy Flow Analysis (DEFA) were tested against the actual physical problem to be solved through a series of benchmarks. Both theories are compared across several simple acoustic problems. It is concluded that both methods do not fit to the initial acoustic optimization requirement due to their intrinsic assumptions that restrict their applicative range. More fitted numerical techniques are now investigated: among new candidates, the Virtual SEA (VSEA) technique that allows the creation of a numerical model of coupled acoustic cavities from the finite element global modes without the serious limitations of ASEA and a matrix approach based on Craigh-Bampton sub-structuring of the cavities.

**4pNSf4. A multiple regression model for predicting rattle noise subjective rating from in-car microphones measurements.** Benoît Gauduin (Genesis S.A., Bâtiment Gérard Mégie, Domaine du Petit Arbois - BP 69, 13545 Aix-en-Provence Cedex 4, France, benoit.gauduin@genesis.fr), Christophe Noel (Genesis S.A., Bâtiment Gérard Mégie, Domaine du Petit Arbois - BP 69, 13545 Aix-en-Provence Cedex 4, France, christophe.noel@free.fr), Jean-Louis Meillier (Renault, Centre Technique d'Aubevoys, Parc de Gaillon, 27940 Aubevoys, France, jean-louis.meillier@renault.com), Patrick Boussard (Genesis S.A., Bâtiment Gérard Mégie, Domaine du Petit Arbois - BP 69, 13545 Aix-en-Provence Cedex 4, France, patrick.boussard@genesis.fr)

In some situations when the road is deformed, the shock absorbers of vehicles may produce a specific sound, called rattle noise. It may be perceived by the driver and wrongly considered as a malfunction of the vehicle.



This sound is part of the global acoustic comfort of the vehicle and hence is studied by RENAULT. The approach presented here aims at predicting the rattle noise subjective rating given by a RENAULT expert on a scale from 0 to 10, by developing a model based on in-car binaural microphones measurements in the ears of the driver. First, a set of 11 metrics has been built, related to temporal aspects, spectral components and time-frequency information of the rattle noise recorded. The corpus is made of 19 different configurations of shock absorbers of a given car. The method used to select the most relevant metrics for the multiple regression model is presented. This selection is based on a statistical robustness estimation of the model. Hence, it appears that only 6 metrics are sufficient to build the model. Finally, the performance of the model is evaluated on 5 new configurations of shock absorbers.

**4pNSf5. Structure borne noise inside a coach.** Joanes Berasategi (Mondragon University, Loramendi 4, Apdo 23, 20500 Arrasate, Spain, joanes.berasategui@alumni.eps.mondragon.edu), Unai Galfarsoro (Mondragon University, Loramendi 4, Apdo 23, 20500 Arrasate, Spain, ugalfarsoro@eps.mondragon.edu), María Jesús Elejabarrieta (Mondragon University, Loramendi 4, Apdo 23, 20500 Arrasate, Spain, mjelejabarrieta@eps.mondragon.edu), Igor Insausti (Mondragon University, Loramendi 4, Apdo 23, 20500 Arrasate, Spain, ugalfarsoro@eps.mondragon.edu)

Nowadays the use of coaches as ground collective transport is generalizing in society and besides, its use is encouraged by all public institutions. Users of this type of vehicles request that they are comfortable, even more in long trips. Therefore, decreasing noise and vibrations inside the coach is an essential requirement to obtain a good quality of the vehicle and the satisfaction of the traveller. To increase the vibroacoustic comfort it is neces-

sary to know qualitatively and quantitatively the noise and vibration sources, as well as their transmission paths. Thus, in this work a procedure has been defined and applied to measure and analyse the vibration and acoustic behaviour of a coach in different operating modes: idling, three constant speeds, acceleration and deceleration. 32 acceleration and sound pressure signals have been acquired, corresponding to different interior and exterior points of the coach. The analysis of the acceleration autospectra has allowed to determine the contribution of elements like floor, glasses and lateral panels to the sound pressure perceived by the traveller in the passenger compartment in the different studied operating modes.

**4pNSf6. Experimental approach for reducing uncertainties associated with road vehicle noise according to ISO 362.** Louis-Ferdinand Pardo (UTAC, Autodrome de Linas-Montlhéry, BP 20212, 91311 Montlhéry, France, louis-ferdinand.pardo@utac.com), Thierry Ageron (UTAC, Autodrome de Linas-Montlhéry, BP 20212, 91311 Montlhéry, France, ageron@utac.com), Serge Ficheux (UTAC, Autodrome de Linas-Montlhéry, BP 20212, 91311 Montlhéry, France, ficheux@utac.com), Celine Berthou (UTAC, Autodrome de Linas-Montlhéry, BP 20212, 91311 Montlhéry, France, berthou@utac.com)

This paper proposes an approach for reduction uncertainties related to sound pressure levels measured in accordance with procedures given in ISO 362 for noise emitted by road vehicles under acceleration. This approach is based on an experimental assessing of several disturbing factors that lead to variation in the resulting level observed day to day and site to site for the same vehicle. The assessment of corrections to build the model is based on an independent analysis on each influent factor (test track, noise measuring device and temperature) taking into account of vehicle noise behaviour depending on power unit and tyres.

**Session 4pNSg****Noise, Architectural Acoustics, and EURONOISE: Noise, Vibration and Acoustics for Medical and Research Facilities and Their Occupants II (Poster Session)**

James West, Cochair  
*Johns Hopkins University*

Jack Evans, Cochair  
*JEAcoustics*

Marc Asselineau, Cochair  
*Peutz & Associés*

Erica Ryherd, Cochair  
*Georgia Institute of Technology*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

**Contributed Papers**

**4pNSg1. Noise control of laboratories: case studies.** Marc Asselineau (Peutz & Associés, 10 rue des Messageries, F75010 Paris, France, m.asselineau@peutz.fr)

Laboratories for research and production usually feature work areas complete with exhaust equipment as well as specific equipment (e.g. heaters, shakers, etc.). More to the point, such research and production areas are often linked to open plan offices where workers process their results, as well as partitioned offices where supervising engineers or researchers work. Such fitting out is supposed to help circulate information around the team and save valuable space. Unfortunately, it easily can be noisy and rather uncomfortable, especially when coming from older fully partitioned laboratories and offices. This paper aims to illustrate a few acoustical highlights of laboratories through a few cases studies, looking at such parameters as spatial sound level decay, background noise level and reverberation time, but also at such additional factors as general background and psychological aspects, and looking at tentative standards. It turns out that both the acoustical quality of the room and the space planning must be developed according to the users' needs.

**4pNSg2. Noise reduction in an operating room: A case study.** James West (Johns Hopkins University, Department of Electrical Engineering, 3400 North Charles Street, Baltimore, MD 21218, USA, jimwest@jhu.edu), Ilene Busch-Vishniac (McMaster University, 1280 Main Street West, Hamilton, ON L9G 4X6, Canada, provost@mcmaster.ca), Joseph King (Shamoon College of Engineering, Math. Department, Bialik/Basel Sts., 84100 Beer Sheva, Israel, joseph.a.king@usa.dupont.com), Natalia Levit (DuPont, 5401 Jefferson Davis Highway, Richmond, VA 23234, USA, natalia.v.levit@usa.dupont.com)

In our previous study reported in INTER-NOISE 2006 we found that operating rooms are among the most problematic areas in the healthcare industry. The maximum peak levels measured during various surgical procedures are extremely high - 100-120 dB, which can potentially lead to the hearing damage and interfere with the speech communication during surgery. Neurosurgery and orthopedic operating rooms are found to be among the noisiest overall. However, introduction of the acoustical materi-

als to the operating room is very difficult due to the strict infectious control requirements. We report here a case study on using sound absorptive panels protected by DuPont Tyvek®, a unique flash spun plexifilamentary film-fibril sheet, combining excellent barrier properties with distinctive porous structure to make it acoustically transparent in the voice frequency range.

**4pNSg3. Case Study of MRI Installation in Existing Hospitals.** James Perry (Cerami & Associates, 404 Fifth Avenue, New York, NY 10018, USA, jperry@ceramiassociates.com)

Medical facilities frequently seek to add function and capacity to their imaging facilities. The addition of new MRI units into existing hospital spaces is quite common, however many manufacturers impose and stringent low-frequency vibration limits which must be met before a unit can be delivered and installed. Vibration sources such as mechanical systems and external transit can degrade the achievable imaging resolution at levels undetectable by unaided humans. A case study is presented for one such project which required significant vibration controls to existing ventilation, water, and electrical systems to ensure acceptable vibration levels at an unlikely equipment installation.

**4pNSg4. Nanotechnology research facility- A vibration and noise control design case study.** Jack B. Evans (JEAcoustics, 1705 West Koenig Lane, Austin, TX 78756, USA, Evans@JEAcoustics.com), Chad N. Himmel (JEAcoustics, 1705 West Koenig Lane, Austin, TX 78756, USA, Himmel@JEAcoustics.com), Daniel J. Kupersztach (JEAcoustics, 1705 West Koenig Lane, Austin, TX 78756, USA, Kupersztach@JEAcoustics.com)

Vibration and noise control would be critical to the success of a proposed nanotechnology and molecular research facility. Roadway traffic, a nearby power generation plant and buildings in the vicinity were potential sources of ground borne vibration disturbances. Mechanical equipment, occupant installed support apparatus and occupant activities were potential sources of internally generated vibration disturbances. On-site ground borne vibration was measured for comparison with generic floor vibration criteria for sensitive installations. Results were analyzed relative to criteria for

potential disturbance of sensitive equipment, perception by occupants, audible radiated structure borne noise in acoustically sensitive spaces and resolution-degrading motion for scanning and for transmission electron microscopes and other nanotechnology clean room equipment. Design guidelines and structural vibration control concepts were recommended to the structural engineers, including de-tuning, damping and isolation methods. Recommendations were provided for mechanical noise control and vibration

isolation. Architectural noise control, sound isolation and room acoustics guidelines were provided for research, office and meeting spaces. This case study will discuss the desired vibration and noise control objectives and the design solutions that were implemented. Building photographs will be presented. Post-construction measurement results will be graphically compared with pre-construction conditions to demonstrate apparent degree of success in mitigating vibration.

THURSDAY AFTERNOON, 3 JULY 2008

P2-A, LEVEL 2, 3:40 TO 5:20 P.M.

### Session 4pNSh

## Noise, ASA Committee on Standards, and EURONOISE: Measurement of Occupational Noise Exposure II (Poster Session)

William Murphy, Cochair  
*National Institute for Occupational Safety and Health*

Beat Hohmann, Cochair  
*Suva, Physics Section*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pNSh1. New findings on noise exposure in the construction trades.**  
Reimer Paulsen (BGIA - Institut für Arbeitsschutz, Alte Heerstr. 111, 53757 Sankt Augustin, Germany, reimer.paulsen@dguv.de)

The degree to which employees on construction sites are exposed to noise often depends on a variety of individual activities. The calculation of the average noise exposure for different construction trades in this sector calls for an enhanced level of work analysis and measurement. Recommended here are personal measurements with simultaneous recording of the various activities and machines employed. With the subsequent assignment of average sound pressure levels and activities, it is then possible to draw conclusions about part exposures and mean exposure. At euronoise 2006 in Tampere, findings for a series of construction trades were presented. In the meantime, this series of measurements has been extended to include further trades. Furthermore subsequent measurements have been performed for individual trades, this was considered necessary because certain machines and working methods have changed in these fields in the last 20 years. The findings show that technical progress does not necessarily result in a reduction in noise exposure.

**4pNSh2. Who can guaranty compliance with the Exposure Limit Value of 87 dB(A)?** Fabien Krajcarz (Gamba Acoustique, 2 rue de la Découverte, BP 163, 31676 Labège Cedex, France, fabien.krajcarz@acoustique-gamba.fr)

The level of daily exposure (Lex8h) of workers has to be measured in accordance with French standard NFS 31084. The attenuation of personal noise protection has to be determined according to standard NF EN ISO 4869-2. In addition to the difficulties of applying these methods, one has to be aware that the actual effectiveness of the protection can deviate considerably from its theoretical effectiveness and, ultimately, it is the effective duration of wearing the protection that really determines the protection of workers. We have provided modest evidence, more than a survey in the

strictest sense of the term, of the conditions in which the protectors are worn (or not) at a large French industrial company. The results have shown that the wearing of hearing protectors is far from being routine, even when the noise levels are high. Thus, the question then arises as to the guarantee which the entity or person responsible for measuring bears for compliance with the Exposure Limit Value of 87 dB (A), when wearing the protection.

**4pNSh3. Measurement Method for Noise Exposure of Jobs of the Construction Sector.** Samuel Quintana (Universidad de Castilla-La Mancha, Campus Universitario, 16071 Cuenca, Spain, Samuel.Quintana@uclm.es), Marcos D. Fernandez (Universidad de Castilla-La Mancha, Campus Universitario, 16071 Cuenca, Spain, Marcos.Fernandez@uclm.es), Noelia Chavarria (Universidad de Castilla-La Mancha, Campus Universitario, 16071 Cuenca, Spain, noe\_chava@hotmail.com), Jose A. Ballesteros (Universidad de Castilla-La Mancha, Campus Universitario, 16071 Cuenca, Spain, Josea.Ballesteros@uclm.es), Isabel Gonzalez (Universidad de Castilla-La Mancha, Campus Universitario, 16071 Cuenca, Spain, Isabel.Gonzalez@uclm.es)

Noise is one of the physical contaminants with a high presence in the construction sector. Nowadays, several negative effects produced by the exposure to noise are known, mainly regarding hearing. Although there is evidence of the existence of many other effects, some of them are not characterized precisely yet. Due to the importance that these effects have on the worker's health and well-being, it is necessary to develop some mechanisms to study and suggest preventive solutions on these questions. In this work, it has been studied the most appropriate measurement method for taking, as precisely as possible, the noise levels that the workers of the construction sector are exposed to. Several measures have been taken and analyzed to determine the best indexes and parameters to characterize the noise in the construction. For which, the current European regulations regarding the noise exposure (Directive 2003/10 of the EU) have always been taken into account.

**4pNSH4. Impulsive occupational exposure for workers in drop-forge foundries.** Edward L. Zechmann (National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Mailstop C-27, Cincinnati, OH 45226-1998, USA, cri6@cdc.gov), Scott Brueck (National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Mailstop C-27, Cincinnati, OH 45226-1998, USA, zcd6@cdc.gov), William J. Murphy (National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Mailstop C-27, Cincinnati, OH 45226-1998, USA, wjm4@cdc.gov)

Occupational noise exposure limits for impulsive sounds in the United States are established by the Occupational Safety and Health Administration and recommended by the National Institute for Occupational Safety and

Health to permit no impulsive exposure in excess of 140 dB peak SPL. Peak pressure levels in excess of this limit can be found in industrial sectors such as construction, law enforcement and manufacturing. Taylor et al. [J. Acoust. Soc. Am 76:807-819, 1984] reported equivalent A-weighted levels,  $L_{Aeq}$ , of 108 dB for forge operators sampled in seven drop-forge foundries in the United Kingdom. In a recent study of noise exposure at two drop-forge manufacturing plants in the U.S., peak impulse levels were measured in a range of 117 to 154 dB peak SPL and 8-hour equivalent A-weighted levels,  $L_{Aeq8}$ , for forge operators between 95 and 116 dB and for press operators between 95 and 105 dB. Kurtosis, A, B C and D-durations were assessed for the impulsive and continuous noise samples. This paper will present the results of measurement and analysis of noise exposures typical for a drop-forge facility.

THURSDAY AFTERNOON, 3 JULY 2008

P2-A, LEVEL 2, 3:40 TO 5:20 P.M.

### Session 4pNSi

## Noise, Structural Acoustics and Vibration, Signal Processing in Acoustics, and EURONOISE: Airframe Noise Measurement, Prediction, and Control II (Poster Session)

Joe Posey, Cochair  
NASA

Denis Gely, Cochair  
ONERA

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pNSi1. Overview of revolutionary aircraft for quiet communities workshop.** Joe W. Posey (NASA, Langley Research Center, MS 461, Hampton, VA 23681, USA, joe.w.posey@nasa.gov)

At some time in the future, technical advances, environmental imperatives, societal expectations for mobility, and economic drivers will dictate that radically different aircraft will be built and flown. Therefore, aircraft designs will change even if low noise were not one of the environmental imperatives, and acousticians must be aware of the possible directions for aircraft design and the resulting opportunities and challenges for noise control. To address this need, the National Aeronautics and Space Administration sponsored a workshop entitled Revolutionary Aircraft for Quiet Communities in Hampton, VA, 24-26 July 2007. Twenty-six talks covered aircraft design, interior noise challenges, airframe noise, propulsion, and aircraft noise prediction. Revolutionary aircraft will employ dramatically improved materials, propulsion systems, and flow control technology to improve efficiency and enhance mobility. Five hours of discussion surfaced many concerns and recommendations, including the increasing need for acousticians to be involved on highly integrated teams throughout the vehicle design process. Also, other discipline experts participating in aircraft design need some education in acoustics to increase their sensitivity to noise control issues. The National Institute of Aerospace hosted the event, and the Council of European Aerospace Societies and the American Institute of Aeronautics and Astronautics were co-sponsors.

**4pNSi2. Adaptive closed-loop control of cavity flows.** Louis Cattafesta (University of Florida, 231 MAE-A, P.O. Box 116250, Gainesville, FL 32611, USA, cattafes@ufl.edu), Srinivasan Arunajatesan (Combustion Research and Flow Technology, Inc., 6210 Keller's Church Road, Pipersville, PA 18947, USA, ajs@craft-tech.com), Qi Song (University of Florida, 231 MAE-A, P.O. Box 116250, Gainesville, FL 32611, USA, song@ufl.edu), Cesar Moreno (University of Florida, 231 MAE-A, P.O. Box 116250, Gainesville, FL 32611, USA, cesargm@ufl.edu), Miguel Palavicini (University of Florida, 231 MAE-A, P.O. Box 116250, Gainesville, FL 32611, USA, mrp1124@ufl.edu)

Results from a combined experimental and computational study are presented on the development of an adaptive feedback controller for the suppression of cavity pressure loads. The experiments are performed in a variable-sized cavity in a high-speed wind tunnel, while the computations are performed using the CRAFT CFD flow solver. The adaptive control system incorporates recursive algorithms for system identification with disturbance rejection algorithms for feedback control. Results are presented using unsteady surface pressure sensors on the cavity walls and an array of zero-net mass-flux (ZNMF) actuators at the leading edge. The experimental data are used to compare with and validate the computations. These novel simulations form a virtual experiment testbed that is used to assess, for example, actuator type, placement, and requirements and also candidate identification and control algorithms.

4p THU. PM



## Session 4pNSj

## Noise and EURONOISE: Soundscape in the Heritage of Urban and Natural Areas II (Poster Session)

Bennett Brooks, Cochair  
*Brooks Acoustics Corp.*

Giovanni Brambilla, Cochair  
*CNR Institute of Acoustics*

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## Contributed Papers

**4pNSj1. Noise masking as a soundscaping measuring procedure.** Mohammed Boubezari (CAPS, Instituto Superior Técnico, TU Lisbon, Av. Rovisco Pais, P-1049-001 Lisbon, Portugal, boubezari@gmail.com), J. Luis Bento Coelho (CAPS, Instituto Superior Técnico, TU Lisbon, Av. Rovisco Pais, P-1049-001 Lisbon, Portugal, bcoelho@ist.utl.pt)

Loudness is the perceived intensity of a sound that could solve the problem of measuring a sound that is already filtered by the listening procedure. However, technically it is impossible to separate physically the sources of a noise into its components as well as human perception does. Loudness cannot be operative if a sound is merged with other noises. The principle of the proposed solution consists on gradually mixing a white noise during listening until the selected sound that one wants to measure is totally masked. The level of the masking white noise is controlled independently from the listener and measured with dB(A). The listener controls the limit of audibility of the selected sound, which is masked by the white noise. This paper describes the patent deposited around this method and shows the results obtained so far, and how this technique can be helpful in a visual translation of the sound space composition.

**4pNSj2. Adaptive Characterization of Near and Far field Elements in the Soundscape.** David R. Barclay (Marine Physical Lab, Scripps Institution of Oceanography, UCSD, 9500 Gilman Dr. M/C 0238, La Jolla, CA 92093-0238, USA, dbarclay@mpl.ucsd.edu)

Characterization of a soundscape through objective parameters relies on our understanding of psychoacoustics and ability to model the complex signal processing of the mind. Certain physical parameters such as loudness and timbre are easily retrieved from data while other descriptive parameters are more difficult to measure objectively. Several signal processing algorithms are presented here in the context of describing a soundscape in terms of keynote sounds (background noise) and sound signals (foreground sounds). Simulations and stereo field data recorded in San Diego are analyzed. Adaptive matched field processing is used in conjunction with conventional spectral analysis for the detection and categorization of near field events. These sounds are then removed to provide a more accurate description of keynote sounds. Spatial distribution of the soundscape is measured using conventional beamforming algorithms.

**4pNSj3. Internet and mobile technologies for a public role in noise surveying.** Charlie Mydlarz (The University of Salford, C.S.E., Salford University, The Crescent, M5 4WT Manchester, UK, c.mydlarz@salford.ac

.uk), Ian Drumm (The University of Salford, C.S.E., Salford University, The Crescent, M5 4WT Manchester, UK, i.drumm@salford.ac.uk), Trevor J. Cox (University of Salford, Acoustics Research Centre, Newton Building, M5 4WT Salford, UK, t.j.cox@salford.ac.uk)

The traditional method of noise surveying is to use trained professionals to go to a specific site to measure and assess noise levels using dedicated and expensive equipment. This project aims to enfranchise the public by providing them with the opportunity to play an active role in noise measurement and assessment, as well as how their soundscapes are shaped. With the implementation of mobile phone, PDA and PC applications, alongside web based collation techniques; we aim to empower the public in the gathering of context specific data on soundscapes. The methodology will provide a case study for the wider research community in developing public participation-based research activities of this kind. This will provide a better understanding of the public's relationship with their soundscape and how this relationship varies with location and demographic data. Inferences will be compared from the analysis of data sets generated with other soundscape research with a view towards validating the techniques and gaining new insights into the field.

**4pNSj4. From Descriptive to Predictive Soundscape Representation.** Mohammed Boubezari (CAPS, Instituto Superior Técnico, TU Lisbon, Av. Rovisco Pais, P-1049-001 Lisbon, Portugal, boubezari@gmail.com), J. Luis Bento Coelho (CAPS, Instituto Superior Técnico, TU Lisbon, Av. Rovisco Pais, P-1049-001 Lisbon, Portugal, bcoelho@ist.utl.pt)

Regarding the knowledge on psychoacoustics (Zwickers), on phenomenology of perception (Gibson, J. J.) and the results obtained at CAPS-IST (Boubezari-Bento Coelho), a descriptive and qualitative sound map was drawn and a predictive procedure was developed. The method shows that the introduction of human perception during the process of analysis and signal processing makes it possible to target measurements on one or more noise sources selected separately from their background noise. Contrary to conventional measurements, which yield overall values of LAeq, without distinguishing the sources, the method presented here allows a space description of a sound space by making each sound stand out from its context. The method allows the measurement of the range of a noisy source in a given place, the testing of the range or the masking of an urban device (fountain) or contribute to the decision and the design of a specific architectural project, for example. This paper describes the method (and its limits) and the results obtained and shows how a predictive qualitative sound map with low cost is now feasible.

## Session 4pNSk

## Noise and EURONOISE: Potential to Reduce Tire/Road Noise II (Poster Session)

J. Stuart Bolton, Cochair

*Ray W. Herrick Labs., School of Mech. Eng., Purdue University*

Ernst-Ulrich Saemann, Cochair

*Continental AG*

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## Contributed Papers

**4pNSk1. Frequency loci veering in deformed rotating tyres.** Ruud Van Doorn (Eindhoven University of Technology, Dept. of Mechanical Engineering, Dynamics & Control, P.O. Box 513, 5600MB Eindhoven, Netherlands, r.r.j.v.doorn@student.tue.nl), Ines Lopez (Eindhoven University of Technology, Dept. of Mechanical Engineering, Dynamics & Control, P.O. Box 513, 5600MB Eindhoven, Netherlands, i.lopez@tue.nl), René Van Der Steen (Eindhoven University of Technology, Dept. of Mechanical Engineering, Dynamics & Control, P.O. Box 513, 5600MB Eindhoven, Netherlands, r.v.d.steen@tue.nl), N.b. Roozen (Eindhoven University of Technology, Dept. of Mechanical Engineering, Dynamics & Control, P.O. Box 513, 5600MB Eindhoven, Netherlands, n.b.roozen@tue.nl), Henk Nijmeijer (Eindhoven University of Technology, Dept. of Mechanical Engineering, Dynamics & Control, P.O. Box 513, 5600MB Eindhoven, Netherlands, h.nijmeijer@tue.nl)

In previous work [1] a methodology to model tyre vibrations has been developed, which exploits a modal base determined in a standard FE package and includes rotational effects by a coordinate transformation. In the present paper, the effect of rotation on the eigenfrequencies of a deformed tyre is examined. It is well-known that rotation splits the eigenfrequencies of an undeformed tyre symmetrically around the eigenfrequencies of the non-rotating tyre, where the slope of the eigenfrequency-lines is determined by the circumferential wave number and tyre radius only. However, the eigenfrequency-lines of a deformed tyre demonstrate a fascinating mutual repulsion behaviour if the velocity is increased. This phenomenon is known as frequency loci veering and is induced by the a-periodicity resulting from the tyre deformation. Besides the effect of veering on the eigenfrequencies, the corresponding eigenmodes interact in the transition zones and finally interchange. The effects of veering are extremely intensified by the high

modal density of the tyre structure, which results in a decreasing eigenfrequency distribution when the rotational velocity increases. [1] I. Lopez et al, JSV 307, 481-494, (2007)

**4pNSk2. A model to evaluate the importance of tangential contact forces for tyre/road noise generation.** Patrick Sabiniarz (Chalmers University of Technology, Division of Applied Acoustics, SE-41296 Gothenburg, Sweden, patrick.sabiniarz@chalmers.se), Wolfgang Kropp (Chalmers University of Technology, Division of Applied Acoustics, SE-41296 Gothenburg, Sweden, wolfgang.kropp@chalmers.se)

The interaction between tyre and road is a complex non-linear process including radial and tangential contact forces between tyre and road surface. During recent years models have been developed which allow for predicting radial forces as function of surface and tyre properties. These models can be used to quantify the influence of radial forces on the noise generation. Whereas in many cases time varying radial forces are the main reason for the generation of tyre noise, in some cases other mechanisms seem to be dominant. This paper aims on understanding the influence of tangential contact on tyre/road noise generation. For this reason the model developed by the Chalmers Tyre Road Noise Group has been extended to also include tangential interaction. The model is based on the same concept as that used by McIntyre and Woodhouse for modelling string/bow contact of musical instruments. It computes the time varying normal and tangential contact forces for a tyre rolling over a rough surface at constant speed. A small selection of simulation results are presented aiming at showing the effect of different parameters, such as friction coefficient and normal load, on the occurrence of instability phenomena such as stick/slip vibrations.

## Session 4pNSI

## Noise and EURONOISE: Noise from Wind Power Projects II (Poster Session)

Eddie Duncan, Cochair  
*Resource Systems Group (RSG)*

Kerstin Persson Waye, Cochair  
*Dept. of Environ. Medicine, The Sahlgrenska Acad. of Gothenburg Univ.*

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## Contributed Papers

**4pNSI1. A rational approach for regulating windturbine noise.** Martin Van Den Berg (Ministry VROM, Po. Box 30945, IPC 635, 2500GX Den Haag, Netherlands, martin.vandenberg@minvrom.nl)

Windturbines have evolved from the cosy, wooden Dutch windmill type to large industrial generators of sustainable energy. Their noise production luckily did not evolve in step, but in some cases they can be a perhaps unsustainable nuisance for population living nearby. Presently the local and national authorities are trying to get a hold on the problem, often still using methods from the wooden machine age. The paper describes a modern approach for dealing with windturbine noise, using harmonized methods and recently acquired insights in the reaction of people to this particular noise. Tentative dose-effect relations in terms of Lden and Lnight enable comparisons with noise from transport.

**4pNSI2. Wind farm aural and visual impact in the Netherlands.** Frits Van Den Berg (University of Groningen - Science & Society Group, Nijenborgh 4, 9747AG Groningen, Netherlands, fvdberg@ggd.amsterdam.nl), Eja Pedersen (Occupational and Environmental Medicine, Göteborg University, PO Box 100, SE-405 30 Göteborg, Sweden, eja.pedersen@set.hh.se), Roel Bakker (Northern Centre for Healthcare Research, University Medical Centre Groningen, PO Box 30001, 9700 RB Groningen, Netherlands, R.H.Bakker@med.umcg.nl), Jelte Bouma (Northern Centre for Healthcare Research, University Medical Centre Groningen, PO Box 30001, 9700 RB Groningen, Netherlands, j.bouma@med.umcg.nl)

The WINDFARM perception project, carried out in 2006 in the Netherlands, aimed to explore the impact of wind farms on people living close to wind farms. The study sample was selected in three types of area (countryside, countryside with major road, built up area) by means of a Geographic Information System (GIS). Each selected address was within 2.5 km of a wind turbine of at least 500 kW electric power and a similar turbine

within 500 m of the first. Aural impact was calculated according to three different sound propagation models: the international ISO-9613 standard, the model legally required in the Netherlands, and a simplified model as in the New Zealand Standard NZS-6808. Visual impact was quantified in two ways: the vertical angle determined by the height of the apparently tallest turbine, and the solid angle determined by all turbines where each turbine was replaced by a vertical rectangle just enclosing the turbine. Immission sound levels from the wind farms at 1948 receiver locations varied from 21 to 54 dB(A), relative size from 0.01% to 30% of the total field of view. Results show that all impact measures are highly correlated with distance to the nearest wind turbine.

**4pNSI3. Dispersal of measured sound power levels for wind turbines.** René Gamba (Gamba Acoustique, 2 rue de la découverte, BP 163, 31676 Labège Cedex, France, rene.gamba@acoustique-gamba.fr), Sébastien Garrigues (Gamba Acoustique, 2 rue de la découverte, BP 163, 31676 Labège Cedex, France, sebastien.garrigues@acoustique-gamba.fr)

The standard IEC 61400-11 provides guidance in the measurement, analysis and reporting of acoustic emissions (sound power levels) from wind turbine generator systems. The application of this standard aims to provide accurate results that can be replicated by others. We did several measurement operations according to this standard on various wind farms fitted with many turbine manufacturers on different ground types. Important differences have been noticed with equal working conditions between the most and the less noisy wind turbine on a single farm. We will present these results compared to the manufacturers' guaranteed values and initiate explanations (like the difficulties to link the wind speed at 10m above ground with the wind speed received at hub height; or the influence of wind incidence on blades).

## Session 4pNSm

## Noise and EURONOISE: Vibration Perception II (Poster Session)

Patricia Davies, Cochair

*Ray W. Herrick Lab., School of Mechanical Engineering, Purdue Univ.*

Michael Griffin, Cochair

*Institute of Sound and Vibration Research, University of Southampton*

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## Contributed Papers

**4pNSm1. Hand-arm equal sensation curves for steering wheel translational and axial vibration.** Mickael Sauvage (PSA Peugeot Citroën, Centre Technique de Vélizy, Route de Gisy, 78943 Vélizy-Villacoublay, France, mickael.sauvage@mpsa.com), Elise Gressant (PSA Peugeot Citroën, Centre Technique de Vélizy, Route de Gisy, 78943 Vélizy-Villacoublay, France, elise.gressant@u-cergy.fr), Olivier Lescop (PSA Peugeot Citroën, Centre Technique de Vélizy, Route de Gisy, 78943 Vélizy-Villacoublay, France, olivier.lescop@mpsa.com), Vincent Roussarie (PSA Peugeot Citroën, Centre Technique de Vélizy, Route de Gisy, 78943 Vélizy-Villacoublay, France, vincent.roussarie@mpsa.com)

The aim of the study was to determine the hand-arm equal sensation curves for steering wheel translational and axial vibration. A sensory panel of 10 trained judges performed a two-step procedure. The test stimuli used were sinusoidal vibrations in the range from 4 to 60 Hz, with amplitude of  $0.2 \text{ m/s}^2$  rms. The first step was to determine perceived sensations for each frequency. Four families of vibration were defined (pumping movement, shaking sensation, trembling sensation and prickling sensation). The second step was a three-down-one-up test based on these families to determine the equal sensation level. Results showed that perceived intensity depends on vibration family, frequency and excitation direction (translational or axial).

**4pNSm2. A new instrument for the measurement of occupational vibration.** Laurent Faiget (01dB-Metravib, 200 Chemin des Ormeaux, F-69578 Limonest, France, laurent.faiget@01db-metravib.com), Fernand

Dupont (01dB-Metravib, 6 Avenue Louis Blériot, F-31570 Ste Foy D'Aigrefeuille, France, fernand.dupont@01db-metravib.com), Christine Aujard (01dB-Metravib, 200 Chemin des Ormeaux, F-69578 Limonest, France, christine.aujard@01db-metravib.com)

Directive 2002/44/CE relative to the risks arising from vibration has been transcribed into national laws of most Member States. This directive deals with the determination of limits and action values for the daily exposure to vibration. Two physiological domains are addressed: the "hand-arm" and the "whole-body" domains, the acceptable statutory values of which are specified in the text. The experimental protocol, as well as indicators relevant for assessment, is defined in Standards ISO 5349-2 and ISO 2631-1. The equivalent frequency-weighted acceleration shall be measured on the 3 axes x, y and z, the bandwidth of which is defined for each domain. 01dB-Metravib introduces a new portable instrument that perfectly meets the requirements of this statutory application. A blind metrological instrument connected to a triaxial accelerometer is installed on site. The instrument is remotely controlled by the operator using a wireless remote control of the Pocket PC type. This remote control is used to manage measurement configurations ("whole-body", hand-arm"), to start acquisitions (immediate, delayed mode) and to collect measured data for post-processing and archiving purposes. We will present metrological and operational advancements of this new instrument and describe a real case study dealing with the assessment of daily exposure to vibration.



**Session 4pNSn****Noise, Computational Acoustics, and EURONOISE: Source Identification in Radiation and Scattering II  
(Poster Session)**

J. Stuart Bolton, Cochair

*Ray W. Herrick Labs., School of Mech. Eng., Purdue University*

Peter Juhl, Cochair

*Institute of Sensors, Signals and Electrotechnics, University of Southern Denmark*

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***Contributed Papers***

**4pNSn1. Adaptation of the propagator for numerical acoustic holography of a wheel type object.** Thibault Le Bourdon (Institut Jean Le Rond d'Alembert, UMR CNRS 7190, UPMC, 2 Place de la Gare de Ceinture, 78210 Saint-Cyr l'Ecole, France, lbourdon@ccr.jussieu.fr), Vincent Martin (Institut Jean Le Rond d'Alembert, UMR CNRS 7190, UPMC, 2 Place de la Gare de Ceinture, 78210 Saint-Cyr l'Ecole, France, vmartin@ccr.jussieu.fr)

Holography procedure of a vibrating object can be geometrically interpreted resulting in a guarantee of its identified vibratory velocity, the quality of which depends on the pressure measurement at an array of microphones and on the propagation model. In case of a wheel with the panel body, a plane array parallel to the visible side is the only one possible. The velocity is then accessible on the side concerned and needs a propagation model with adequate acoustic conditions over the whole plane with the visible side (source plane). Having in mind such a configuration, a theoretical work in the 3D space has shown the surrounding influence on the reconstructed velocity on the visible side. It has appeared that vibrations other than the ones of the front side and the rear acoustic load can be described with an admittance on the source plane. We present here an exhaustive search for the adequate admittance concerned leading to a correct propagator and a correct velocity identified. The success achieved in the procedure may rely on the single velocity and single model liable to radiate a given pressure on a sufficiently large antenna.

**4pNSn2. Exact solutions to the acoustic source reconstruction problem.** Cédric Maury (Université de Technologie de Compiègne, Centre de Recherche Royallieu, BP20529, 60205 Compiègne, France, cedric.maury@utc.fr), Teresa Bravo (Université de Technologie de Compiègne, Centre de Recherche Royallieu, BP20529, 60205 Compiègne, France, teresa.bravo-maria@utc.fr)

In this study analytical solutions are derived for the singular radiation and velocity patterns of a baffled elastic beam, thus leading to closed-form expressions for the singular value expansion of a number of integral operators which map a boundary velocity onto the acoustic pressure distribution radiated in far-field or intermediate regions. Exact solutions to this problem involve prolate spheroidal wave functions which correspond to a set of independent distributions with finite spatial support and with maximal energy concentration in a given bandwidth in the wavenumber domain. A stable solution to the inverse source reconstruction problem is obtained by decomposing the unknown boundary velocity into a number of efficiently radiating singular velocity patterns which corresponds to the number of degrees of freedom of the radiated field. It is found that the degree of ill-posedness of the inverse problem is significantly reduced when considering a hemi-circular observation arc with respect to a linear array of sensors, by a factor scaling on the small angular aperture subtended by the observation line. Estimates are derived of the spatial resolution limits that can be achieved in the source reconstruction problem from the dimension of the efficiently radiating subspace.

**Session 4pNSo****Noise, ASA Committee on Standards, and EURONOISE: Prominent Discrete Tones II (Poster Session)**

Robert Hellweg, Cochair  
Wellesley, MA

Lothar Schmidt, Cochair  
Currenta GmbH & Co. OHG

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**Contributed Papers**

**4pNSo1. Tonalness perception of harmonic complex tones.** Sebastian Fingerhuth (Institute of Technical Acoustics, RWTH Aachen University, Neustrasse 50, 52066 Aachen, Germany, [sfi@akustik.rwth-aachen.de](mailto:sfi@akustik.rwth-aachen.de)), Etienne Parizet (Laboratoire Vibrations Acoustique, Insa Lyon, 25 bis, av. J. Capelle, 69621 Villeurbanne Cedex, France, [etienne.parizet@insa-lyon.fr](mailto:etienne.parizet@insa-lyon.fr))

Many everyday sounds have a more or less tonal characteristic. This normally means they have some peaks in the spectrum and generate one or more of the pitch sensations. In this work we present the results of a listening test which deals with the perception of tonalness of harmonic complex sounds. It consisted in presenting sounds via headphones to the 30 listeners who had to evaluate the tonalness using magnitude estimation with a reference sound. The sounds presented to the listeners varied in some parameters which modified the perception of tonalness. The parameters are: i) tone to noise level (3dB steps), ii) number of harmonics (1, 2, 4, 8 and 8 with  $f_0$  and  $f_1$  removed) and iii) spectral shape of the harmonics (constant, -5dB/oct and 5dB/oct). The loudness of the stimuli were equalized. Tonalness increases for higher tone to noise ratio. A change in the spectral shape is most important for a high number of harmonics. More harmonics also increase tonalness mostly for constant and increasing spectral shape. The results of the test were also compared with the tonalness calculation algorithm from Aures' and the german DIN45681 model. Both gave a correlation coefficient > 0.95.

**4pNSo2. Methods for automating prominent tone evaluation and for considering variations with time or other reference quantities.** Wade R. Bray (HEAD acoustics, Inc., 6964 Kensington Road, Brighton, MI 48116, USA, [wbray@headacoustics.com](mailto:wbray@headacoustics.com))

Information Technology acoustic protocols include identifying prominent tones according to likely subjective importance. Most existing methods for calculating tone-to-noise ratio (TNR) require a suspect tone to be selected by the analyst, who must also mark the width; both are potential sources of uncertainty and variability. One purpose of this paper is to present an automatic tone-detection and width-assessment methodology for more robust, less operator-intensive TNR calculation in accordance with ECMA-74. The paper will also present a process giving a complete spectral representation of the prominence ratio (a specific prominence ratio) by iterating the calculation at each frequency bin of the Fourier transform of the time-signal. The ECMA-74 tone-detection procedure for TNR is also applicable to the specific prominence ratio and is automatable, yielding tones-only prominence results without user intervention for any or all tones qualifying as prominent. The conventional average prominent tone evaluation over an operating interval can cause mis-assessment or no assessment of subjectively prominent tones which change frequency and/or level with time. We will therefore also discuss applying the described procedures as functions of time or other reference quantities, and suggest considering the subjective significance of time-domain effects such as modulation occurring from certain multiple prominent tone situations.

## Session 4pNSp

## Noise and EURONOISE: Tire-Road Noise from the Road Perspective II (Poster Session)

Paul Donovan, Cochair  
*Illingworth & Rodkin, Inc.*

Fabienne Anfosso-Ledee, Cochair  
*Laboratoire Central des Ponts et Chaussées*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pNSp1. Dense road surfaces with small aggregate size - tyre/road noise reduction after repaving.** Nils-Åke Nilsson (Acoustic Control AB, Tumstocksvägen 1, SE-187 66 Taeby, Sweden, na.nilsson@acoustic.se), Nils Ulmgren (NCC Roads Sweden AB, R&D Center, Bryggervägen 13, SE194 36 Upplands Väsby, Sweden, nils.ulmgren@ncc.se)

Open, porous road surfaces normally suffers from pore clogging particularly in inner city situations with low vehicle speed. After a relatively short time pore clogging could almost eliminate the noise reduction effect. A dense low noise road surface with smaller aggregate stone size could here offer substantial noise reduction due to reduced surface roughness, despite its low sound absorption factor. In this paper is presented data on the "repaving insertion loss" in dB(A) when exchanging a worn older SMA16 by a newly paved dense SMA8. Comparative results will also be shown for various types of newly paved SMA16 in comparison to the a newly paved SMA8. The technique for noise reduction mentioned above could be believed to have an "improved long term endurance" as compared to the same degree of noise reduction achieved by open graded technique particularly in city centres. For the Scandinavian countries where studded tyres are allowed, further reduction of stone size could be possible at acceptable wear rate if studded tyres are taxed or prohibited e.g. in environmental zones. This could lead to a further increased reduction of tyre/road noise in such zones. Grant from European Commission to Project Quiet City Transport (QCITY) is acknowledged.

**4pNSp2. Measurement of pavement absorption using ISO 13472-2.** Andrew F. Seybert (University of Kentucky, Dept. of Mech. Engineering, 151 RGAN Bldg., Lexington, KY 40506, USA, seybert@engr.uky.edu), Jun Han (University of Kentucky, Dept. of Mech. Engineering, 151 RGAN Bldg., Lexington, KY 40506, USA, jhan@engr.uky.edu)

This paper discusses some recent experience with ISO draft standard 13472-2 for the measurement of the absorption of pavements. The draft standard is meant to be used for the measurement of the sound absorption coefficient of "reflective" pavements, e.g., pavements of test tracks used for passby noise measurements of vehicles, tires, etc. An impedance tube has been constructed based on the requirements of the draft standard, and the device has been used in the laboratory (with pavement samples) and in situ to measure pavement absorption. The papers discusses these results and other important features of the draft standard including microphone calibration, parasitic tube absorption, and the construction features of the impedance tube and source.

**4pNSp3. A first step toward a close proximity noise map.** Moises Bueno (Laboratory of Acoustics Applied to Civil Engineering (LA2IC), Universidad de Castilla-La Mancha, Department of Applied Physics, 13071 Ciudad Real, Spain, moises.bueno@uclm.es), Urbano Viñuela (Laboratory of Acoustics Applied to Civil Engineering (LA2IC), Universidad de Castilla-La Mancha, Department of Applied Physics, 13071 Ciudad Real, Spain, urbano.vinuela@uclm.es), Fernando Terán (Laboratory of Acoustics Applied to Civil Engineering (LA2IC), Universidad de Castilla-La Mancha, Department of Applied Physics, 13071 Ciudad Real, Spain, fernando.teran@uclm.es), S. E. Paje (Laboratory of Acoustics Applied to Civil Engineering (LA2IC), Universidad de Castilla-La Mancha, Department of Applied Physics, 13071 Ciudad Real, Spain, santiago.exposito@uclm.es), Jeanne Luong (Mecanique d'ingenierie des systemes, Université Pierre et Marie Curie, 75252 Paris, France, jeanne.luong@etu.upmc.fr)

In recent years, environmental noise has become a serious issue for civil infrastructure and environmental administration due to public concern over the subject of noise pollution. The most significant deterioration of environmental acoustics conditions comes from road traffic transportation. The predominant noise source is the combination of the tire/pavement interaction and the propulsion systems of the vehicles. Generally, tire/pavement interaction is the principal source of noise for speeds above 40 km/h in the case of most modern cars. In this research, geo-referenced close proximity rolling noise is used for acoustical characterization of asphalt concrete surfaces in an urban environment. A close proximity noise map of streets with low speed limits is presented for a reference speed of 50 km/h. Different pavements and pavement conditions, common in urban streets, are analyzed: dense and semidense asphalt concrete, with Spanish denomination D-8 and S-12, respectively, and on the other hand, dense pavement at the end of its service life (D-8\*). Noise levels from dense surfaces (D-8) increase significantly over time, principally due to the appearance of surface defects such as cracks and ruts.

**4pNSp4. Road pavement classification based on noise emission characteristics.** Joel P. Paulo (ISEL, R. Conselheiro Emídio Navarro, 1, 1959-007 Lisbon, Portugal, jpaulo@deetc.isel.ipl.pt), J. Luis Bento Coelho (CAPS, Instituto Superior Técnico, TU Lisbon, Av. Rovisco Pais, P-1049-001 Lisbon, Portugal, bcoelho@ist.utl.pt)

The measurement procedure to evaluate the influence of road surface characteristics on vehicle and traffic noise is designated by Close-Proximity (CPX) method, as described in the ISO 11819-2 draft. This procedure consists on acquiring the vehicle rolling noise signal near the tires and close to the surface by means of at least two microphones, in a special arrangement for the determination of the Close-Proximity Sound Index (CPXI). Road

traffic noise is estimated by taking into account the absorption characteristics of road surface on the propagation of sound and the speed and type of vehicles. However, the particular characteristics of the different pavement types, which may influence the sound radiation, are not considered. The main goal of this research is to identify and classify different types of road pavements, for different stress conditions, using the CPX method. Such information can be used as a guideline for calibrating noise mapping models in order to achieve more realistic and accurate results. The classification of the different road surfaces consists on a supervised learning technique based on the Support Vector Machine, SVM, algorithms. Results based on error analysis are presented and discussed.

**4pNSp5. A single wheel trailer for tire/road noise measurements enabling both the CPX- and pass-by methods.** Martin Höjer (Acoustic Control AB, Tumstocksvägen 1, SE-187 66 Taebby, Sweden, martin.hojer@acoustic.se), Nils-Åke Nilsson (Acoustic Control AB, Tumstocksvägen 1, SE-187 66 Taebby, Sweden, na.nilsson@acoustic.se)

A single wheel trailer for tyre/road noise measurements has been developed. It has a towing beam, up to 8 m long, which ensures low background noise from support wheels and towing vehicle. Shielding hood is not necessary. The single wheel trailer can thus be used both for CPX-measurements with on-board carried microphones and pass-by measurements using stationary road-side microphones. The advantage of measuring in the free field with a minimum influence from the suspension attachments etc. is that the emission frequency spectra will be almost undisturbed. It can thus be more easily compared to the sound levels at road-side residents. Due to the long towing beam, tyre prototypes >10 dB quieter can be studied without disturbing background noise. The trailer has been specially designed to ensure stable vehicle dynamics, even though the normally loaded measurement tyre is rolling up to 8 meters behind the support trailer. Another interesting feature is the telescopic design of the towing beam, which enable

compaction of the trailer so that the measurement wheel is landed on the trailer support part, enabling convenient transportation to the measurement site.

**4pNSp6. A noise classification system for low-noise road surfacings - experiences and status.** Bent Andersen (Danish Road Institute/Road Directorate, Guldalderen 12, 2640 Hedehusene, Denmark, bea@vd.dk), Jørgen Kragh (Danish Road Institute/Road Directorate, Guldalderen 12, 2640 Hedehusene, Denmark, kragh@vd.dk), Sigurd N. Thomsen (Danish Road Institute/Road Directorate, Guldalderen 12, 2640 Hedehusene, Denmark, snt@vd.dk)

As a result of an increasing pressure from the population as well as from road administrations a noise classification system for road surfacings was put into operation in the autumn 2006. The so-called SRS-system (a Danish abbreviation for noise reducing wearing courses) is provisional and non-committal for a couple of years until sufficient experience has been collected. It is a comprehensive system including: 'Guidelines on the use of low-noise asphalt surfacings for noise abatement. 'Method for declaration and documentation of noise reduction based on the CPX (close proximity)-method. 'Paradigm for use in preparation of tendering documents and contracts. The measurement method is based upon the ISO draft standard describing the CPX-method, but includes further specification. Reference values for dense asphalt concrete are defined, and 2 dB wide noise reduction classes A, B, and C are defined at 50 and 80 km/h. The method also specifies an inter-calibration procedure (including definition of reference road sections) to be applied for different CPX measurement devices. Each year a specific calibration constant is issued for each device. The paper will also discuss the recent developments in the selection of future reference tyres for CPX measurements. Finally, a status of the classified surfacings will be presented.

THURSDAY AFTERNOON, 3 JULY 2008

P2-A, LEVEL 2, 3:40 TO 5:20 P.M.

## Session 4pNSq

### Noise and EURONOISE: Fan Noise and Low-Mach Number Rotating Blade Noise II (Poster Session)

Scott Morris, Cochair  
*Notre Dame*

Michel Roger, Cochair  
*Ecole Centrale de Lyon*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pNSq1. Prediction of blade trailing-edge noise of an axial flow fan.** Alain Guedel (CETIAT, BP 2042, 69603 Villeurbanne Cedex, France, alain.guedel@cetiat.fr), Arthur Finez (Ecole Centrale de Lyon, 36 Avenue Guy de Collongue, Centre Acoustique, 69134 Ecully, France, arthur.finez@gmail.com), Gérald Perrin (CETIAT, BP 2042, 69603 Villeurbanne Cedex, France, gerald.perrin@cetiat.fr), Michel Roger (Ecole Centrale de Lyon, 36 Avenue Guy de Collongue, Centre Acoustique, 69134 Ecully, France, michel.roger@ec-lyon.fr)

Broadband noise is a major part of the noise radiated by low-speed fans such as industrial and domestic fans. One of the main mechanisms of

broadband noise is blade trailing-edge noise, which is due to the convection of the turbulent flow of the boundary layer past the blade trailing edge. The objective of this work is to predict trailing-edge noise of an axial flow-fan with an analytical model adapted from Amiet's formulation. The input data of the model are the spectra and correlation length scales of the wall pressure fluctuations on the blade suction side close to the trailing edge. These data are measured with small pressure transducers flush mounted on the blade suction side. A first study was performed on a 800-mm axial fan without shroud. The comparison between the predicted and measured far-field sound pressure spectra proved quite good, which validated the model in this impeller configuration. The next stage in progress is to insert a shroud on the



same impeller and validate the noise prediction in this case for different fan operating points. The importance of additional broadband noise sources, such as blade tip noise, could also be assessed on this fan configuration.

**4pNSq2. Fan Blade Trailing-Edge Noise Prediction Using RANS Simulations.** Yannick Rozenberg (ONERA, BP 72 - 29 avenue de la Division Leclerc, 92322 Chatillon Cedex, France, yannick.rozenberg@onera.fr), Michel Roger (Ecole Centrale de Lyon, 36 Avenue Guy de Collongue, Centre Acoustique, 69134 Ecully, France, michel.roger@ec-lyon.fr), Stéphane Moreau (Valeo Thermal Systems, rue Louis Normand, 8, 78321 La Verrière, France, stephane.moreau@valeo.com)

An analytical model based on Paterson & Amiet's work dealing with the trailing-edge noise of a blade has been previously validated thanks to a dedicated experiment on a low speed axial fan. Wall-pressure spectra near the trailing-edge of the blade and at different radii are needed for an accurate prediction. Only experiments and LES simulations are able to provide them. In an industrial context, both methods can not be applied since they are too expensive and time-consuming. To overcome this difficulty, RANS simulations are combined with semi-empirical wall-pressure spectra to obtain the needed input data. The effect of the mean-pressure gradient is taken into account. The model is applied first to the noise radiated by an airfoil placed in the open-jet of an anechoic wind tunnel, then to an automotive cooling fan and finally to an aircraft engine fan. RANS simulations are post-processed to run the analytical model with appropriate input data. The noise predictions are then compared with experimental results.

**4pNSq3. Experimental investigation of wind turbine noise.** Maud Leroux (Laboratoire d'Etudes Aérodynamiques (LEA), Université de Poitiers - ENSMA - CNRS, Bâtiment K, 40 Avenue du Recteur Pineau, 86022 Poitiers, France, maud.leroux@lea.univ-poitiers.fr), Yves Gervais (Laboratoire d'Etudes Aérodynamiques (LEA), Université de Poitiers - ENSMA - CNRS, Bâtiment K, 40 Avenue du Recteur Pineau, 86022 Poitiers, France, yves.gervais@lea.univ-poitiers.fr), Jacques Borée (Laboratoire d'Etudes Aérodynamiques (LEA), Université de Poitiers - ENSMA - CNRS, Bâtiment K, 40 Avenue du Recteur Pineau, 86022 Poitiers, France, jacques.boree@lea.ensma.fr), Arnaud Ménoret (Signal Développement, 12 Boulevard Chasseigne, 86000 Poitiers, France, a.menoret@signal-developpement.com)

Broadband noise is nowadays the major contribution to the total spectrum of noise generated by wind turbines. The mechanisms that generate airfoil self-noise have been studied through years and many authors agree that dominant noise comes from inflow turbulence, and interaction between tur-

bulent boundary layer and trailing edge of the airfoil. This study presents results from combined experimental techniques in order to better identify and predict noise issuing from a NACA 0012. Noise from trailing edge is most investigated in a 2D configuration in an anechoic wind tunnel, using microphone array (far field measurements), wall pressure fluctuations, hot wire. The data base turns out to be useful in order to improve extensions of Amiet and Brooks models. Some aspects of noise mechanisms and their characteristics are better identified and refined when wind tunnel results are compared to measurements of noise on full scale wind turbines. Measurements in situ achieved with a microphone array of 30 meters wide are used to provide complementary informations on 3D rotating source in terms of localisation and specific directivity.

**4pNSq4. Measurement of wall pressure fluctuations for noise prediction in axial flow fans.** Jérémy Hurault (Lab. d'Energétique et de Mécanique des Fluides Interne, Arts et Métiers ParisTech, 151 Boulevard de l'Hôpital, 75013 Paris, France, jeremy.hurault@paris.ensam.fr), Smaïne Kouidri (LIMSI-CNRS, BP 133, 91403 Orsay Cedex, France, smaine.kouidri@limsi.fr), Farid Bakir (Lab. d'Energétique et de Mécanique des Fluides Interne, Arts et Métiers ParisTech, 151 Boulevard de l'Hôpital, 75013 Paris, France, farid.bakir@paris.ensam.fr), Robert Rey (Lab. d'Energétique et de Mécanique des Fluides Interne, Arts et Métiers ParisTech, 151 Boulevard de l'Hôpital, 75013 Paris, France, robert.rey@paris.ensam.fr)

The unsteady pressure field on fan blades is an important investigation topic. Both numerical simulation and experimental techniques are used in order to achieve this purpose. However neither has given yet entire satisfaction. The CFD tools using the resolution of the averaged Navier Stokes equations do not really give the unsteady aerodynamic characteristics of the flow needed for an accurate noise prediction. In addition, tools using large eddy simulation are still expensive for industrial users in the case of a complex geometry such as turbomachinery. When a lifting surfaces goes through turbulence, pressure fluctuations occur on their surfaces that can radiate noise. To calculate these fluctuations and thus the noise requires a theoretical model of the unsteady aerodynamics. The validation and development of these models require data and understanding from experiments. Unsteady surface pressure measurements were carried out on one fan blade with an array of pressure transducers with high sensitivity. The fan studied is a low pressure and low Mach number axial flow fan. Investigations of unsteady surface pressure are carried out in different configuration, spanwise, chordwise, pressure side and suction side. Data are gathered through a slip ring by an analyser. The unsteady wall pressure spectra is used as an input for trailing edge noise analytical prediction model.

## Session 4pNSr

## Noise and EURONOISE: Cultural Variations in Sound/Noise Assessment II

Catherine Guastavino, Cochair

*Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - School of Information Studies - McGill Univ.,  
3459 McTavish, Montreal, QC H3A1Y1, Canada*

Danièle S. Dubois, Cochair

*CNRS, LCPE/LAM 11 rue de Lourmel, Paris, 75015, France*

## Invited Papers

5:20

**4pNSr1. Acoustical Variety of Soundscapes - Comparison of Soundscapes.** Klaus Genuit (HEAD acoustics GmbH, Ebertstrasse 30a, 52134 Herzogenrath, Germany, klaus.genuit@head-acoustics.de), André Fiebig (HEAD acoustics GmbH, Ebertstrasse 30a, 52134 Herzogenrath, Germany, andre.fiebig@head-acoustics.de)

Noise of urban places varies not only with regard to its physical parameters, but also it is perceived and assessed differently because of cultural, sociological, historical and economic influences. The physical description of a soundscape does not cover the complexity of human perception in a specific environmental setting. Therefore, the task of soundscape researchers has to consider more aspects than the measurement and interpretation of the acoustical differences caused by location-specific noise sources. Moreover, semantic and cognitive aspects relating to culture, tradition and economy must be extensively analyzed. Cultural and sociological conditions influence the people's evaluations of their surroundings. A deeper understanding is necessary to adequately analyze soundscapes, especially where specific noise phenomena - e.g. temporal and spectral effects - are of more importance to the listeners' well-being than an averaged SPL value. The presented paper compares and analyses environmental sounds of different cities scattered all over the world by means of conventional and advanced acoustical analyses. The results could provide reliable data for further investigations covering sociological and cultural issues.

5:40

**4pNSr2. Naïve and expert listeners use different strategies to categorize everyday sounds.** Guillaume Lemaitre (IRCAM - UMR CNRS 9912, Equipe Perception et Design Sonores, 1, place Igor Stravinsky, 75004 Paris, France, lemaître@ircam.fr), Olivier Houix (IRCAM - UMR CNRS 9912, Equipe Perception et Design Sonores, 1, place Igor Stravinsky, 75004 Paris, France, olivier.houix@ircam.fr), Nicolas Misdariis (IRCAM - UMR CNRS 9912, Equipe Perception et Design Sonores, 1, place Igor Stravinsky, 75004 Paris, France, misdariis@ircam.fr), Patrick Susini (IRCAM - UMR CNRS 9912, Equipe Perception et Design Sonores, 1, place Igor Stravinsky, 75004 Paris, France, susini@ircam.fr)

We report an experiment investigating the influence of the expertise of listeners on the strategy used to categorize sounds. A set of sixty kitchen sounds was selected, based on their causal uncertainty (ranging from very well identified to ambiguous). Thirty listeners were selected on the basis of their expertise in sound and music: fifteen "experts" and fifteen "naïves". First, they had to group together the sounds. Second, they had to describe the properties shared by the sounds in each category. Finally, they were provided with a description of different strategies of classification previously identified (acoustical, causal or semantic similarity), and required to indicate, for each category, which one they had used. The results show a strong influence of the expertise of the participants: while naive listeners made categories mainly on the basis of the events that they identified as having caused the sounds, experts made mainly categories of sounds on the basis of the perceived acoustical properties (timbre, time patterns, etc.). This result is coherent with the available literature demonstrating the coexistence of different strategies of listening, and links these strategies to the skills of the listeners. [This work is founded by the FP6 NEST Pathfinder European project CLOSED]

## Contributed Paper

6:00

**4pNSr3. Cultural variations and constants in emotional reactions to sounds.** Daniel Västfjäll (Chalmers University of Technology, Division of Applied Acoustics - Chalmers Room Acoustics Group, Sven Hultins gata 8a, 41296 Gothenburg, Sweden, daniel.vastfjall@psy.gu.se)

In this talk I will argue that emotional reactions to sounds can be both constant and vary between different cultures. A model of sound perception that was developed by Chalmers room acoustics group, the Emotional

Reaction Model (ERM) will be reviewed. The ERM predicts that emotional reactions to auditory events can be both elicited by 1) certain form features (eg. a steep rise time, a loud sound, a rough sound) and 2) content features (eg. the qualitative experience of the sound source; the loud, rough sound is a tiger roaring). The influence of form features are expected to be rather constant across people and cultures, while content features are expected to vary more between people. Empirical evidence supporting these predictions will be reviewed and the implications for our understanding sound perception will be discussed.

6:20

**4pNSr4. When exposed to sounds, would perceived loudness not be affected by social context?** Pieter Jan Stallen (Leiden University, PX Box 9555, 2300 RB Leiden, Netherlands, [stallen@fsw.leidenuniv.nl](mailto:stallen@fsw.leidenuniv.nl))

After decades of predominantly correlation studies of non-auditory factors and environmental noise annoyance, Maris, Stallen, Vermunt and Steensma (2007a,2007b) have demonstrated experimentally that a negative and positive relationship between producer and receiver of environmental sounds can cause the receiver to be more and less annoyed by the sound, respectively. This finding raises the question whether the context of environmental sound, and social context in particular, could already be determining responses to environmental sounds at earlier stages of auditory processing. This paper will present answers regarding perceived loudness, which is generally considered to be a more immediate (and less evaluative) response to sound exposure than the feeling of annoyance. From a statistics point of view, there is room for early influences as isophones which are based upon equal loudness judgments seem to hide no less, if not more, variation than the mean annoyance score at various sound pressure levels (cf. Berglund and Preis, 1997). It will be argued that seemingly conflicting empirical findings could be reconciled by postulating different attentional mechanisms as they vary with task characteristics and demands. Ideas will be presented by which the presumed theoretical model could be tested experimentally.

6:40

**4pNSr5. Comparison of Japanese and English language descriptions of piano performances captured using popular multichannel microphone arrays.** William L. Martens (McGill University, Schulich School of Music, 555 Sherbrooke Street West, Montreal, QC H3A 1E3, Canada, [wlm@music.mcgill.ca](mailto:wlm@music.mcgill.ca)), Sungyoung Kim (Yamaha, 203 Matsunokijima, 438-0192 Iwata, Japan, [sungyoung\\_kim@gmx.yamaha.com](mailto:sungyoung_kim@gmx.yamaha.com)), Atsushi Marui (Tokyo University of the Arts, 1-25-1 Senju, Adachi, 120-0034 Tokyo, Japan, [marui@ms.geidai.ac.jp](mailto:marui@ms.geidai.ac.jp))

In a cross-cultural comparison of musical sound evaluations, the way in which bipolar adjective pairs are used by native speakers of Japanese and English language was studied via a subjective rating task. These ratings were collected in response to eight solo piano performances that had been captured using four popular multichannel microphone arrays, reproduced via a standard 5-channel loudspeaker array, re-recorded binaurally, and finally presented via headphones. This allowed nearly identical stimuli to be presented to all listeners, without any modulation of the loudspeaker signals via listener head movements. Average ratings were compared to acoustical measures made on the 32 binaural stimuli, and to salient perceptual dimensions that previously had been derived from pairwise dissimilarity ratings between the stimuli. Results showed close agreement in how the selected terms were used by native speakers of Japanese and English language in the context of this study.

THURSDAY AFTERNOON, 3 JULY 2008

P3-A, LEVEL 3, 3:40 TO 5:20 P.M.

## Session 4pPAa

### Physical Acoustics: Ultrasonics: Material Characterization II (Poster Session)

Jim A. Tencate, Cochair  
*Los Alamos National Laboratory*

Koen Van Den Abeele, Cochair  
*K. U. Leuven, Campus Kortrijk*

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### Contributed Papers

**4pPAa1. Structure and ultrasonic properties of vanadium tellurite glasses containing copper oxide.** Nadia Abd El-Aal (National Institute of Standards, 136 Tersa St, El-Haram, 12211 Giza, Egypt, [n\\_nadia\\_99@yahoo.com](mailto:n_nadia_99@yahoo.com)), Hisham Afifi (National Institute of Standards, 136 Tersa St, El-Haram, 12211 Giza, Egypt, [hmafifi@hotmail.com](mailto:hmafifi@hotmail.com))

The elastic properties of vanadium tellurite ( $65\text{TeO}_2\text{-(35-x) V}_2\text{O}_5\text{-xCuO}$ ) glasses with different compositions of copper ( $x = 7.5$  to  $17.5$  mol% in steps of  $2.5$  mol %) have been studied at room temperature (300K). The ultrasonic velocity measurements have been made using a transducer having resonating frequency of 4 MHz (both longitudinal and shear). The density,

molar volume, and ultrasonic velocities show an interesting observations, which are used to explore the structural changes in the network. Elastic moduli, Poisson ratio, crosslink density, microhardness, and Debye temperature of the glasses have been determined using the experimental data. The composition dependence of the elastic properties explores useful information about the physical properties of the vanadium tellurite glasses doped with copper. Quantitative analysis has been carried out in order to obtain more informations about the structure of the glass under the study, based on bond compression model, and ring deformation model, i.e., the cation-anion bond of each oxide. The observed results through ultrasonic non-destructive evaluation investigate the structural changes and mechanical properties of the glass.

**4pPAa2. Elastic characterization of ceramic balls using resonant ultrasound spectroscopy of spheroidal modes.** François Deneuville (Institut d'Electronique de Microélectronique et de Nanotechnologie, Département Opto-Acousto-Electronique (UMR CNRS 8520), Université de Valenciennes et du Hainaut-Cambrésis, Le Mont-Houy, 59313 Valenciennes cedex 9, France, francois.deneuville@gmail.com), Marc Duquennoy (Institut d'Electronique de Microélectronique et de Nanotechnologie, Département Opto-Acousto-Electronique (UMR CNRS 8520), Université de Valenciennes et du Hainaut-Cambrésis, Le Mont-Houy, 59313 Valenciennes cedex 9, France, Marc.Duquennoy@univ-valenciennes.fr), Mohammadi Ouafthouh (Institut d'Electronique de Microélectronique et de Nanotechnologie, Département Opto-Acousto-Electronique (UMR CNRS 8520), Université de Valenciennes et du Hainaut-Cambrésis, Le Mont-Houy, 59313 Valenciennes cedex 9, France, mohammadi.ouafthouh@univ-valenciennes.fr), Frédéric Jenot (Institut d'Electronique de Microélectronique et de Nanotechnologie, Département Opto-Acousto-Electronique (UMR CNRS 8520), Université de Valenciennes et du Hainaut-Cambrésis, Le Mont-Houy, 59313 Valenciennes cedex 9, France, frederic.jenot@univ-valenciennes.fr), Mohamed Ourak (Institut d'Electronique de Microélectronique et de Nanotechnologie, Département Opto-Acousto-Electronique (UMR CNRS 8520), Université de Valenciennes et du Hainaut-Cambrésis, Le Mont-Houy, 59313 Valenciennes cedex

9, France, Mohamed.Ourak@univ-valenciennes.fr), Sébastien Desvaux (SKF Group, Z.I. n°2 Batterie 900 Rouvignies, 59309 Valenciennes, France, sebastien.desvaux@snfa.com)

The use of ceramic balls, in particular silicon nitride balls, allows a substantial improvement of bearing performances. For critical aerospace and space applications, there is a need for developing new nondestructive techniques for the characterization of these balls. We propose in this work to study the possibilities offered by a specific technique of resonant ultrasound spectroscopy of spheroidal modes. As shown by the theoretical study on elastic sphere vibrations, these modes allow to characterize the whole volume of balls or only the close-to-surface layers, according to the considered frequency range. To acquire the resonance spectra of these modes, a specific measurement system composed of a piezoelectric ultrasonic probe and an optical interferometer was developed. A self-implemented numerical processing of measured spectra allows to detect the resonance frequencies and to deduce from them the propagation velocity of the spheroidal waves in each inspected subsurface layers. Then, we propose a method based on these results that permit to estimate the elastic coefficients of the balls according to various inspection depths. This method has the advantage of providing very high precision evaluations of the elastic coefficients over a wide frequency range.

THURSDAY AFTERNOON, 3 JULY 2008

P3-A, LEVEL 3, 3:40 TO 5:20 P.M.

### Session 4pPAb

## Physical Acoustics and Computational Acoustics: Combustion Noise and Thermo-Acoustics II (Poster Session)

Tim Lieuwen, Cochair  
*Georgia Institute of Technology*

Maria Heckl, Cochair  
*Keele University*

Rafael Piscoya, Cochair  
*Technische Fachhochschule Berlin, Univ. of Applied Sciences*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pPAb1. Inversion of the bulk viscosity in nonequilibrium media with heat release and new acoustical properties of such media.** Nonna Molevich (P.N. Lebedev Physical Institute Samara branch, Pervomaiskaya Str 21-36, 443100 Samara, Russian Federation, molevich@fian.smr.ru)

New acoustical properties, which are caused by the inversion of the bulk (second) viscosity in nonequilibrium media, are investigated. Negative bulk viscosity can take place due to the positive feedback between the sound perturbation and the nonequilibrium heat release. During previous years, the conditions for the negative bulk viscosity existence were found in a large number of nonequilibrium media such as a vibrationally excited gas with stationary nonequilibrium, nonisothermal plasma, chemical active mixtures with irreversible reactions, media with nonequilibrium phases, upper

atmosphere layers, and earth magma with bubbles. The following phenomena are discussed for gaseous media with negative viscosity: (1) New dispersion characteristics (in particular, the low-frequency sound speed can exceed the high-frequency one); (2) Acoustically instability of nonequilibrium media, nonlinear mechanisms of the wave growth stabilization and stationary self-sustaining structures; (3) Linear and non-linear sound beam refraction including the self-focusing (due to two self-action mechanisms in acoustically active media: the gas cooling by sound and the excitation of acoustical streaming in direction opposite to sound propagation) and the anomalous reflection (with reflection coefficient  $R > 1$ ) on a boundary between equilibrium and nonequilibrium media; (4) The vortex and thermal wave amplification due to intensive parametric interactions in acoustically active media.



**4pPAb2. Stationary structures in acoustically active nonequilibrium media with one relaxation process.** Rinat Galimov (Samara State Aerospace University, Magnitnaya st. 15, 443017 Samara, Russian Federation, renrk@mail.ru), Nonna Molevich (P.N. Lebedev Physical Institute Samara branch, Pervomaiskaya Str 21-36, 443100 Samara, Russian Federation, molevich@fian.smr.ru)

Chemical active mixtures with irreversible reactions, vibrationally excited gases, and nonisothermal plasmas are examples of acoustically active nonequilibrium media. In such media it is possible the existence of stationary nonlinear structures that are different from the step-wise shock wave structures. In the first part of the present paper it is investigated the solutions of a general acoustical equation, describing in the second order perturbation theory a nonlinear evolution of wide spectrum acoustical disturbances in nonequilibrium media with one relaxation process. Its low- and high-frequency limits correspond to Kuramoto-Sivashinsky equation and the Burgers equation with a source, respectively. Stationary structures of general equation, the conditions of their establishment and all their parameters are found analytically and numerically. In acoustically active media it is predicted the existence of the stationary solitary pulse. Then, we consider 1-D relaxing gas dynamics system of equations with simple Landau-Teller model of relaxation. The possible stationary profiles are shown in nonequilibrium degree- stationary wave speed bifurcation diagram. The boundaries of this diagram are obtained in analytical forms. The field of weak shock wave instability is shown in this bifurcation diagram. Unstable shock wave disintegrates into the sequence of solitary pulses described by the general acoustical equation.

**4pPAb3. Thermoacoustic waves near the liquid-vapor critical point.** Pierre Carles (Université Pierre et Marie Curie, 4 Place Jussieu, 75005 Paris, France, carles@ccr.jussieu.fr)

The thermal relaxation in a fixed-volume cell filled with a near-critical fluid is governed by the rapid expansion of thermal boundary layers, which drive a series of thermo-acoustic waves in the bulk fluid. The long-term cumulative effect of these waves is to increase the pressure in the cell, which in turn leads to a global temperature increase (a process named the Piston Effect). Recently and for the first time, the thermo-acoustic waves produced by the Piston Effect have been measured experimentally using interferometric methods [Y. Miura et al., to appear in Phys. Rev. E (2006)]. In the present work, we use asymptotic methods in order to derive a complete theoretical model of the Piston-Effect-driven acoustic waves, applicable to real fluid equations of state and to arbitrary reduced temperatures. The predictions of this model are compared to the above-mentioned experimental data, and an excellent agreement is observed without any fitting parameter. This result confirms the high precision of the data in question, and shows that asymptotic models such as ours can be a powerful tool for analyzing the results of such experiments.

**4pPAb4. Prediction of noise emissions from industrial flares.** Carl-Christian Hantschk (Müller-BBM GmbH, Robert-Koch-Str. 11, 82152 Planegg, Germany, Carl-Christian.Hantschk@MuellerBBM.de), Edwin Schorer (Müller-BBM GmbH, Robert-Koch-Str. 11, 82152 Planegg, Germany, Edwin.Schorer@MuellerBBM.de)

In many industries where combustible waste gases are obtained, flares are used to burn these gases in a controlled manner. Among other environmental aspects, the noise emissions associated with flaring are becoming increasingly important in many countries as population density goes up and residential and industrial areas move closer together. Installing noise control equipment on flares is almost impossible while they are in service, since flares are typically a safety related plant component that can only be turned off after the connected plant has been shut down. Accordingly, in order to plan appropriate noise control measures in time and to avoid unnecessary costs, predicting the noise emissions of flares as early in the design process as possible is crucial. This requires knowledge of the relevant individual noise sources associated to the flare system and the ability to calculate their respective contribution - in the operating condition in question - to the overall noise emission, based on the data available in the planning stage. The present paper summarizes these sources and outlines some of the individual effects and parameters having an influence on the acoustical characteristics of flares.

**4pPAb5. Modelling of acoustic losses with the wave equation for the analysis of combustion instabilities.** Elke Wanke (Lehrstuhl für Thermodynamik, Technische Universität München, Boltzmannstrasse 15, 85747 Garching, Germany, wanke@td.mw.tum.de), Fabian Weyermann (Lehrstuhl für Thermodynamik, Technische Universität München, Boltzmannstrasse 15, 85747 Garching, Germany, weyermann@td.mw.tum.de), Christoph Hirsch (Lehrstuhl für Thermodynamik, Technische Universität München, Boltzmannstrasse 15, 85747 Garching, Germany, hirsch@td.mw.tum.de), Thomas Sattelmayer (Lehrstuhl für Thermodynamik, Technische Universität München, Boltzmannstrasse 15, 85747 Garching, Germany, sattelmayer@td.mw.tum.de)

A numerical design tool for the assessment of the stability of combustion chambers has been developed, which is able to compute geometrically complex systems with thermoacoustic feedback in the time domain. It is shown that internal acoustic losses can be considered although the method is based on the solution of the wave equation. The presented method overcomes a serious limitation of the original approach and allows to make quantitative predictions. The model is based on the Bernoulli equation and derives the required information from the spatial distribution of the loss of total pressure. For the purpose of a comprehensive validation of the model, simulations were carried out in the frequency domain before the model was implemented. As the internal acoustic losses in combustors stem almost completely from the flow separation at the exit of the burners the losses are independent from temperature. For this reason the influence of the flame was neglected in the study to only focus on the modelling of acoustic losses. The numerical results are validated with single burner test rig experiments.

## Session 4pPac

## Physical Acoustics: Quantum Acoustics II (Poster Session)

Michel DeBilly, Cochair  
*Institut Jean le Rond d'Alembert*

Walter Lauriks, Chair  
*Katholieke Universiteit Leuven*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pPac1. Acoustic studies of the martensite phase transition in Ti-Ni alloys.** Andrew A. Abramovich (St. Petersburg State Technological University of Plant Polymers, Chernykh str., 4, Vasenko str., 5/15, apt.32, 195197 St. Petersburg, Russian Federation, andrew@ns2740.spb.edu), Elena V. Charnaya (Department of Physics, St. Petersburg State University, Petrodvorets, Uljanovskaya, 1, 198504 St. Petersburg, Russian Federation, charnaya@paloma.spbu.ru), Sergei P. Belyaev (Department of Mathematics and Mechanics, St. Petersburg State University, Petrodvorets, Uljanovskaya, 1, 198504 St. Petersburg, Russian Federation, belyaev@paloma.spbu.ru), Aleksander E. Volkov (Department of Mathematics and Mechanics, St. Petersburg State University, Petrodvorets, Uljanovskaya, 1, 198504 St. Petersburg, Russian Federation, volkov@paloma.spbu.ru)

Acoustic studies of elastic properties and attenuation of ultrasonic waves (longitudinal and transverse) at the martensite phase transition were carried out in the Ti-Ni based polycrystalline alloys. Measurements were carried out using a pulse ultrasonic technique within a temperature range of 190 to 440 K upon continuous warming and cooling the samples after various hardening and annealing treatment. Anomalies of ultrasound velocity and attenuation were observed through the martensite phase transition which depended on wave polarization and thermal history of the samples, peaks of attenuation were seen only for transverse waves. A pronounced thermal hysteresis of acoustic features was also observed upon warming and cooling. The obtained results were treated on the basis of the phenomenological Landau theory for the B2-B19' ferroelastic phase transition.

**4pPac2. Observation of induced shear acoustic phonons by Brillouin scattering.** Taisuke Yoshida (Faculty of Engineering Doshisha University, 1-3 Tatara Miyakodani, 610-0321 Kyotanabe, Japan, dth0186@mail4.doshisha.ac.jp), Sigeo Murata (Faculty of Engineering Doshisha University, 1-3 Tatara Miyakodani, 610-0321 Kyotanabe, Japan, dth0186@mail4.doshisha.ac.jp), Takahiko Yanagitani (Faculty of Engineering Tohoku University, 6-6-05 Aramaki Aoba Aoba-ku, 610-0321 Kyotanabe, Japan, yanagi@ecei.tohoku.ac.jp), Mami Matsukawa (Doshisha University, 1-3, Tatara Miyakodani, 610-0321 Kyotanabe, Japan, mmatsuka@mail.doshisha.ac.jp)

The Brillouin scattering measurement is an efficient nondestructive and noncontact method which enables the simultaneous measurement of longitudinal and shear wave velocities at hypersonic frequencies. However, the measurement accuracy of shear wave velocities is low because of the weak

Brillouin scattering from the thermal phonons. In this study, therefore, we have tried to overcome this problem by making use of the induced coherent shear phonons. We have adopted the Reflection Induced  $\theta A$  (RI $\theta A$ ) scattering geometry for Brillouin scattering measurement. To induce shear acoustic phonons, we used an uniaxially aligned ZnO film transducer developed in our laboratory, which can be fabricated on any solid materials without the epitaxy technique. As a result, we obtained the intense Stokes peak in case of silica glass sample which is larger than that obtained from the thermal phonons. It means that observation of induced shear acoustic phonons was achieved. Because the RI $\theta A$  geometry enables the simultaneous measurement of longitudinal and shear phonons in plane, this technique opens the new feature for the nondestructive elasticity measurement.

**4pPac3. Study of acoustical phonon modes in superlattices with SiGe QDs.** Anatoliy Yaremko (Lashkaryov Institute of Semiconductor Physics, NAS of Ukraine, Prospekt Nauky 45, 03028 Kyiv, Ukraine, yaremko@isp.kiev.ua), Volodymyr Yurkymchuk (Lashkaryov Institute of Semiconductor Physics, NAS of Ukraine, Prospekt Nauky 45, 03028 Kyiv, Ukraine, yurkym@isp.kiev.ua), Volodymyr Dzhagan (Lashkaryov Institute of Semiconductor Physics, NAS of Ukraine, Prospekt Nauky 45, 03028 Kyiv, Ukraine, dzhagan@isp.kiev.ua), Mykhailo Valakh (Lashkaryov Institute of Semiconductor Physics, NAS of Ukraine, Prospekt Nauky 45, 03028 Kyiv, Ukraine, valakh@isp.kiev.ua)

Multilayers with SiGe nanoislands (QD's) grown in a broad temperature range are studied using Raman spectroscopy, HRXRD and compared with similar multilayers without islands. As the growth temperature increases, Si content in the islands increases, partially relieving strain. These structural transformations manifest themselves in both the intensity and frequency of the low-frequency Raman peaks. Due to composition- and strain-induced changes in the island band structure, excitation conditions come out of resonance, reducing Raman peak intensity. We have shown that at the interpretation of the Raman scattering by folded acoustic phonons for structures with nanoislands the real morphology of the island layer should be considered. The observed series of the low-frequency Raman peaks, for the multilayered structures with the number of QD layers below ten, is due to the acoustic phonon modes within the islands. The enhancement of the scattering intensity due to resonance of the excitation light with the electronic transitions within the islands plays a significant role. Theoretical analysis was fulfilled in the framework of microscopic approach with using Green function method and taking into account the real structure of QD's and their interaction with surrounding matrix.

**4pPac4. Evolution of ultrasound transmission during the sol-gel transition.** Bożena Gadomska (University of Warsaw, Department of Chemistry, Laboratory of Physicochemistry of Dielectrics and Magnetics, ul. Żwirki i Wigury 101, 01-905 Warsaw, Poland, bogad@chem.uw.edu.pl), Wojciech Gadomski (University of Warsaw, Department of Chemistry, Laboratory of Physicochemistry of Dielectrics and Magnetics, ul. Żwirki i Wigury 101, 01-905 Warsaw, Poland, gado@chem.uw.edu.pl)

The velocity of the ultrasound wave has been detected during the gelation process of the aqueous gelatin solution. We observe monotonic decrease of the ultrasound transmission in train of the gel network formation. In order to find the dependance of the ultrasound velocity on the number of the intermolecular bonds, responsible for gel formation, we have performed the simultaneous measurements of the optical activity evolution. The experiments have been performed in the gelatin gel and in the silica gel. We have described the process in terms of the percolation theory by power law, according to Landau theory of the second order phase transitions, and by fitting the theoretical results to the experimental curves we have found the appropriate critical exponents. We have also shown how the ultrasound velocity depends on the concentration of the gelling substance and on the temperature, in which the process of the sol-gel transition has been performed.

**4pPac5. Acoustic wave propagation in cubic piezoelectric semiconductor plates.** Bernard Collet (Université Pierre et Marie Curie - Institut Jean le Rond d'Alembert-UMR-CNRS 7190, T65-Case 162, 4 place Jussieu, 75252 Paris Cedex 05, France, bc@ccr.jussieu.fr)

Piezoelectric materials can be either dielectric or semiconductors. An acoustic wave propagating in a piezoelectric crystal is usually accompanied by an electric field. When the crystal is also a semiconductor, the electric

field produces currents and space charges, resulting in dispersion and acoustic loss. The interaction between a travelling acoustic wave and mobile charges in piezoelectric semiconductors is currently called the acoustoelectric effect. It was shown experimentally and proved theoretically that an acoustic wave travelling in a piezoelectric semiconductor can be amplified by the application of an initial dc electric field. Piezoelectric semiconductor devices often have structural design of plates or rods. Here we study the thin piezoelectric semiconductor plates. Two-dimensional equations for coupled extensional, flexural and thickness-shear motions are obtained systematically from the three-dimensional equations by retaining lower order terms in power series expansions in the plate thickness coordinate. The two-dimensional equations are specialized to crystals of cubic (43m) symmetry. Propagation of extensional, flexural and thickness-shear waves and their amplification by a dc electric field are analyzed.

**4pPac6. Ultrasonic annealing of radiation defects in silicon.** Artem Podolian (T Shevchenko Kiev National University, Dept of Physics, 03680 Kiev, Ukraine, gogi@univ.kiev.ua), Oleg Korotchenkov (T Shevchenko Kiev National University, Dept of Physics, 03680 Kiev, Ukraine, olegk@univ.kiev.ua)

Cold annealing of radiation defects in silicon is reported and the likely origin of the effect is suggested. The data on photocurrent transients and photoluminescence spectra are contrasted in  $\gamma$ -irradiated float zone and Czochralski silicon. The ultrasonic load is shown to slow down the current decay, indicative of the decreased defect densities. The microscopic model includes the likely pathways for E-center annealing in zone and C-O-V<sub>2</sub> complex redistribution in Czochralski materials. The carbon involvement into the annealing effect is further evidenced by taking the photoluminescence spectral evolution with applied ultrasonic loads. The usage of the presented technique in improving the performance of radiatively damaged silicon detectors is furthermore discussed.

THURSDAY AFTERNOON, 3 JULY 2008

P3-A, LEVEL 3, 3:40 TO 5:20 P.M.

## Session 4pPAd

### Physical Acoustics: General Topics in Nonlinear Acoustics II (Poster Session)

Thomas Matula, Cochair  
*University of Washington*

Murray Korman, Cochair  
*U.S. Naval Academy*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pPAd1. Parametric audible sounds by phase-cancellation excitation of primary waves.** Tomoo Kamakura (Dept. of Electronic Eng., Univ. of Electro-Communications, 1-5-1, Chofugaoka, 182-8585 Chofu-shi, Japan, kamakura@ee.uec.ac.jp), Shinichi Sakai (Dept. of Electronic Eng., Univ. of Electro-Communications, 1-5-1, Chofugaoka, 182-8585 Chofu-shi, Japan, sakashin@ew3.ee.uec.ac.jp), Hideyuki Nomura (Dept. of Electronic Eng., Univ. of Electro-Communications, 1-5-1, Chofugaoka, 182-8585 Chofu-shi, Japan, nomu@ee.uec.ac.jp), Masahiko Akiyama (Dept. of Electronic Eng., Univ. of Electro-Communications, 1-5-1, Chofugaoka, 182-8585 Chofu-shi, Japan, akiyama@ew3.ee.uec.ac.jp)

An ultrasound source with a simple configuration is considered as a theoretical model. The source with a circular aperture consists of two coaxially arranged planar emitters: i.e., one is an inner disc emitter and the other is an outer ring emitter. The active areas of these emitters are the same. The outer diameter of the source is 20 cm. Both the emitters are driven individually at the same frequencies of 40 and 42 kHz but different phase angles. Especially, we focus on two extreme cases of the usual in-phase driving and out-of-phase driving. Numerical computation using the KZK equation demonstrates that when the driving signals are in phase the difference frequency beam of a 2-kHz wave has a candle-flame-like directivity. The beam has a

similar directivity when the signals are out-of-phase by 180 degrees, although the peak of the sound pressure level decreases by few decibels. Interestingly, the second harmonic pressure level of the difference frequency reduces by ten decibels and more. Needless to say, the pressure amplitudes of the primary waves are suppressed considerably near the beam axis. Experimental verification is done using an airborne ultrasound source with a 19.2-cm circular aperture.

**4pAd2. The Generation of Acoustic Waves and Cavitation Processes in Regime of Phase Synchronization During Multichannel Discharges in Electrolyte.** Vyacheslav S. Teslenko (Lavrentyev Institute of Hydrodynamics SB RAS, Lavrentyev ave., 15, 630090 Novosibirsk, Russian Federation, vteslenko@academ.org), Alexey P. Drozhzhin (Lavrentyev Institute of Hydrodynamics SB RAS, Lavrentyev ave., 15, 630090 Novosibirsk, Russian Federation, ruslan@hydro.nsc.ru), Ruslan N. Medvedev (Lavrentyev Institute of Hydrodynamics SB RAS, Lavrentyev ave., 15, 630090 Novosibirsk, Russian Federation, drozh@hydro.nsc.ru)

In the paper the problems of acoustic waves generation of required frequency, amplitude and profile by the phase synchronization of multichannel discharges are considered. The phase synchronization of discharges is supplied by the implementation in the discharge circuit of additional inductance which acts as a dynamic feedback and ensures the selfsynchronization of N generated bubbles and acoustic wave's radiation [1]. The devices of such types may be used to form acoustic fields with required frequency and shape. References: I. V. S. Teslenko, R. N. Medvedev, and A. P. Drozhzhin, Self-Synchronization of Electrohydrodynamic Autooscillations during Multichannel Discharges in Electrolyte // ISSN 1063-7850, Technical Physics Letters, 2007, Vol. 33, No. 10, pp. 833-836. <http://www.swsl.newmail.ru/publ/TEPL833.pdf>

**4pAd3. Acoustic properties of TaCl - TaBr mixed crystals.** Farkhad Akhmedzhanov (Navoi State Mine Institute, 27a Yuzhnaya Street, 210100 Navoi, Uzbekistan, farkhad2@yahoo.com)

Attenuation coefficient and phase velocity of transversal and longitudinal acoustic waves in TaCl - TaBr mixed crystals have been investigated by Bragg light diffraction on the acoustic waves. The measurements were carried out at the frequencies from 200 to 1200 MHz at home temperature. Moreover, the phase velocity was determined by using Raman-Nath diffraction and optical heterodyning at 10 MHz. The investigations shown, that the change of acoustic properties in the investigated mixed crystals is nonadditive. Nonlinear interaction constants have been calculated taking into consideration various factors, which can influence on the propagation of acoustic waves in mixed crystals. The results were compared with similar investigations in NaCl - NaBr mixed crystals.

**4pAd4. Adjusting the phase of the signals transmitted from dual frequency probe for reducing second harmonic during propagation.** Pasovic Mirza (Université de Lyon, 43 boulevard du 11 Novembre 1918, 69622 Villeurbanne, France, mirza.pasovic@creatis.insa-lyon.fr), Christian Cachard (Université de Lyon, 43 boulevard du 11 Novembre 1918, 69622 Villeurbanne, France, christian.cachard@creatis.insa-lyon.fr), Guillaume Matte (Biomedical Engineering, Erasmus MC, P.O. Box 2040, 3000 CA Rotterdam, Netherlands, g.matte@erasmusmc.nl), René Van Der Steen (Eindhoven University of Technology, Dept. of Mechanical Engineering, Dynamics & Control, P.O. Box 513, 5600MB Eindhoven, Netherlands, r.v.d.steen@tue.nl), Nico De Jong (Erasmus MC, Dr Molewaterplein 50 room Ee2302, 3015GE Rotterdam, Netherlands, n.dejong@erasmusmc.nl), Olivier Basset (Université de Lyon, 43 boulevard du 11 Novembre 1918, 69622 Villeurbanne, France, olivier.basset@creatis.insa-lyon.fr)

During propagation of the ultrasound wave, nonlinearities of the medium, causes rise of higher harmonics that are seen in amplitude spectrum. In ultrasound contrast harmonic imaging this is unwanted effect, since it is expected to image only higher harmonics produced by contrast agents. In previous studies Christopher, Krishnan and Shen proposed transmitting either an inverted signal (collected by hydrophone), transmitting 2nd harmonic with broadband transducer or transmitting third harmonic to reduce the second harmonic at measurement point. Our study uses a dual frequency

probe which transmits one wave at frequency  $f_0$  and one wave at double frequency  $2f_0$  named second harmonic reduction signal (SHRS) for reducing the second harmonic in the propagating signal. The hardest task is the appropriate adjustment of the phase of the SHRS to reduce second harmonic during propagation in the focal area. We outline how to set phase and established that the phase of the SHRS is not affected by the nonlinear parameter  $B/A$  and excitation pressure  $P_0$  but rather by the excitation frequency and the ultrasound system geometry (transducer size and distance to focal area). A reduction of 25dB has been obtained in simulation ( $P_0=100\text{kPa}$ ,  $f_0=2.25\text{ MHz}$ ). Experiments in water tank, have shown the reduction of 2nd harmonic by 30dB.

**4pAd5. Energy pumping in acoustics.** Romain Bellet (Laboratoire de Mécanique et d'Acoustique - CNRS, 31 chemin Joseph Aiguier, 13402 Marseille, France, bellet@lma.cnrs-mrs.fr), Bruno Cochelin (Laboratoire de Mécanique et d'Acoustique - CNRS, 31 chemin Joseph Aiguier, 13402 Marseille, France, bruno.cochelin@ec-marseille.fr), Pierre-Olivier Mattei (Laboratoire de Mécanique et d'Acoustique - CNRS, 31 chemin Joseph Aiguier, 13402 Marseille, France, mattei@lma.cnrs-mrs.fr), Philippe Herzog (Laboratoire de Mécanique et d'Acoustique - CNRS, 31 chemin Joseph Aiguier, 13402 Marseille, France, herzog@lma.cnrs-mrs.fr)

Energy pumping corresponds to a particular vibratory regime of a non-linear system coupled to a linear primary structure. Its non-linear behaviour allows to reduce vibrations of the primary structure, and is thus a new passive vibration control technique. This phenomenon has mainly been studied in mechanical engineering, so we transposed its principle to noise control in an acoustic medium. The presentation will focus on results about acoustic energy pumping that we observed both experimentally and numerically, in the time and frequency domains. These results highlight two main points: physically, energy pumping corresponds to an irreversible transfer of energy from the primary system to the non-linear absorber, and practically, its effects are a noise level limitation in the acoustic medium in permanent regime and a much faster sound extinction in transient regime.

**4pAd6. Shock wave propagation in heterogeneous medium, from ultrasound to sonic boom.** Lili Ganjehi (Institut Jean Le Rond d'Alembert, Université Pierre et Marie Curie, Boîtes 161 et 162, 4 place Jussieu, 75252 Paris Cedex 05, France, ganjehi@lmm.jussieu.fr), François Coulouvrat (Institut Jean Le Rond d'Alembert, Université Pierre et Marie Curie, Boîtes 161 et 162, 4 place Jussieu, 75252 Paris Cedex 05, France, coulouvrat@ccr.jussieu.fr), Jean-Louis Thomas (Centre National de la Recherche Scientifique, Institut des NanoSciences de Paris, Université Pierre et Marie Curie, 4 place Jussieu, 75252 Paris Cedex 05, France, thomasjl@ccr.jussieu.fr), Régis Marchiano (Institut Jean Le Rond d'Alembert, Université Pierre et Marie Curie, Boîtes 161 et 162, 4 place Jussieu, 75252 Paris Cedex 05, France, marchi@lmm.jussieu.fr)

Strong variability of sonic boom due to the atmospheric turbulence is known since the first test flight recordings in the 1960's. To simulate this, a laboratory scale experiment is conducted with ultrasonic shock waves in water at 1 MHz. The experiment is designed for an optimal 1:100,000 scaling with sonic boom. It includes single or multiple heterogeneities of varying sizes but comparable to the acoustical wavelength. Its deterministic aspect allows detailed comparisons with the results of a numerical model based on a nonlinear wide angle parabolic approximation. The experiments show the following features of the shock wave propagation : wavefront folding, local amplification (acoustical lens effect), increase of the rise time, strong variability of the time waveforms. All these features are in qualitative agreement with sonic boom observations. They are observed for a single heterogeneity provided this one is sufficiently large, but are amplified in case of multiple heterogeneities. Comparisons with results of numerical simulations show good agreement in various configurations. Improvements provided by the wide angle approach with respect to the standard parabolic approximation will also be discussed. Preliminary simulations for N wave propagation in a randomly heterogeneous medium will finally be presented.



#### **4pPAAd7. A route to chaotic state on an electrodynamic loudspeaker.**

Antonio Petosic (Faculty of Electrical Engineering and Computing, Unska 3, 10000 Zagreb, Croatia, antonio.petosic@fer.hr), Ivan Djurek (Faculty of Electrical Engineering and Computing, Unska 3, 10000 Zagreb, Croatia, ivan.djurek@fer.hr), Djurek Danijel (AVAC, Kesten brijeg 15, 10000 Zagreb, Croatia, danijel.djurek@zg-net.hr)

The low frequency electrodynamic loudspeaker (EDL) unit has been measured and analyzed in terms of chaotic behavior. It was found that an electrodynamic loudspeaker can function as a chaotic system. Loudspeaker impedance and vibration amplitude as function of driving frequency were measured at various driving currents, and well-known cut-off effect from nonlinear dynamical systems has been noticed. In the frequency region near cut-off and at higher driving currents the period doubling and later chaotic state occur. The experimentally obtained chaotic state was confirmed theoretically solving nonlinear equation of motion with strong nonlinear effective stiffness spatial dependency. It was found that statically measured suspension effective stiffness does not enable chaotic state when included in differential equation, and it has been concluded that membrane viscoelastic properties enhance the restoring force far enough to obtain chaos. The nonlinear equation describing anharmonic periodically driven oscillator has been solved numerically and the theoretical results were compared to experimental results.

#### **4pPAAd8. Analysis of the effects of the oscillations of a rigid sphere inside a cylindrical cavity containing a standing acoustic wave.**

Edgar A. Torres (Universidad Nacional Autonoma de Mexico, Centro de Ciencias Aplicadas y Desarrollo Tecnológico, CCADET-UNAM, Circuito Exterior s/n, Cd. Universitaria, A. P. 70-186, 02510 Mexico, D.F., Mexico, edgar.augusto.torres.gallegos@gmail.com), Arturo Santillan (Universidad Nacional Autonoma de Mexico, Centro de Ciencias Aplicadas y Desarrollo Tecnológico, CCADET-UNAM, Circuito Exterior s/n, Cd. Universitaria, A. P. 70-186, 02510 Mexico, D.F., Mexico, arturo.orocho@ccadet.unam.mx)

Under certain driving conditions of a single-axis acoustic levitation device, a suspended sample leaves its stability state and starts to oscillate vertically around the initial equilibrium position. A published theory on such instabilities [J. Rudnick and M. Barmatz, "Oscillational instabilities in single-mode acoustic levitators," J. Acoust. Soc. Am., 87(1), 81-92, 1990]

predicts the occurrence of time delays between the response of the cavity of the device and the motion of the sample inside it. In this paper, a theoretical and experimental investigation on similar time delay effects will be described. A solid sphere was moved in a controlled way inside a closed cylindrical cavity by means of a rod connecting the object to the outside of the system. A standing wave was generated inside the cavity by using a speaker. In this way, oscillations of the sphere were produced and the response of the sound field to such movement was studied. The effect of the frequency of the oscillations of the sphere on the time delay between the sound pressure and the movement of that object will be reported. In addition, the relations between the obtained results and the published theory on oscillational instabilities will be discussed.

#### **4pPAAd9. Manipulation of the behavior of SiC particles and oil bubbles using ultrasonic standing wave field.**

Seung Hyun Cho (Korea Research Institute of Standards and Science, Doryong dong 1, Yuseong gu, 305-340 Daejeon, Republic of Korea, seungcho@kriss.re.kr), Dae-Cheol Seo (Korea Research Institute of Standards and Science, Doryong dong 1, Yuseong gu, 305-340 Daejeon, Republic of Korea, dcseo@kriss.re.kr), Bong Young Ahn (Korea Research Institute of Standards and Science, Doryong dong 1, Yuseong gu, 305-340 Daejeon, Republic of Korea, ahnby@kriss.re.kr)

Using ultrasound, particles submerged or flowing in fluid can be manipulated since ultrasound has an effect on the behavior of particles. Specifically, in standing wave field, particles generally move to pressure nodes or pressure antinodes due to acoustic radiation force. In this work, the behavior of SiC particles and oil bubbles in flowing water by standing wave field was investigated. Standing wave field in frequencies between 2 and 2.5 MHz was formed in a few mm narrow flow channel using a water coupled ultrasonic transducer and a steel reflector. We observed the effect of the standing wave parameters such as frequency, flow channel width, or sound intensity on the behavior of the particles. Various interesting results were obtained through some experiments. We separated SiC particles and oil bubbles. It was shown that the operating frequency of standing wave can control the particle moving location. Sound intensity increase also leads to the entrapment of moving particles. The resulted observations reveal the possibility of various applications of the ultrasonic standing wave to the manipulation of particles.

## Session 4pPae

## Physical Acoustics: Ultrasonics: NDT and Layered Systems II (Poster Session)

Vincent Pagneux, Cochair

*Laboratoire d'Acoustique de l'Université du Maine*

Claire Prada, Cochair

*Université Paris 7*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pPae1. Dual frequency ultrasound- A pulse-echo technique for analysis of layered material.** Ossi Riekkinen (University of Kuopio, POB 1627, 70211 Kuopio, Finland, Ossi.Riekkinen@uku.fi), Mikko Hakulinen (University of Kuopio, POB 1627, 70211 Kuopio, Finland, Mikko.Hakulinen@uku.fi), Juha Töyräs (Kuopio University Hospital, POB 1777, 70211 Kuopio, Finland, Juha.Toyras@kuh.fi), Jukka Jurvelin (University of Kuopio, POB 1627, 70211 Kuopio, Finland, Jukka.Jurvelin@uku.fi)

Ultrasound (US) thickness gauges typically analyse layered materials by utilizing ultrasound reflections between different layers and prior knowledge for the material order within the layered structure. In this study, a dual frequency ultrasound (DFUS) technique is applied to eliminate the effect of overlying layered structure on the measurements of the object of interest without prior knowledge of the order of materials within the multilayered structure. DFUS technique utilizes prior knowledge on US attenuation coefficient and speed at two frequencies in multilayered materials, consisting of two different material types. Then, US reflection from the front (first) and the back (last) surfaces of the multilayered structure is measured using two different US frequencies. No reflections from the internal interfaces are needed. The technique was validated using several elastomer samples and their combinations, measured at 2.25 MHz and 5.0 MHz. DFUS reduced the mean error, induced by the overlying elastomers, in reflection from the object of interest from 103.6 - 289.4% to -15.9 - 5.6% and from 37.5 - 77.5% to -12.0 - 4.9% with 5.0 MHz and 2.25 MHz, respectively. Based on these results, DFUS is a straightforward technique to analyse the multilayered structure without the need for echoes from internal interfaces.

**4pPae2. Ultrasonic pulse propagation in a bonded three-layered structure.** Jose Luis San Emeterio (Instituto de Acustica CSIC, Serrano 144, 28006 Madrid, Spain, jluis@ia.cetef.csic.es), Antonio Ramos (Instituto de Acustica CSIC, Serrano 144, 28006 Madrid, Spain, aramos@ia.cetef.csic.es), Emilia Pardo (Instituto de Acustica CSIC, Serrano 144, 28006 Madrid, Spain, empargo@ia.cetef.csic.es), Julio C. Leite (COPPE, Universidad Federal Rio de Janeiro, 68501 Rio de Janeiro, Brazil, juliobastosleite@jahoo.com.br), Jose Miguel Alvarez (INTA, Carretera Ajalvir, Km 4, 28850 Madrid, Spain, miguelal@inta.es), Carlos Perez Trigo (INTA, Carretera Ajalvir, Km 4, 28850 Madrid, Spain, pereztrc@inta.es)

Bonded layers are used in the assembly of many critical functional parts of industrial equipment. In this work, ultrasonic pulse propagation in a steel-rubber-rubber bonded composite structure is investigated by means of computer simulation and pulse echo experimental evaluation. Ultrasonic pulse propagation is modelled using a 2D time domain finite-difference software. For the experimental measurements, two test samples were fabricated by bonding a thin layer of steel and two thin layers of rubber, including deb-

onded areas at marked regions of each interface. Several ultrasonic traces were acquired by contact pulse-echo testing, using a 5 MHz wideband transducer, from the external steel surface. The large acoustic impedance mismatch existing between steel and rubber layers makes that only a very small part of the ultrasonic energy is transmitted through the first (steel-rubber) interface. The high attenuation in rubber materials and the possible overlapping of multiple echoes are additional characteristics of the complex ultrasonic pulse propagation in this flat structure. Some differences in time and frequency domains, between the received signals from normal bonded areas and completely debonded areas are discussed, looking for defect detection at the first (steel-rubber) and second (rubber-rubber) interfaces.

**4pPae3. Inverse Problem to Determine an Epoxy Film Parameters in a Tri-Layer Metal/Adhesive/Metal Structure.** Naima Ismaili Alaoui (Laboratoire d'Electronique Signaux-Systemes et Informatique LESSI, Université Sidi Mohammed Ben Abdellah, Faculté des Sciences Dhar el Mehraz, 30000 Fes, Morocco, nanialaoui@hotmail.com), 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), Driss Chenouni (Laboratoire d'Electronique Signaux-Systemes et Informatique LESSI, Université Sidi Mohammed Ben Abdellah, Faculté des Sciences Dhar el Mehraz, 30000 Fes, Morocco, d\_chenouni@yahoo.fr), Zakia Lakhlii (Laboratoire d'Electronique Signaux-Systemes et Informatique LESSI, Université Sidi Mohammed Ben Abdellah, Faculté des Sciences Dhar el Mehraz, 30000 Fes, Morocco, zlakhliai@hotmail.fr), Jean-Louis Izbicki (LOMC FRE-3102 CNRS, Groupe Ondes Acoustiques, University of Le Havre, Place Robert Schuman, BP 4006, 76610 Le Havre, France, jean-louis.izbicki@univ-lehavre.fr), Bruno Morvan (LOMC FRE-3102 CNRS, Groupe Ondes Acoustiques, University of Le Havre, Place Robert Schuman, BP 4006, 76610 Le Havre, France, bruno.morvan@univ-lehavre.fr)

The aim of this work is to give the elastic parameters and the thickness of a slim epoxy film of a tri-layer metal/adhesive/metal structure, by inverse problem on the experimental data, knowing the characteristic equation of the structure. This equation is firstly solved and the sensitivity of some modes to the parameters of interest is studied. The couple of solutions wavenumber-frequency obtained from the direct problem are reintroduced in the characteristic equation, looking for the longitudinal and shear waves velocities, and the epoxy thickness that minimise this equation. A criterion that takes into account the averaged and the variance values of each parameter is deduced and allows these parameters determination in  $\pm 3\%$  of the used values. To test the reliability of the method, the wavenumber solution is randomly perturbed in  $\pm 5\%$  range, as the experimental wavenumber is expected to be weakly different from the theoretical one. The velocities and the thickness obtained still in the  $\pm 3\%$  range around the used values. This

method is then applied on experimental data obtained from a Duralumin-epoxy/Duraluminum structure. The longitudinal and shear velocities and the thickness of the epoxy film obtained are in  $\pm 3\%$  around the given values.

**4pPAe4. The analysis of diffraction effects of acoustic waves on the crack's top.** Yulia V. Zhitlukhina (Institute of Metal Physics, 18, Sofia Kovalevskaya St., GSP-170, 620041 Ekaterinburg, Russian Federation, julika\_sun@mail.ru), Dmitry V. Perov (Institute of Metal Physics, 18, Sofia Kovalevskaya St., GSP-170, 620041 Ekaterinburg, Russian Federation, peroff@imp.uran.ru), Anatoly B. Rinkevich (Institute of Metal Physics, 18, Sofia Kovalevskaya St., GSP-170, 620041 Ekaterinburg, Russian Federation, rin@imp.uran.ru)

Diffraction effects and features of acoustic wave propagation in elastic media with microcrack were investigated in detail for pulse probing signals. The crack's plane was oriented across the direction of longitudinal ultrasonic wave incidence so that the detection of such a crack with so "inconvenient" spatial location is difficult enough by using traditional acoustic techniques. By using laser interferometer, the set of instantaneous pictures of acoustic field on the specimen's surface, corresponding to different time moments was obtained what allowed investigating and visualizing of acoustic field propagation and diffraction's effects on the crack's top in dynamics. Using various methods of numerical modeling of diffraction processes of acoustic wave on the crack's edge and top for pulse signals the origin of V-like structures on the snapshots of acoustic fields was explained and analyzed.

**4pPAe5. Structural Health Monitoring using cross-correlation of an ambient noise field.** Najib Abou Leyla (UVHC, IEMN-DOAE, Le Mont Houy, 59313 Valenciennes, France, najib.abouleyla@univ-valenciennes.fr), Emmanuel Moulin (UVHC, IEMN-DOAE, Le Mont-Houy, 59313 Valenciennes, France, emmanuel.moulin@univ-valenciennes.fr), Jamal Assaad (UVHC, IEMN-DOAE, Le Mont-Houy, 59313 Valenciennes, France, jamal.assaad@univ-valenciennes.fr), Sébastien Grondel (UVHC, IEMN-DOAE, Le Mont-Houy, 59313 Valenciennes, France, sebastien.grondel@univ-valenciennes.fr), Pascal Poussot (UVHC, IEMN-DOAE, Le Mont Houy, 59313 Valenciennes, France, pascal.poussot@univ-valenciennes.fr)

Theoretical and experimental studies in underwater acoustic, seismology and more recently ultrasonic have demonstrated that an estimate of the Time Domain Green Function (TDGF) between two receivers could be obtained from the cross-correlation of a diffuse acoustic noise field of these two receivers. The aim of the work is to exploit this technique in order to characterize Structural Health Monitoring (SHM) of aeronautic structures without the use of active sources. In this case, the aero-acoustic and/or mechanical sources (engine) generate an ambient noise field with some imperfections for the application. Indeed, source concentrations, source directivity and non-random components in time can appear which leads to an erroneous estimate of the TDGF. A key point to the study is thus to understand the influence of such imperfections. Therefore, experimental measurements have been performed using different types of acoustic noise sources

(localized or diffuse). Cross-correlation results obtained in each case are then compared and theoretically interpreted. Finally, the potential of this technique in terms of damage detection is verified.

**4pPAe6. Prediction of ultrasonic noise and attenuation for the simulation of non destructive testing.** Frédéric Jenson (CEA-LIST, Centre de Saclay, 9119 Gif-sur-Yvette, France, frederic.jenson@cea.fr), Vincent Dorval (CEA-LIST, Centre de Saclay, 9119 Gif-sur-Yvette, France, vincent.dorval@cea.fr), Gilles Corneloup (LCND - Université de la Méditerranée, IUT Aix Provence, Avenue Gaston Berger, 13625 Aix en Provence Cedex, France, gilles.corneloup@univmed.fr)

Ultrasonic non destructive testing of some polycrystalline materials can be significantly affected by their microstructure. In such materials a fraction of the acoustic energy is redirected in all directions, which leads to both attenuation and structural noise and causes significant loss in detection performances. Consequently, being able to predict these phenomena would help in designing better testing procedures. During previous works at CEA-LIST, noise and attenuation models have been developed and implemented into the simulation software for non destructive testing CIVA. The noise model describes the microstructure of the material as a set of point-like scatterers and the attenuation model uses a filtering approach. They both require reference ultrasonic measurements to reproduce the behaviour of a given material. The connection of this approach to a scattering model relating noise and attenuation to microstructural characteristics is studied in this work. The selected model is based on the Born approximation and allows one to relate physical quantities such as the scattering cross section and the attenuation coefficient to second order statistical properties of the microstructure and to elastic properties of a single crystallite. This model accounts for important effects such as anisotropic scattering and mode conversions, but neglects multiple scattering events. Simulation results obtained with this approach are compared to experimental results.

**4pPAe7. Ultrasonic control of the adhesion quality of two aluminium sheets.** Naima Taifi (Fac. des sciences; Univ. Chouaib Doukkali, B.P: 20, 24000 El Jadida, Morocco, taifinaima@yahoo.fr), Bouazza Faiz (Ibn Zohr University, FS Agadir, 80000 Agadir, Morocco, faizbou@hotmail.com), Ali Moudden (Ibn Zohr University, FS Agadir, 80000 Agadir, Morocco, ali\_moudden@yahoo.fr), Gerard Maze (LAUE, Université du Havre, Place Robert Schuman, F-76610 Le Havre, France, gerard.maze@univ-lehavre.fr), Dominique Decultot (LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, dominique.decultot@univ-lehavre.fr), Driss Izbaïm (Ibn Zohr University, FS Agadir, 80000 Agadir, Morocco, driss\_izbaim@yahoo.fr)

In this work, we present two ultrasonic methods allowing to control the quality of adhesion of two aluminium sheets with the same thickness. These methods are based on the analysis of the ultrasonic signals retrodiffused by the Al/glue/Al structure. The first method consists in controlling the behaviour of the mode of the sheets which is splitted. Two parameters controlling the transfers were allowed to characterize any type of adhesion: good, bad and intermediate. The second method is based on control of the width of the mode of the adhesive. The representation of this width by Argand diagram allows to control the quality of adhesion from the measurement of the diagram diameter.

## Session 4pPAf

## Physical Acoustics: Ultrasonics: Industrial NDT II (Poster Session)

Bertrand Nongaillard, Cochair  
IEMN-DOAE

Christophe Aristegui, Cochair  
Université Bordeaux

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pPAf1. Simultaneous Sum-Frequency and Vibro-Acoustography Imaging for Nondestructive Evaluation (NDE) and Testing (NDT) Applications.**

Farid Mitri (Dep. of Physiology and Biomed. Eng., Mayo Clinic College of Medicine, 200 First Street SW, Rochester, MN 55905, USA, mitri.farid@mayo.edu), Glauber T. Silva (Instituto Nacional de Matematica Pura e Aplicada, 22460 Rio de Janeiro, Brazil, mitri@ieec.org), James Greenleaf (Dep. of Physiology and Biomed. Eng., Mayo Clinic College of Medicine, 200 First Street SW, Rochester, MN 55905, USA, jfg@mayo.edu), Mostafa Fatemi (Dep. of Physiology and Biomed. Eng., Mayo Clinic College of Medicine, 200 First Street SW, Rochester, MN 55905, USA, fatemi@mayo.edu)

High-resolution ultrasound imaging systems for inspection of defects and flaws in materials are of great demand in many industries. Among these systems, Vibro-acoustography (VA) has shown excellent capabilities as a non-contact method for non-destructive high-resolution imaging applications. This method consists of mixing two confocal ultrasound beams, slightly shifted in frequency, to produce an acoustic emission field at the difference frequency of the primary incident ultrasound beams. In addition to the difference frequency signal, there exists another signal at the sum frequency, formed in the intersection region of the two primary beams. The goal of this study is to investigate the formation of high-resolution images using the sum frequency of ultrasound waves in VA while concurrently forming the conventional difference-frequency VA image, thereby increasing the amount of information acquired during a single scan. A theoretical model describing the sum frequency wave propagation, including beam forming and image formation in the confocal configuration is developed and verified experimentally. Moreover, sample experiments are performed on a flawed fiber-reinforced ceramic composite plate. Images at both the difference and sum frequencies are compared and discussed. Results show that the sum frequency image produces a high-resolution C-scan of the plate by which the flaws and structural details of the plate can be detected.

**4pPAf2. Ultrasonic polar c-scan system for range of material sizes and its capabilities for non-destructive testing.**

Kyle Barbour (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, nfdaaa@gmail.com), John M. Vander Weide (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, nico.declercq@me.gatech.edu), Sarah W. Herbison (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, sherbison@gatech.edu), Nico F.

Declercq (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, nico.declercq@me.gatech.edu)

The principle of multi-directional incident ultrasound has already shown to be a promising technique for the nondestructive evaluation of composites and other materials. The advantage is the correspondence between stiffness, damage and the registered double through transmission patterns. C-scans are widely used as a tool for the detection of defects in materials. A new ultrasonic scanner has been developed, called the Polar C-Scan, which enables efficient polar scan measurements in combination with C-scan capabilities. The scanner opens many possibilities for nondestructive testing by means of polar scans, C-scans, single transmission, double through transmission and even reflection. Furthermore the system produces almost no noise, which enables highly sensitive measurements in the time domain and the frequency domain. The presented work shows a thorough investigation of all the capabilities of the system and presents results for fiber reinforced composites after different fatigue cycles.

**4pPAf3. Ultrasonic and acoustic method for viscoelastic complex media characterization.**

Georges Nassar (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, gnassar@univ-valenciennes.fr), Fabrice Lefebvre (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, Fabrice.Lefebvre@univ-valenciennes.fr), Alain Skaf (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, alain.skaf@univ-valenciennes.fr), Bertrand Nongaillard (IEMN - DOAE, Université de Valenciennes, Le Mont - Houy, 59313 Valenciennes, France, Bertrand.Nongaillard@univ-valenciennes.fr)

In this paper, the potentialities of a low frequency ultrasonic/acoustic technique devoted to the study and characterization of the viscoelastic complex media is investigated. This work shows the limit of the use of ultrasound in a viscoelastic media with a complex matrix. In this context the cheese was indicated as a model of propagation medium, such a product having a very complex matrix in term of texture, openings, crystallization, moisture,... . Theoretical basis of sound attenuation in cheese is recalled, especially the effects of the matrix viscoelasticity and the scattering of ultrasonic energy by holes and cracks. Depending on the degree of openness, ultrasonic velocity or attenuation is chosen to represent the evolution of the cheese. For very high degree of openness, ultrasounds are no longer usable and a tap-test acoustic technique is employed and allows a quality indicator to be constructed. Experimental validations were done with optical images of cut cheeses and rheological measurements. The results indicate that a high degree of sensitivity can be reached with ultrasonic / acoustic non destructive technique.



## Session 4pPAg

Physical Acoustics: Nonlinear Acoustics of Consolidated Materials and Non Destructive Testing II  
(Poster Session)

James Tencate, Cochair

*EES-11 (Geophysics) - Los Alamos National Laboratory*

Koen Van Den Abeele, Cochair

*K.U.Leuven Campus Kortrijk*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pPAg1. Theoretical investigations into the use of linear and nonlinear inhomogeneous waves for non-destructive testing.** John M. Vander Weide (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, nico.declercq@me.gatech.edu), Nico F. Declercq (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, nico.declercq@me.gatech.edu)

Following the shift to incorporate more materials by design into modern engineering is the theoretical need to model an increasingly more complicated material structure and predict a design life for complex engineered materials. The presence of defects, interface cracks and delaminations adds to the inherent nonlinearity of a structure by degrading material stiffness and other physical properties contributing to fracture. The presence of local defects either at interfaces between two different materials or grain boundaries within the bulk material provide a natural location for the formation and subsequent propagation of cracks. This work provides initial theoretical investigations into the use of ultrasonic inhomogeneous waves to nondestructively probe such interface defects. Characterized theoretically by a complex valued wave number, the inhomogeneous wave is the more general plane wave solution of the wave equation. The complex valued wave number allows for the material effects of damping along the propagation direction and wave inhomogeneity or amplitude decay along the wavefront. To study the interaction with cracks, a singular integral equation formulation like that of Krenk and Schmidt is extended to farfield simulations with ultrasonic inhomogeneous waves.

**4pPAg2. Acoustic emission of structures basis on the A3B5 compounds.** Vitaly V. Veleschuk (Institute of Semiconductors Physics, NASU, Prospect Nauki 45, 03028 Kyiv, Ukraine, vvvit@ukr.net), Oleg V. Lyashenko (Institute of Semiconductors Physics, NASU, Prospect Nauki 45, 03028 Kyiv, Ukraine, lyashenk@mail.univ.kiev.ua), Olexander I. Vlasenko (Institute of Semiconductors Physics, NASU, Prospect Nauki 45, 03028 Kyiv, Ukraine, vvvit@pochta.ru)

Failure of internal mechanical pressure at origin and movement of dislocations in processes of fast degradation and defect formation in LED and semiconductor lasers lead to occurrence of acoustic emission (AE) - to the phenomenon of radiation of pulse spontaneous acoustic wave's noise character. This failure of internal mechanical pressure in local volumes with occurrence AE is possible only at action of external fields (influences) of the certain physical nature, in our case - non-uniform fields of the thermomechanical pressure created by a constant direct current. In work were investigated n+-n-p- structures basis on the GaP:N, GaP0.85As0.15:N, Zn-O, In-GaN/GaN and p+-p-n- structures basis on the Ga0.7Al0.3As and Ga0.65Al0.35As. During step-by-step increase of a current 10-25 multiple

change of AE occurrence threshold and a destruction threshold of structures was revealed at natural ageing structures after 6'108 s. For some samples of structures at low temperatures (77) AE occurrence threshold came nearer or even corresponded to a destruction threshold. The given effect explains gradual saturation of dislocations by atoms of impurity with formation of Cottrell cloud, that considerably lowers their mobility and increases activation energy and accordingly a AE occurrence threshold

**4pPAg3. Polar scans as a nonlinear acoustics tool.** John M. Vander Weide (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, nico.declercq@me.gatech.edu), Nico F. Declercq (Georgia Tech Lorraine - G.W. Woodruff School of ME, UMI Georgia Tech - CNRS 2958, 2 rue Marconi, 57070 Metz, France, nico.declercq@me.gatech.edu)

Nonlinear acoustic wave response provides a powerful means of assessing material properties. One very interesting area of application is testing for material damage. While an undamaged material may have a very linear acoustic response, damage such as crack formation creates strong nonlinearity with acoustic excitation amplitude. Many nonlinear acoustics techniques for nondestructive testing analyze the harmonic content in the wave scattered from a crack. A planar c-scan can be used in conjunction with the nonlinear acoustics techniques to provide defect detection and imaging. Polar scans are a complementary technique for damage assessment which has not yet found application in nonlinear acoustics. In the polar scan, the sound field in the hemisphere around a material point is measured and plotted to display a unique signature of local material properties. This work provides initial investigations into the use of the polar scan technique in nonlinear acoustics by detection of the second harmonic.

**4pPAg4. Detection of the nonlinearity evolution in concrete samples subject to quasi-static loadings.** Paola Antonaci (Politecnico di Torino, Corso Duca degli Abruzzi 24, 10129 Torino, Italy, paola.antonaci@polito.it), Pietro Bocca (Politecnico di Torino, Structural Engineering Department, 10129 Torino, Italy, pietro.bocca@polito.it), Caterina Bruno (Politecnico di Torino, Structural Engineering Department, 10129 Torino, Italy, caterina.bruno@polito.it), Antonio S. Gliozzi (Politecnico di Torino, Corso Duca degli Abruzzi 24, 10129 Torino, Italy, antonio.gliozzi@polito.it), Davide Masera (Politecnico di Torino, Structural Engineering Department, 10129 Torino, Italy, davide.masera@polito.it), Marco Scalerandi (Politecnico di Torino, Corso Duca degli Abruzzi 24, 10129 Torino, Italy, marco.scalerandi@infm.polito.it)

The signature of nonlinearity in the elastic response of a specimen to an impinging ultrasonic wave is usually determined through Fourier analysis, which provides low amplitude signals, often below noise level. We suggest here an alternative, based on the amplitude dependence of the response of the system. Our procedure is conceptually simple and easy to implement. In addition, it keeps simultaneously into account the nonlinear signature effects on phases, amplitudes and frequencies of the response. The procedure is described and used to analyse the variation of the nonlinearity in a concrete bar subject to quasi-static loadings of increasing amplitude. The sensitivity of the approach allows to distinguish the compaction phase (up to a load of 30% of the rupture loading) from a microdamage progression (up to a load of 60%) and the pre-rupture phases.

**4pPag5. Nondestructive tests of cylindrical steel samples using the ultrasonic projection method and the ultrasound transmission tomography method.** Krzysztof J. Opielinski (Wrocław University of Technology/Institute of Telecommunications, Teleinformatics and Acoustics, Wybrzeże Wyspiańskiego 27, 50-370 Wrocław, Poland, krzysztof.opielinski@pwr.wroc.pl), Tadeusz Gudra (Wrocław University of Technology/Institute of Telecommunications, Teleinformatics and Acoustics, Wybrzeże Wyspiańskiego 27, 50-370 Wrocław, Poland, Tadeusz.Gudra@pwr.wroc.pl)

The paper presents some methods of NDT of cylindrical steel samples by means of ultrasonic projection (UP) method and ultrasound transmission tomography (UTT) method. Some ways of scanning were proposed, using different measurement geometries and rendering possible the characterization and visualization of the inner structure of steel samples by projection and tomographic images of measured acoustic parameters. The measurements proposed in the paper allow us to obtain at the same time the distributions of mean values of a number of acoustic parameters characterizing the structure of samples: the mean amplitude of the ultrasonic wave after running through the sample, the mean runtime on the transmitter-receiver path, and the mean decrease of the ultrasonic wave frequency after running through the sample. These parameters measured in the tomographic scanning setup from many directions around the samples allow us to reconstruct the distributions of the local values of acoustic parameters such as respectively: the ultrasonic wave attenuation coefficient, the group velocity of the ultrasonic wave, the derivative of attenuation coefficient along the frequency. The reconstructed distributions of the local values of acoustic parameters render possible the imaging of the samples' internal structure cross-sections, each of the parameters characterizing different features of the structure.

**4pPag6. Use of point-source/point-receiver elastic waves in NDT-application.** Alexandr I. Korobov (Dept. of Acoustics, Physics Faculty, M.V. Lomonosov Moscow State University, Leninskie gory 1, 119991 Moscow, Russian Federation, akor@acs465a-1.phys.msu.ru), Natalya I. Odina (Dept. of Acoustics, Physics Faculty, M.V. Lomonosov Moscow State University, Leninskie gory 1, 119991 Moscow, Russian Federation, niodina@mail.ru), Anna V. Abramova (Dept. of Acoustics, Physics Faculty, M.V. Lomonosov Moscow State University, Leninskie gory 1, 119991 Moscow, Russian Federation, akor@acs465a.phys.msu.ru)

The ultrasonic automated experimental setup and technique for research of anisotropy of elastic properties of micro- and nanocrystalline metals and residual stresses are developed. Setup works in range of frequencies of 0.2-5 MHz. Accuracy of measurement of propagation time of elastic wave is equal to 0.2 ns, amplitude - 1%. Setup allows to make diagnostics of metals using volume, Raleigh and Lamb waves and to carry out scanning on linear and angular coordinates. Use of point source and receiver of acoustic waves and the high time resolution allows to carry out research of anisotropy and residual pressure with high spatial resolution. The x-y-coordinate device operated by personal computer, allows to carry out two-dimensional scanning of the sample by elastic waves with step of 10 micron. Experimental results of diagnostics of anisotropy of elastic properties in a number of microcrystalline constructional materials with residual stresses (alloys of aluminum and steel), and also a steel plate in the field of welded seam are presented. Significant anisotropy of elastic properties caused by these defects is revealed in investigated samples. The work is supported by RFBR.

**4pPag7. Acoustic attenuation in silicon and silicon oxide.** Anatolij P. Onanko (Taras Shevchenko Kyiv National University, physics faculty, prosp. Glushkova, 2/1, 03028 Kyiv, Ukraine, onanko@univ.kiev.ua), Oleg V. Lyashenko (Institute of Semiconductors Physics, NASU, Prospect Nauki 45, 03028 Kyiv, Ukraine, lyashenk@mail.univ.kiev.ua), Inna O. Lyashenko (Taras Shevchenko Kyiv National University, physics faculty, prosp. Glushkova, 2/1, 03028 Kyiv, Ukraine, lyashik@ukr.net), Yuriy A. Onanko (Taras Shevchenko Kyiv National University, physics faculty, prosp. Glushkova, 2/1, 03028 Kyiv, Ukraine, onanko@univ.kiev.ua)

In the present work the non-destructive acoustic attenuation (AA) method of the technological control the structure defects of semiconductor plates after various doses of the x-ray irradiation is developed. For measurement of temperature dependences of AA the method of resonant fluctuations of a plate on frequency 1.5 kHz was used at elastic deformation in vacuum. Measurements of temperature dependences AA in silicon plate a 460–470 micron thick after drawing a 600 nm layer of silicon oxide. Disk of a p-type silicon, doped B, orientation (100) with specific electroresistance 7.5 ohm-cm. The small maximums of AA were observed at temperature 346 K and 380 K. The main maximum of AA was observed at temperature 510 K. The affinity received by us of value of energy of activation 0.8 eV of AA at 510 K to energy of migration interstitial atoms 0.85 eV of silicon allows to assume the relaxation mechanism caused by reorientation interstitial atoms of silicon in dumbbell configurations.

**4pPag8. Nondestructive Evaluation of heterogeneous materials using acoustic emission and ultrasound.** Adil Faiz (ENSIM - LAUM, Université du Maine, rue Aristote, 72085 Le Mans, France, adil.faiz@univ-lemans.fr), Rachid El Guerjouma (ENSIM - LAUM, Université du Maine, rue Aristote, 72085 Le Mans, France, Rachid.El\_Guerjouma@univ-lemans.fr), Mouras Bentahar (Université du Maine LAUM, Lab. d'Acoustique Université du Maine, UMR CNRS 6613, 72085 Le Mans Cedex 9, France, mourad.bentahar@univ-lemans.)

Heterogeneous materials as Composites and concrete are advantageous as structural components in many applications. However, damage detection in such materials is difficult due to their heterogeneity and anisotropy. Furthermore, conventional non destructive technique as X-radiographic is mostly not very sensitive to early damage and very time consuming and expensive. The purpose of this contribution is to study the capabilities of several methods, non destructive and very sensitive for damage characterisation, as Acoustic Emission (AE) and ultrasonic for the structural health monitoring of heterogeneous materials as polymer based composite materials and concrete. These materials are instrumented by piezoelectric sensors in order to detect acoustic emission and to measure the ultrasonic velocity. The mechanisms of the damage events and their space-time localizations are identified from AE. Simultaneously, the longitudinal ultrasonic velocity is measured in situ by transmission through the composites thickness. The AE is very well correlated with the loss of stiffness determined from ultrasonic velocity measurements showing the potentiality of this combined approach for in situ structural health monitoring.

**4pPag9. In-situ measured Q- factor dependence from load in short and long time periods.** Nora A. Vilchinska (LAA-Latvian Acoustics Association, 3 Kurzemes pr, LV-1064 Riga, Latvia, vilcinska@hotmail.com)

Dynamic loaded large object is under research. Long time periods (8 years) measurements are made by variety loads: from quasi static till strong motion. Short time measurements start from quasi static and load goes step-by-step till strong motion- maximum loaded- and return in the same way to quasi static. This experiment lasts 60 minutes. An assessment of material quality factor (Q-factor) in places of measurements was made taking into account the absorbed and emitted energy. The Q-factor for energy is calculated from RMS response spectra curves. The Q-factor and its changes in long time and short time experiments in some MPs are compared. The smaller is the Q-factor, the higher is the concentration of interior invisible cracks. Structural alterations, opening of fractures and their closure under load, and transitional processes are reflected in the spectra of emitted acoustic signals and in the nonlinearity of response Q-factor.

## Session 4pPAh

## Physical Acoustics: Acoustic Probes of Planetary Environments II (Poster Session)

Andi Petculescu, Cochair  
*University of Louisiana*

Martin Towner, Cochair  
*PSSRI*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pPAh1. An ultrasonic anemometer for Mars.** Colin Wilson (University of Oxford, Department of Atmospheric, Oceanic and Planetary Physics, OX1 3PU Oxford, UK, wilson@atmos.ox.ac.uk), David Hutchins (University of Warwick, School of Engineering, CV4 7AL Coventry, UK, D.A.Hutchins@warwick.ac.uk), Lee A. Davis (University of Warwick, School of Engineering, CV4 7AL Coventry, UK, lee.davis@warwick.ac.uk), Martin C. Towner (PSSRI, The Open University, Walton Hall, MK7 6AA Milton Keynes, UK, m.c.towner@open.ac.uk)

Ultrasonic anemometers are often used for studies of 3-D atmospheric turbulence on Earth, due to their robust calibration and fast operation ( $>10$  Hz). The same qualities make ultrasonic anemometry attractive for use on Mars, where similar atmospheric turbulence is found. The low density of Martian atmosphere - a hundredth that of the Earth's - is problematic, because of the large acoustic impedance mismatch between the atmosphere and piezoelectric transducers. One solution to this problem is to use piezoelectric transducers with a hornlike resonator bonded to their front surface. Another solution is to use a capacitive membrane transducer instead. We report on performance of various ultrasonic transducers in carbon dioxide Martian pressures, and present an instrument design capable of meeting the stringent mass, power, and environmental requirements of the European ExoMars lander.

**4pPAh2. Acoustic tomography of the internal wave-associated fluctuations in the lower atmosphere.** Igor Chunchuzov (Obukhov Institute of Atmospheric Physics, 3 Pyzhevskii Per., 119017 Moscow, Russian Federation, igor.chunchuzov@gmail.com), Sergey Kulichkov (Obukhov Institute of Atmospheric Physics, 3 Pyzhevskii Per., 119017 Moscow,

Russian Federation, snk@ifaran.ru), Vitaly Perepelkin (Obukhov Institute of Atmospheric Physics, 3 Pyzhevskii Per., 119017 Moscow, Russian Federation, vitaliper54@gmail.com), Astrid Ziemann (Leipzig Institute for Meteorology, Stephanstr. 3, 04103 Leipzig, Germany, ziemann@uni-leipzig.de), Klaus Arnold (Leipzig Institute for Meteorology, Stephanstr. 3, 04103 Leipzig, Germany, Arnold@uni-leipzig.de), Anke Kniffka (Leipzig Institute for Meteorology, Stephanstr. 3, 04103 Leipzig, Germany, kniffka@rz.uni-leipzig.de)

The two different schemes of acoustic tomography of the atmospheric boundary layer (ABL) were used in the field experiments conducted near Melpitz (Germany) and Zvenigorod (Russia). The mesoscale effective sound speed fluctuations (periods 1min-1h) averaged over different acoustic ray paths were retrieved from the fluctuations of sound travel time between sources and receivers. It was found that a major contribution to the retrieved fluctuations comes from the wind speed fluctuations. By using a coherence analysis of the retrieved and measured wind speed fluctuations in the spatially distanced points the wave like fluctuations with periods of 16-20min, 8-10min, 4-5min, 1-2min have been filtered, and their horizontal translation velocities and scales have been estimated. Similar periods were also found in the variations of the vertical turbulent fluxes of momentum and heat near ground. The mechanism of origination of these periods in the observed fluctuations is proposed. The effect of the wind shear variations induced by internal waves on the turbulence intensity was observed. This effect showed a substantial role played by internal waves in the origin of an intermittency of turbulence in the stably stratified ABL. This work was supported by RFBR, grants 06-05-64229, 05-05-64973, 07-05-91555.

## Session 4pPAi

**Physical Acoustics, Acoustical Oceanography, and Biomedical Ultrasound/Bioresponse to Vibration: Acoustically Activated Bubble Dynamics and Applications II (Poster Session)**

Erich Everbach, Cochair  
*Swarthmore College*

Joachim Holzfuss, Cochair  
*Institute of Applied Physics, TU Darmstadt*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

*Contributed Papers*

**4pPAi1. Sound propagation in viscoelastic pipe with liquid-bubble mixture.** Semyon Levitsky (Shamoon College of Engineering, Math. Department, Bialik/Basel Sts., 84100 Beer Sheva, Israel, levits@sce.ac.il), Rudolf Bergman (Shamoon College of Engineering, Math. Department, Bialik/Basel Sts., 84100 Beer Sheva, Israel, rudolf@sce.ac.il), Jehuda Haddad (Shamoon College of Engineering, Math. Department, Bialik/Basel Sts., 84100 Beer Sheva, Israel, jehuda@sce.ac.il)

Propagation of acoustic waves in thin-walled polymeric tube with viscous liquid is investigated. Dynamics of the tube - liquid interaction is studied within conjugated quasi-one-dimensional formulation; the tube material is supposed to follow linear viscoelastic model with appropriate choice of the compliance function. It is assumed that the liquid contains fine air bubbles; the concentration of free gas is supposed to be small. Compressibility of liquid in the wave in the presence of bubbles can be almost entirely attributed to compressibility of the gas phase; it is accounted for within dispersion equation for bubbly liquid. Both heat and viscous losses are included in the phase interaction description at the liquid-gas interface. The resulting dispersion equation for the waveguide with liquid-gas mixture is studied in the long-wave range, where sound length is larger from the waveguide diameter. Results of simulations illustrate strong influence of the pipe mechanical properties and parameters of the gas phase on sound dispersion and attenuation.

**4pPAi2. Acoustic measurement of bubble size and position in an ink jet printhead.** Arjan Van Der Bos (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, j.a.vanderbos@tnw.utwente.nl), Roger Jeurissen (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, r.j.m.jeurissen@tnw.utwente.nl), Jos De Jong (Oce Technology, P.O. Box 101, 5900 MA Venlo, Netherlands, jos.dejong@oce.com), Detlef Lohse (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, d.lohse@utwente.nl), Michel Versluis (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, m.versluis@utwente.nl), Hans Reinten (Oce Technology, P.O. Box 101, 5900 MA Venlo, Netherlands, hans.reinten@oce.com), Marc Van Den Berg (Oce Technology, P.O. Box 101, 5900 MA Venlo, Netherlands, marc.van.den.berg@oce.com), Herman Wijshoff (Oce Technology, P.O. Box 101, 5900 MA Venlo, Netherlands, herman.wijshoff@oce.com), Richard Stevens (Physics of Fluids, University of Twente, P.O. Box 217, 7500 AE Enschede, Netherlands, r.j.a.m.stevens@student.utwente.nl)

An acoustic measurement method of the volume and position of a bubble in an ink jet print head, is presented. The system is driven by a piëzo actuator. The actuator is also used as a sensor by measuring the current through the piëzo. The method used to determine the volume and position of the bubble is based on a linear model of the investigated system. This model predicts the current for a given position and volume of the bubble. The inverse problem is to infer the position and volume of the bubble from the measured current through the piëzo actuator. The solution of the inverse problem is demonstrated. Thus, an acoustical measurement method of these properties is obtained. The results from the acoustical measurement method correspond closely with results from optical measurements. This indicates validity of the presented method.

**4pPAi3. Single bubble oscillations in viscoelastic media.** Anatoliy Strybulevych (Dept. of Physics and Astronomy, Univ. of Manitoba, Winnipeg, MB R3T 2N2, Canada, anatoliy@physics.umanitoba.ca), Valentin Leroy (Univ. of Manitoba, Dept. of Physics and Astronomy, Univ. of Manitoba, Winnipeg, MB R3T 2N2, Canada, valeroy77@yahoo.fr), Martin G. Scanlon (Univ. of Manitoba, Dept. of Food Science, 250 Ellis Bldg., Winnipeg, MB R3T 2N2, Canada, scanlon@cc.umanitoba.ca), John H. Page (Dept. of Physics and Astronomy, Univ. of Manitoba, Winnipeg, MB R3T 2N2, Canada, jhpage@cc.umanitoba.ca)

We report a new ultrasonic technique for determining the viscoelasticity of soft materials based on the oscillations of single bubbles injected into the material of interest. It is known that bubbles in a liquid act as strong acoustic scatterers that exhibit a low frequency resonance known as the Minnaert resonance. In the case of viscoelastic media, the complex frequency-dependent shear modulus causes the Minnaert frequency to be shifted to a higher value and leads to additional ultrasonic absorption. Therefore, both storage and loss shear moduli can be determined from the resonance frequency and the damping rate of the acoustic oscillations of a single bubble that has been injected into the sample. Experiments were performed on optically transparent commercial hair gel, agar gel, and PDMS rubber, allowing independent measurements of the bubble sizes to be made by an optical imaging technique. The acoustical properties of the samples were measured by sweeping the frequency of a continuous sinusoidal signal from 4 to 50 kHz. Because the Minnaert frequency is inversely proportional to the radius of the bubble, experiments on bubbles of different sizes enabled the frequency dependence of the complex shear moduli of the materials to be determined.



**4pPAi4. Experimental examination on the interactive force between two bubbles under ultrasound irradiation: Influence of the distance between two bubbles on bubble behavior.** Takaaki Fujikawa (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, dth0106@mail4.doshisha.ac.jp), Kenji Yoshida (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, etf1103@mail4.doshisha.ac.jp), Yoshiaki Watanabe (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, kwatanab@mail.doshisha.ac.jp)

The interactive force called the secondary Bjerknes force works among multiple bubbles under ultrasound irradiation, which results in the complicated behaviors of bubbles. In this study, we have experimentally examined the direction of the interactive force depending on the distance between two bubbles. When two bubbles vibrate individually, the direction of the interactive force depends on the phase difference between vibrations of these bubbles. In addition to this theory, considering the influence of the radiated acoustic wave from a bubble vibration on the other bubble vibration, Ida has pointed that this influence induces a change in the direction of the interactive force [M. Ida et al., Phys. Rev. E 67, 056617 (2003)]. In order to examine this theoretical prediction, the behaviors of both a free bubble and a bubble adhered to the polymer were observed under ultrasound irradiation, using a high-speed video camera. From the results, we found the reversal of the direction of the interactive force due to the variation in the distance between two bubbles. These innovative experimental results give an interesting point of view to understand the dynamics of multiple bubbles.

**4pPAi5. Destruction of the gas-filled capsule using effects of the collapsing bubble near the capsule.** Jun Miyabe (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, dth0144@mail4.doshisha.ac.jp), Kenji Yoshida (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, etf1103@mail4.doshisha.ac.jp), Daisuke Koyama (Tokyo Institute of Technology, 4259 Nagatsuda, 226-8503 Yokohama, Japan, dkoyama@sonic.pi.titech.ac.jp), Yoshiaki Watanabe (Faculty of Engineering, Doshisha Univ., 1-3 Miyakodani Tatara, 610-0321 Kyotanabe, Japan, kwatanab@mail.doshisha.ac.jp)

A new technique to destroy the gas-filled capsule is proposed, making use of the collapsing bubble near the capsule in the ultrasound field. At the moment of collapses under the ultrasound irradiation, the bubble causes flow and acoustic wave radiation. These phenomena are expected to assist the destruction of the capsule. We observed the effect of the bubble on this destruction, using a high-speed video camera which enables the real-time imaging. In this experiment, we use gas-filled capsule made of polyvinyl chloride. When no bubble was near the capsule at sound pressure of 60kPa, the capsule showed no vibration. However, in the presence of neighboring bubble, the capsule showed the destruction behavior of the deformation and the emission of internal gas. At higher sound pressure such as over 100kPa,

the capsule showed above destruction behavior even if the bubble was not located near the capsule. In the presence of the bubble, however, the capsule was destructed in a shorter time. These experimental results give significant knowledge to a technique manipulating the gas-filled capsule destruction in the drug delivery system.

**4pPAi6. Acoustic transmission through one plane of bubbles.** Valentin Leroy (Univ. of Manitoba, Dept. of Physics and Astronomy, Univ. of Manitoba, Winnipeg, MB R3T 2N2, Canada, valeroy77@yahoo.fr), Anatoliy Strybulevych (Dept. of Physics and Astronomy, Univ. of Manitoba, Winnipeg, MB R3T 2N2, Canada, anatoliy@physics.umanitoba.ca), Martin G. Scanlon (Univ. of Manitoba, Dept. of Food Science, 250 Ellis Bldg., Winnipeg, MB R3T 2N2, Canada, scanlon@cc.umanitoba.ca), John H. Page (Dept. of Physics and Astronomy, Univ. of Manitoba, Winnipeg, MB R3T 2N2, Canada, jhpage@cc.umanitoba.ca)

We measured the transmission of ultrasonic waves through one layer of bubbles, for frequencies ranging from 30 to 250 kHz. The layer was a true 2D structure obtained by injecting very monodisperse bubbles (with radius  $r=100\text{ }\mu\text{m}$ ) into a yield-stress polymer gel. Even for layer with a low concentration of bubbles (areal fraction,  $n\pi r^2$ , of 10-20%, where  $n$  is the number of bubbles per unit area), the transmission was found to be significantly reduced by the presence of bubbles (-20 to -50 dB) and showed a sharp minimum at a particular frequency. Interestingly, this frequency did not correspond to the individual Minnaert resonance of the bubbles, but depended on the concentration, which we interpret as an indication of strong coupling between the bubbles. We propose a simple model, based on a self-consistent relation, which takes into account the coupling between the bubbles and gives good agreement with the measured transmission.

**4pPAi7. Nonlinear ultrasonic waves in water-air mixtures.** Christian Vanhille (Universidad Rey Juan Carlos, Tulipán, s/n., 28933 Móstoles, Madrid, Spain, christian.vanhille@urjc.es), Cleofé Campos-Pozuelo (Instituto de Acústica, CSIC, Serrano, 144, 28006 Madrid, Spain, ccampos@ia.cetef.csic.es)

In this paper we present some features of nonlinear ultrasonic waves in water-air mixtures. This analysis is based on the coupling of the linear wave equation to the bubble equation in a volume formulation. The system is solved via the development of a numerical model (SMOW-BL code). The main restrictions of the model are: the nonlinear behaviour comes exclusively from the bubble vibration, all the bubbles have the same size, air inside the bubbles is adiabatic. Continuous waves at low ultrasonic frequency and wide band pulses propagation in an open domain are analysed. Results obtained by considering several bubbly layers in water are given. The existence of soliton in a water-air mixture is demonstrated. Some results corresponding to standing waves are also presented. A model which allows us to consider the self-generation of air bubbles in water, i.e., cavitation, is proposed.

## Session 4pPAj

## Physical Acoustics: Ultrafast Acoustics II (Poster Session)

David Hurley, Cochair

*Materials Characterization Department, Idaho National Laboratory*

Clément Rossignol, Cochair

*LMP, UMR CNRS 5469, Université Bordeaux I*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pPAj1. Complete characterization in thin film using picosecond ultrasonics and nanostructured transducer.** Pierre-Adrien Mante (IEMN-CNRS, Cité Scientifique - Avenue Poincaré, BP 60069, 59652 Villeneuve d'Ascq Cedex, France, pierre-adrien.mante@isen.fr), Jean-François Robillard (IEMN-CNRS, Cité Scientifique - Avenue Poincaré, BP 60069, 59652 Villeneuve d'Ascq Cedex, France, jean-francois.robillard@isen.fr), Arnaud Devos (IEMN-CNRS, Cité Scientifique - Avenue Poincaré, BP 60069, 59652 Villeneuve d'Ascq Cedex, France, arnaud.devos@isen.fr), Isabelle Roch-Jeune (IEMN-CNRS, Cité Scientifique - Avenue Poincaré, BP 60069, 59652 Villeneuve d'Ascq Cedex, France, isabelle.roch-jeune@iemn.univ-lille1.fr)

Picosecond ultrasonics is an efficient method to excite and detect vibrations within a thin film. A strong optical pulse warms a material surface, which leads to the creation of an acoustic wave propagating at the sound velocity. The waves propagation modifies the optical properties of the material that can be detected by a second time-shifted optical pulse. With an usual metallic transducer, only longitudinal waves can be generated. The use of this technique on a nanostructured transducer adds in-plane propagating waves. In the case of an isotropic medium, we have access to all acoustic properties. In order to confirm this statement, we realized and studied 2D lattices of metallic nanostructures. We used e-beam lithography to obtain defect free lattices of aluminum nanocubes. Both cubes width and lattice parameters were chosen to optimize the properties measurements [1]. This nanostructuration allows us to get information about longitudinal and in-plane waves, in the gigahertz frequency range. The experiments were carried out with an aluminum lattice on a 600nm-thick silica film. This method, demonstrated on a well-known material, is suitable for any thin film. [1] J.-F. Robillard, A. Devos and I. Roch-Jeune "Time-resolved vibrations of two-dimensional hypersonic phononic crystals", Phys. Rev. B. 76, 092301 (2007)

**4pPAj2. Generation of acoustical phonons by femtosecond laser pulses in GaAs in the presence of external electric field.** Philippe Babilotte (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, philippe.babilotte.etu@univ-lemans.fr), Pascal Ruello (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, pascal.ruello@univ-lemans.fr), Denis Mounier (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, denis.mounier@univ-lemans.fr), Daniel Pugliese (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans Cedex 09, France, Daniel.Pugliese@univ-lemans.fr), Mathieu Edely (LPEC/UMR 6087/CNRS/Université du Maine, Avenue Olivier

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Experimental results on the generation and the detection by fs laser pulses of the acoustical phonons at frequencies from tens to hundreds GHz in GaAs in the presence of external electric field are presented. The influence of the magnitude and the direction of the applied electric field on the parameters of ps ultrasound is investigated. Results obtained in GaAs and low-temperature GaAs are compared. The experimental opportunities to discriminate the acoustical phonons produced by laser-induced inverse piezoelectrical effect and the acoustical phonons due to the thermoelastic mechanism and to the mechanism of electron-phonon deformation potential are discussed. This study was supported by ANR project BLAN06-3-136284.

**4pPAj3. Temperature Dependence of Elastic Constant Measurements on Thin Films by Picosecond Ultrasonics.** Patrick Emery (IEMN-CNRS, Cité Scientifique - Avenue Poincaré, BP 60069, 59652 Villeneuve d'Ascq Cedex, France, patrick.emery@isen.fr), Arnaud Devos (IEMN-CNRS, Cité Scientifique - Avenue Poincaré, BP 60069, 59652 Villeneuve d'Ascq Cedex, France, arnaud.devos@isen.fr)

The temperature dependence of mechanical parameters is well-known for bulk materials. New methods have to be developed to access such characteristics on thin films. These measurements are needed for understanding the temperature behavior of acoustic components in microelectronics, as for Bulk Acoustic Wave (BAW) resonators used in wireless communication. A BAW resonator uses the thickness mode resonance of a piezoelectric layer (Aluminium Nitride). In the radio-frequency range, a BAW resonator is a complex stack of various materials in thin film. The temperature dependences measurements offer a way to regulate the performances' drift induced by the warming of the device and to design temperature-independent components. Here, we present a method based on a variable temperature picosecond ultrasonics setup. The procedure is first validated on silica, then applied on various BAW technology materials (AlN, Mo, SiN, W).

**4pPAj4. Photothermal and photoacoustic imaging by ultrafast optical sampling.** Stefan Dilhaire (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, stefan.dilhaire@u-bordeaux1.fr), Jean-Michel Rampnoux (LMP, UMR CNRS 5469,

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We describe a new ultrafast imaging technique based on the use of two new generation Ytterbium lasers emitting at 1030 nm at 50 MHz repetition rate. Ultrafast acquisition is achieved by slightly shifting the repetition rate of the "pump" and the "probe" beams. In that conditions a single shot response is acquired in 1ms that allows sweeping the surface of the device or the material and obtain movies of the reflectivity field of the surface. This technique allows filming the reflectance changes of a material at very high speed typically 1 Tera frame per second during 20ns. We will show applications of this technique in acoustic imaging of surface waves and non destructive evaluation of microelectronic materials.

**4pPAj5. The role of coherent phonons in the vibronic laser.** Wojciech Gadowski (University of Warsaw, Department of Chemistry, Laboratory of Physicochemistry of Dielectrics and Magnetism, ul. Żwirki i Wigury 101, 01-905 Warsaw, Poland, gado@chem.uw.edu.pl), Bożena Gadowska (University of Warsaw, Department of Chemistry, Laboratory of Physicochemistry of Dielectrics and Magnetism, ul. Żwirki i Wigury 101, 01-905 Warsaw, Poland, bogad@chem.uw.edu.pl)

Herewith we present the quantum theory of the vibronic laser, which is based on the crucial role played by the host crystal phonons in the laser dynamics. In the solid state transition-metal ion laser the operation takes place between the vibronically broadened electronic levels of the gain center, thus the laser action is accompanied by the creation or annihilation of the host lattice phonons. The nonequilibrium coherent phonons, produced in the process of the nonradiative energy transition from the photoexcited impurity ion, pump the energy level, from which the laser action takes place.

One can say that the laser photons are produced at the cost of lattice phonons. This is the reason why the coherent phonons strongly influence the character of the laser action and govern the complicated laser dynamics. In the laser equations, derived by us, the number of coherent phonons is treated as one of dynamical variables coupled with the number of photons. In solution we observe the effect of energy pulling between photons and phonons. Phonons follow all forms of the photon dynamics including self-pulsations and chaotic dynamics.

**4pPAj6. Picosecond ultrasonics signal in biological materials: Comparison between predictions and experiments.** Mathieu Ducousso (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, m.ducousso@lmp.u-bordeaux1.fr), Clément Rossignol (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, c.rossignol@lmp.u-bordeaux1.fr), Bertrand Audoin (LMP, UMR CNRS 5469, Université Bordeaux I, 351, cours de la Libération, 33405 Talence, France, b.audoin@lmp.u-bordeaux1.fr), Fabien Guillemot (INSERM U577, Université Victor Segalen Bordeaux 2, 146, rue Léo-Saignat, Case 45, 33076 Talence, France, fabien.guillemot@bordeaux.inserm.fr), Marie-Christine Durrieu (INSERM U577, Université Victor Segalen Bordeaux 2, 146, rue Léo-Saignat, Case 45, 33076 Talence, France, durrieumc@bordeaux.inserm.fr)

Picosecond ultrasonics is a non-destructive method for measuring mechanical properties such as velocity or stiffness coefficients for nanometric materials [1]. This technique uses femtosecond laser pulses for generating and detecting acoustics waves from GHz to THz. Its resolution is about nanometers in depth and a few micrometers laterally. For transparent materials it allows generating the so-called Brillouin oscillations, which frequencies are determined by the material sound velocity and the light beam wavelength. In this paper this technique is applied to biological cells. Measurements suggest promising perspectives for the imaging inside a single living cell; frequency content is from 5 to 20 GHz. A theoretical model based on Fourier heat and acoustic wave equations has been developed. These equations include heat diffusion and acoustic propagation respectively. It permits numerical simulations in time domain. First experimental results on animal and vegetal cells are presented and confronted with these calculated waveforms. [1] C. Thomsen, H. T. Grahn, H. J. Maris, J. Tauc, Phys. Rev. B 34, 4129, 1986

## Session 4pPAk

## Physical Acoustics: Thermoacoustics II (Poster Session)

Steven Garrett, Cochair  
Penn State University

Philippe Blanc-Benon, Cochair  
Ecole Centrale de Lyon

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pPAk1. Experimental evaluation of the wavenumber in stacked screen regenerators.** Yuki Ueda (Tokyo University of Agriculture and Technology, Nakacho 2-24-16, 187-8588 Koganei, Tokyo, Japan, uedayuki@cc.tuat.ac.jp)

The experimental method to evaluate the wavenumber of the acoustic wave propagating in pores media is proposed. The method was applied to four types of stacks composed of square channels and seven types of regenerators consisting of stacked mesh screens. The experimental results of the stacks agreed well with the theoretical results. The experimentally obtained wavenumber in the stacked screen regenerators were found to be similar to that in circular channels. However, it was found that they depend on the types of regenerators. Based on the dependence on the regenerator types of the evaluated wavenumber, the effective radius in the stacked screen regenerators was addressed.

**4pPAk2. A thermoacoustic device for sound reproduction.** Fotios Kontomichos (Audio and Acoustic Technology Group, Wire Communications Laboratory, Electrical Engineering & Computer Technology Department, University of Patras, 26500 Patras, Greece, fotkon@wcl.ee.upatras.gr), Alexandros Koutsoubas (University of Patras Dept. of Physics, Rio, 26500 Patras, Greece, alexandk@physics.upatras.gr), John Mourjopoulos (Audio and Acoustic Technology Group, Wire Communications Laboratory, Electrical Engineering & Computer Technology Department, University of Patras, 26500 Patras, Greece, mourjop@wcl.ee.upatras.gr), Nikolaos Spiliopoulos (University of Patras Dept. of Physics, Rio, 26500 Patras, Greece, nspiliop@physics.upatras.gr), Alexandros Vradis (University of Patras Dept. of Physics, Rio, 26500 Patras, Greece, vradis@physics.upatras.gr)

Many current research efforts focus on alternative electroacoustic transduction devices having no moving parts, in order to achieve sufficient audio performance from compact solid state devices. Thermoacoustic loudspeakers are transducers based on the conversion of A/C current signals to thermal energy, causing a local fluctuation of air pressure which generates acoustic waves. A thermoacoustic actuator does not involve any movement of solid components in order to generate an acoustic wave and it is based on a mechanism of a "virtual" piston produced by vibrating air molecules via alternating heat transfer to the medium. This work examines such novel and alternative audio transduction technologies based on a novel hybrid thermoacoustic transducer prototype which was developed at the University of Patras through the cooperation between Audio and Acoustic Technology Group and Solid State Physics Laboratory. This hybrid solid state device without moving parts is based on the thermoacoustic method of sound re-

production and preliminary measurements of its performance are presented. The theoretical principles of these systems are also simulated, resulting into comparisons with the measured performance of the prototype.

**4pPAk3. Numerical modelling of acoustic streaming in resonators.** Abdenour Boufermel (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, boufermel@hotmail.fr), Nicolas Joly (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, nicolas.joly@univ-lemans.fr), Pierrick Lotton (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, Pierrick.Lotton@univ-lemans.fr)

The acoustic wave propagation in thermoviscous fluid can generate slow phenomena, such as streaming and thermoacoustic effects, by nonlinear processes localised mainly in the viscous and thermal boundary layers. The model presented consists in computing numerically by the finite element method, (i) first, the harmonic solution for linear acoustics in thermoviscous fluid including the boundary layers, and (ii) second, the unsteady solution for the acoustic streaming and heat transfer. The model formulation is based on the mass and momentum conservation equations for the streaming, and the energy conservation equation for heat transfer. The streaming is presented as a standard form of an incompressible flow for velocity vector for mass transport, where the nonlinear effects of acoustics are considered as excitation forces for the streaming and sources for heat transfer. As the performance of thermoacoustic devices is limited by the convective heat transfer of the streaming, this study is suited for the optimisation of these systems. Another application is the development of microfluidic devices.

**4pPAk4. Experimental validations of a new thermoacoustic simulation software CRISTA.** Adrien Bétrancourt (LIMSI-CNRS, BP 133, F-91403 Orsay, France, adrien.betrancourt@limsi.fr), Thierry Le Pollès (Hekyom SARL, 2 rue Jean Rostand, 91400 Orsay, France, thlepollès@hekyom.com), Gérard Defresne (LIMSI-CNRS, BP 133, F-91403 Orsay, France, defresne@limsi.fr), Diana Baltean-Carlès (LIMSI-CNRS, BP 133, F-91403 Orsay, France, baltean@limsi.fr), Patxi Duthil (Institut de Physique Nucléaire d'Orsay, 15 rue Georges Clémenceau, 91400 Orsay, France, duthil@ipno.in2p3.fr), Jean-Pierre Thermeau (Institut de Physique Nucléaire d'Orsay, 15 rue Georges Clémenceau, 91400 Orsay, France, thermeau@ipno.in2p3.fr), Maurice-Xavier François (LIMSI-CNRS, BP 133, F-91403 Orsay, France, mxf@limsi.fr)

A new simulation software CRISTA has been developed at LIMSI-CNRS. It is based on the Rott's equations approximation. It computes all thermal and acoustic parameters of a given thermoacoustic device whose geometry is previously designed with another program TADESIGN. To realize



the simulation, the user needs only to define a drive ratio at some point of the system and the heat exchanger temperatures. Note that for a prime mover the hot heat exchanger temperature is a simulation result. Every converged solution guarantees the physical principles. Moreover, CRISTA allows computing the quality factor of the resonator. The experimental validations have been successfully performed on different devices coupled to the same prime mover: a simple RLC load, an acoustic amplifier, a pulse tube refrigerator and a lumped boost pulse tube refrigerator.

#### 4pPAk5. Numerical simulation of a thermoacoustic wave amplification.

Omar Hireche (LIMSI-CNRS, BP 133, F-91403 Orsay, France, hireche@lmsi.fr), Catherine Weisman (LIMSI-CNRS, BP 133, F-91403 Orsay, France, weisman@lmsi.fr), Diana Baltean-Carlès (LIMSI-CNRS, BP 133, F-91403 Orsay, France, baltean@lmsi.fr), Luc Bauwens (University of Calgary, Department of Mechanical and Manufacturing Engineering, 2500 University Drive NW, Calgary, AB AB T2N 1N4, Canada, bauwens@ucalgary.ca), Maurice-Xavier François (LIMSI-CNRS, BP 133, F-91403 Orsay, France, mxf@lmsi.fr), Patrick Le Quere (LIMSI-CNRS, BP 133, F-91403 Orsay, France, plq@lmsi.fr)

We performed a numerical study of the thermal and physical phenomena occurring in thermoacoustic wave generators. The goal of the simulation is to predict the amplification due to thermoacoustics of a wave initially of small amplitude. Therefore, we focus on the stack and the two heat exchangers, which we call the active cell, which is acoustically compact. The resonator area is split into two parts: the active cell, in which heat transfer takes place, and a resonator, in which the flow is acoustic. The flow in the two-dimensional active cell can be approximated as a low Mach number viscous, conductive flow, subjected to spatially uniform pressure fluctuations. This model is formally derived using asymptotic expansions in terms of Mach number. The focus here is heat transfer between two successive stack plates. The two-dimensional time-dependent problem resulting from this model is solved numerically. Outside the active cell, flow in the resonator is described by a reversible acoustic one-dimensional model. The coupling between the two zones is obtained by matching the velocity at the interface. The acoustics in the resonance tube can be solved using the d'Alembert solution, relating velocities at the interface to velocity values at an earlier time.

#### 4pPAk6. Green chillies: a practical thermoacoustic refrigerator in day-to-day use since February 2007.

Philip Spoor (CFIC-Qdrive, 302 Tenth St., Troy, NY 12180, USA, pspoor@cficinc.com)

CFIC, Inc. recently designed and built a thermoacoustic food refrigerator for the U.S. Army's Combat Feeding program (based at the Natick Soldier Center in Natick, MA). The Army wants a more combat-hardy alternative to standard vapor-compression devices, which have many leak-prone braze joints. However, the thermoacoustic "Army Fridge" has generated increasing commercial and public interest as the search for environmentally benign alternatives to conventional refrigeration has intensified. The Army fridge uses helium as its working fluid, and it has no pumps or circulating fluids, only heat pipes and fans for heat exchange with the air. Unlike all other thermoacoustic prototypes known to us, this device is expressly designed like a product, with all the thermoacoustic hardware and controls confined to a relatively small enclosure on top of a large (17 cubic foot) cabinet, and operation accomplished by a single power switch and a thermostat. In early 2007 the Army fridge went into long-term testing at CFIC, Inc. as our auxiliary refrigerator. By March 2007 it will have surpassed 1 year (8760 hours) of continuous running. We will present details of construction, performance history, and recommendations for higher efficiency and lower cost.

#### 4pPAk7. Effect of an obstacle on Rayleigh acoustic streaming cells.

Solenn Moreau (Laboratoire d'Etudes Aérodynamiques (LEA), Université de Poitiers, ENSMA, CNRS, Bat K, 40 avenue du recteur Pineau, 86022 Poitiers, France, solenn.moreau@lea.univ-poitiers.fr), Helene Baillet (Laboratoire d'Etudes Aérodynamiques (LEA), Université de Poitiers, ENSMA, CNRS, Bat K, 40 avenue du recteur Pineau, 86022

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Acoustic streaming has harmful consequences on thermoacoustic machines behaviour because of the associated heat transfers. A preliminary study was carried out in order to study the effect of an obstacle on the Rayleigh cells to help in understanding the role of such phenomena in thermoacoustic machines. An obstacle was introduced in a half-wavelength cylindrical wave guide to study its effects on acoustic streaming. The obstacle was placed at various positions along the wave guide axis and experiments were carried out at various acoustic levels. The axial streaming velocity was measured using Laser Doppler Velocimetry (LDV). It was observed that adding an obstacle in the streaming pattern modifies the latter and that new streaming vortices appear in the vicinity of the obstacle. When the obstacle position approaches a maximum of the Rayleigh streaming velocity the number and the amplitude of acoustic streaming vortices at the ends of the obstacle increase. Similar tendencies were observed when the acoustic velocity amplitude was increased. Because streaming in the vicinity of the obstacle end is complex and has a high amplitude, heat effects can be expected to be important and complex at the ends of the thermoacoustic stack where heat exchangers are located.

#### 4pPAk8. Experimental study of the thermoacoustic effect using infrared thermography.

Vincent Feuillet (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, vincent.feuillet@univ-lemans.fr), Guillaume Penelet (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, guillaume.penelet@univ-lemans.fr), Pierrick Lotton (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, Pierrick.Lotton@univ-lemans.fr), Lionel Camberlein (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, lionel.camberlein@univ-lemans.fr)

The thermal metrology used to study experimentally the thermoacoustic effect is almost always based on thermoelectric junctions. This kind of measurement is intrusive and provides a few information in space. This paper presents an experimental set-up where temperature measurement on the thermoacoustic stack is acquired by infrared thermography. This measurement provides more information in space and time to study complex physical phenomena, such as heat transfer through both ends of the stack, non linear edge effects, or optimal spacing between stack and heat exchangers. The experimental set-up consists of a half-wavelength resonator with a square cross section closed by a rigid wall at one end and coupled to an electrodynamic loudspeaker at the other end. The thermoacoustic core is either a single plate or a stack of parallel plates made of Kapton. Temperature measurements are carried out along the stack by the use of an infrared camera. The acoustic pressure is measured by a microphone flush-mounted on the wall at the exit of the resonator. First results show the essential role played by the edge effects and the heat generated by viscous losses.

#### 4pPAk9. LDV measurements of acoustic streaming in a traveling wave, closed-loop resonator.

Cyril Desjoux (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, cyril.desjoux@univ-lemans.fr), Pierrick Lotton (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, Pierrick.Lotton@univ-lemans.fr), Guillaume Penelet (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, guillaume.penelet@univ-lemans.fr), James Blondeau (Laboratoire d'Acoustique de l'Université du Maine, Avenue Olivier Messiaen, 72085 Le Mans, France, james.blondeau@univ-lemans.fr)

The first part of this work deals with the study of an annular acoustic resonator, where the acoustic field is controlled by two loud-speakers in order to generate a purely traveling wave. The results of a complete electroacoustic modeling of this device, where the acoustic field is accurately defined, are compared to the experimental results. The second part deals with the characterization of various non-linear effects occurring in such a device. The generation of acoustic streaming is especially investigated in our study.

Laser Doppler Velocimetry measurements are performed in order to characterize both first order (acoustic) and second order (acoustic streaming) velocity fields. Works are now in progress in order to improve accuracy of streaming measurement and to compare the experimental data to the available theories. This study should contribute to enhance the designing of thermoacoustic devices and should be also useful for its potential applications in microfluidics, especially for the development of micro-pumps and micro-mixers.

**4pPAk10. Weakly non-linear thermoacoustics for general porous media.** Peter In 't Panhuis (Eindhoven University of Technology, P.O. Box 513, 5600 MB Eindhoven, Netherlands, p.h.m.w.panhuis@tue.nl), Sjoerd Rienstra (Eindhoven University of Technology, P.O. Box 513, 5600 MB Eindhoven, Netherlands, s.w.rienstra@tue.nl), Han Slot (Eindhoven University of Technology, P.O. Box 513, 5600 MB Eindhoven, Netherlands, j.j.m.slot@tue.nl), Jos Zeegers (Eindhoven University of Technology, Den Dolech 2, 5612 AZ Eindhoven, Netherlands, j.c.h.zeegers@tue.nl)

A weakly nonlinear theory for thermoacoustics, including acoustic streaming, a temperature-dependent viscosity and slowly varying pores with arbitrarily shaped cross-sections, has been constructed by systematically applying dimensional analysis and small-parameter asymptotics. In this way a set of equations for the acoustic and streaming variables can be derived. For some simple cases explicit solutions can be found, such as the short-stack approximation, but for the more advanced applications we have to resort to a numerical solution. The theory has been implemented both for standing-wave and traveling-wave applications. For the case of a standing-wave system we have compared our computations with experimental data and found a remarkable agreement.

**4pPAk11. Viscous and thermal effects in acoustic radiation problems.** Husnain Inayat Hussain (INSA de Lyon - LVA, Bâtiment St. Exupéry, 25 bis avenue Jean Capelle, F-69621 Villeurbanne Cedex, France, husnain.inayat-hussain@insa-lyon.fr), Jean-Louis Guyader (INSA de Lyon - LVA, Bâtiment St. Exupéry, 25 bis avenue Jean Capelle, F-69621 Villeurbanne Cedex, France, jean-louis.guyader@insa-lyon.fr)

The thermal and viscous effects are generally neglected in acoustic radiation problems. In the present work we observe if these effects have any significance on acoustic radiation characteristics. Several researches have been materialised in this respect, notably, Bruneau, Beltman et al. These models study factors like inertial and viscous terms for example and then keep some while neglect others based on their significance. Our approach is quite general and we take all the factors into consideration. Our model not only deals with the thermal and viscous effects but compressibility and all other terms are taken into consideration. A system of differential equations issuing from conservation principles is linearized. Next harmonic dependence is assumed and a linear system is obtained. This linear system generates different waves among which only those are kept, which follow the sommerfeld condition. These waves produce an equal number of linear systems, and consequently; pressure, temperature, and the normal and tangential acoustic velocities are determined, using the boundary conditions of no slip, isothermal wall and matching normal acoustic velocity. Our aim is to apply these findings to a viscothermal fluid within a double wall to see if acoustic transmission is improved.

**4pPAk12. Physical mechanism and theoretical model of thermoacoustic heat exchangers.** Ercang Luo (Technical Institute of Physics and Chemistry, CAS, Beiyitiao Rd., Zhongguancun St., P.O.Box 2711, 100080 Beijing, China, Ecluo@cl.cryo.ac.cn), Bo Gao (Graduate University of Chinese Academy of Sciences, Zhongguancun St., 100049 Beijing, China, gaobozaihit@163.com)

Unlike a regenerator, there are both oscillating heat exchange flux (first order) and non-zero time-averaged heat exchange flux (second order) in the hot and cold end heat exchangers of thermoacoustic systems. The non-zero time-averaged heat flux just reflects the net heat exchange between the working substance and external heat sinks, which is more important and interesting for design. In this paper, the physical mechanism of oscillating flow heat exchanger is first analyzed. Based on understanding of the heat transfer mechanism, the theoretical model of heat transfer is then developed; in this part, the key of the problem is to obtain transversal distribution of the second order time-averaged temperature,  $T_{20}(y)$ . Eventually, an analytical expression of Nusselt number for the thermoacoustic heat exchangers is obtained under the assumption of laminar flow.

**Session 4pPPa****Psychological and Physiological Acoustics: Role of Temporal Fine Structure in Speech and Non-Speech Perception for Normal and Hearing-Impaired People I**

Christian Lorenzi, Cochair

*Univ Paris Descartes, CNRS, Ecole Normale Supérieure, DEC, 29 rue d'Ulm, Paris, 75005, France*

Brian Moore, Cochair

*University of Cambridge, Department of Experimental Psychology, Downing Street, Cambridge, CB2 3EB, UK****Invited Papers*****2:00****4pPPa1. Role of temporal fine structure cues in speech intelligibility.** Christian Lorenzi (Univ Paris Descartes, CNRS, Ecole Normale Supérieure, DEC, 29 rue d'Ulm, 75005 Paris, France, [lorenzi@ens.fr](mailto:lorenzi@ens.fr))

A number of studies have investigated the role of two temporal features of the narrowband speech signals at the output of auditory filters in speech identification: temporal envelope (E) and temporal fine structure (TFS) cues. To assess the contribution of each feature to speech identification, speech stimuli were split into an array of contiguous analysis bands and processed using several techniques to remove, as far as possible, either E or TFS cues within each band. Overall, the outcome of these studies indicated that, after moderate to substantial training, high levels of speech identification measured in quiet could be obtained for normal-hearing listeners on the basis of E cues or TFS cues alone. The results obtained with TFS cues only may appear surprising, because it is generally considered that, at least for non-tonal languages, E cues carry most of the phonetic information required for speech identification in quiet whereas TFS cues play mainly a role in conveying the pitch cues required for the segregation of speech and background sounds. Further work assessing the extent to which TFS cues alone can convey useful linguistic information in addition to these pitch cues will be reviewed.

**2:20****4pPPa2. Role of temporal fine structure in speech perception.** Fan-Gang Zeng (University of California Irvine, 364 Med Surge II, Irvine, CA 92697, USA, [fzeng@uci.edu](mailto:fzeng@uci.edu))

Recent studies have shown that lack of access to the temporal fine structure cue is a major reason for the difficulty in speech perception in noise by hearing-impaired listeners. To further understand the role of temporal fine structure, we need to define the temporal fine structure and to delineate its relationship to the temporal envelope in both acoustical and perceptual domains. This talk will first examine the relationship between temporal envelope and temporal fine structure in signal processing terms and then relate it to speech production and perception. Acoustically, the temporal fine structure primarily contributes to changes in fundamental frequency, harmonics, and formant transition. Perceptually, while the temporal fine structure can contribute to speech intelligibility via the formant transition cue, it contributes to speech perception in noise by enhancing auditory objection formation rather than increasing speech intelligibility directly.

**2:40****4pPPa3. Temporal fine structure coding for pitch and speech perception.** Andrew J. Oxenham (University of Minnesota, Department of Psychology, 75 E. River Road, Elliott Hall N218, Minneapolis, MN 55455, USA, [oxenham@umn.edu](mailto:oxenham@umn.edu))

Temporal fine structure can be defined mathematically with relative ease. Understanding its perceptual importance, or even how it is coded in the peripheral auditory system, is another matter. This talk will review recent work from our lab that addresses some of these issues. The focus will be on the use and representation of temporal fine structure in complex pitch perception with spectrally resolved and unresolved components, and on the importance of temporal fine structure when using F0 information in speech to segregate target speech for competing talkers and other complex interfering sounds. [Supported by National Institutes of Health grant R01 DC 05216.]

**3:00****4pPPa4. The contribution of temporal fine structure information to the intelligibility of speech in noise.** Kathryn Hopkins (University of Cambridge, Department of Experimental Psychology, Downing Street, CB2 3EB Cambridge, UK, [kh311@cam.ac.uk](mailto:kh311@cam.ac.uk)), Brian Moore (University of Cambridge, Department of Experimental Psychology, Downing Street, CB2 3EB Cambridge, UK, [bcjm@cam.ac.uk](mailto:bcjm@cam.ac.uk))

Temporal fine structure (TFS) information in speech may be particularly useful when listening to a target in a background that contains temporal 'dips'. A change in TFS may allow identification of signal portions where the target-to-background ratio is high. This hypothesis was tested. Speech reception thresholds were measured with steady and an 8-Hz amplitude-modulated noise background for signals processed to contain variable amounts of TFS information. Signals were filtered into channels and channel signals for channel numbers above a 'cut off channel' (CO) were tone-vocoded to remove TFS information, while channel signals with channel numbers of CO and below were left unprocessed. Signals from all channels were combined. Five values of CO were tested for each noise type, with

ten normal-hearing subjects. Subjects benefited more from TFS information when listening in the modulated masker than the steady masker. For steady noise, addition of TFS information above 548 Hz did not improve performance, whereas for modulated noise, addition of TFS at high frequencies did improve performance. The greater benefit from TFS information when listening in modulated noise is consistent with the idea that TFS information is important for listening in the dips of a fluctuating masker.

3:20

**4pPPa5. Discrimination of stochastic patterns of frequency modulation relevant to speech perception.** Stanley Sheft (Parml Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Road, Chicago, IL 60626, USA, ssheft@luc.edu), Christian Lorenzi (Univ Paris Descartes, CNRS, Ecole Normale Supérieure, DEC, 29 rue d'Ulm, 75005 Paris, France, lorenzi@ens.fr)

Recent work has suggested involvement of temporal fine structure (TFS) information in speech perception, especially in adverse listening conditions. However, little is known regarding discrimination of pattern of TFS modulation, an ability that would underlie robust contribution to speech processing. The present study evaluated the ability to discriminate among stochastic patterns of frequency modulation (FM). Contrasting modulators were different samples of lowpass noise that shared a common bandwidth and resulted in the same maximum frequency excursion. Performance levels declined in an orderly manner with modulator bandwidth, similar to results obtained when lowpass noise is used to modulate amplitude (i.e., AM) rather than fine structure. Modeling, however, indicated that the similarity was not simply a result of FM-to-AM conversion at the output of auditory filters. In additional conditions, discrimination of stochastic FM was measured in the presence of wideband noise which in some cases was sinusoidally amplitude modulated at either 4 or 20 Hz. With both unmodulated and modulated maskers, little effect on performance was noted, even with a signal carrier level as low as 15-20 dB above detection threshold. Absence of a substantial masking effect indicates availability of TFS cues for speech perception in noise. [Work supported by NIH.]

3:40-5:20 Posters

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

### *Invited Paper*

5:20

**4pPPa6. The role of temporal fine structure in speech source segregation.** Joshua G. Bernstein (Walter Reed Army Medical Center, Army Audiology and Speech Center, 6900 Georgia Ave. NW, Washington, DC 20307-5001, USA, joshua.bernstein@amedd.army.mil), 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)

Normal-hearing (NH) listeners show better speech recognition when a stationary noise masker is replaced by an opposite-gender competing talker at the same signal-to-noise ratio (SNR). Hearing-impaired (HI) listeners often do not show this interfering-talker benefit (ITB). This may be due to a reduced ability to use temporal fine structure (TFS). Consistent with this idea, NH listeners also show little ITB when TFS is removed. We hypothesized that TFS underlies the ITB by providing source-segregation cues. To test this hypothesis, non-auditory segregation cues were introduced in the form of a video of the talker's face. Speech intelligibility was estimated in NH listeners as a function of SNR for sentences spoken by a female talker and masked by speech-spectrum shaped stationary noise or a single male talker. Target and masker were summed before processing by a 15-channel noise vocoder to remove TFS, and presented with or without accompanying video. Without video, listeners received little ITB, consistent with previous results. Auditory-visual conditions yielded as much as 9 dB of ITB, supporting the hypothesis that a diminished ability to perceptually segregate sources contributes to the lack of ITB in the absence of TFS. Similar results were obtained for unprocessed speech in HI listeners.

### *Contributed Paper*

5:40

**4pPPa7. Relative importance of E and TFS speech cues in low and high-frequency channels.** Marine Ardoint (Univ Paris Descartes, CNRS, Ecole Normale Supérieure, DEC, 29 rue d'Ulm, 75005 Paris, France, marine.ardoint@ens.fr), Christian Lorenzi (Univ Paris Descartes, CNRS, Ecole Normale Supérieure, DEC, 29 rue d'Ulm, 75005 Paris, France, lorenzi@ens.fr)

Previous studies have shown that perfect speech identification in quiet could be obtained for broadband speech processed using the Hilbert transform to preserve either temporal envelope (E) or temporal fine structure (TFS) cues only in narrow frequency bands. However, little is known regarding the respective contribution of each cue to speech intelligibility in

low and high frequency bands. The goal of the present study was to assess the contribution of E and TFS cues of the low- and high-frequency channels to consonant identification. Vowel-consonant-vowel stimuli were split into an array of 16 analysis bands spanning the range 80-8,020 Hz and processed using the Hilbert transform to keep either E or TFS cues only within each band. Identification scores were measured for low-pass and high-pass filtering of the processed stimuli for a group of normal-hearing listeners. Results will be discussed in light of phase-locking properties to E and TFS. In most mammals, phase-locking of auditory-nerve fibers begins to decline above 1 kHz and disappears above 4-5 kHz. TFS-coded speech intelligibility should therefore drop when stimuli are high-pass filtered above 1 kHz and be at chance level for high-pass filtering above 5 kHz. No such drop should be expected for E-coded speech.

4p THU. PM



6:00

**4pPPa8. The importance of temporal fine structure coding for speech perception in listeners with sensorineural hearing impairment as compared to normal hearing listeners.** Emily Buss (University of North Carolina, School of Medicine, 1115 Bioinformatics. Bldg., CB7070, 130 Mason Farm Rd., Chapel Hill, NC 27599, USA, ebuss@med.unc.edu), Joseph W. Hall (University of North Carolina, School of Medicine, 1115 Bioinformatics. Bldg., CB7070, 130 Mason Farm Rd., Chapel Hill, NC 27599, USA, jwh@med.unc.edu), John H. Grose (University of North Carolina, School of Medicine, 1115 Bioinformatics. Bldg., CB7070, 130 Mason Farm Rd., Chapel Hill, NC 27599, USA, jhg@med.unc.edu)

Physiological data suggest that the neural representation of temporal fine structure could benefit speech recognition in normal hearing listeners. Conversely, it has been hypothesized that deficits in the encoding of fine structure could play a role in the poor speech perception of patients with sensorineural hearing loss, particularly for speech presented at moderate to high levels and speech presented in background noise. Experiments on the psychoacoustical abilities of adults with hearing loss have shown a correlation between performance on tasks thought to rely on fine structure cues (e.g., low rate FM detection) and ability to understand speech or lowpass filtered speech, lending support to this hypothesis. This talk will review these data, as well as the mechanisms proposed to account for this result, and will consider the relation of these findings to experiments with normal-hearing listeners.

6:20

**4pPPa9. Fine-structure processing, frequency selectivity and speech perception in hearing-impaired listeners.** Olaf Strelcyk (Centre for applied hearing research, Technical University of Denmark, DTU, Bygn. 352, 2800 Lyngby, Denmark, os@oersted.dtu.dk), Torsten Dau (Centre for applied hearing research, Technical University of Denmark, DTU, Bygn. 352, 2800 Lyngby, Denmark, tda@oersted.dtu.dk)

Hearing-impaired people often experience great difficulty with speech communication when background noise is present, even if reduced audibility has been compensated for. Other impairment factors must be involved. In order to minimize confounding effects, the subjects participating in this study consisted of groups with homogeneous, symmetric audiograms. The perceptual listening experiments assessed the intelligibility of full-spectrum as well as low-pass filtered speech in the presence of stationary and fluctuating interferers, the individual's frequency selectivity and the integrity of temporal fine-structure processing. The latter was addressed in a binaural and a monaural experiment. In the binaural experiment, the lateralization threshold was measured for low-frequency tones with ongoing interaural phase delays. In the monaural experiment, detection thresholds for low-rate frequency modulation were obtained. In addition, these binaural and monaural thresholds were measured in a stationary background noise in order to assess the persistence of the fine-structure processing to interfering noise. Apart from elevated speech reception thresholds, the hearing impaired listeners showed poorer performance than the normally hearing in terms of frequency selectivity and fine-structure processing, despite normal audiometric thresholds at the test frequencies. However, the binaural fine-structure processing was not found to be particularly vulnerable to interfering noise in these listeners.

## Session 4pPPb

**Psychological and Physiological Acoustics: Role of Temporal Fine Structure in Speech and Non-Speech Perception for Normal and Hearing-Impaired People II (Poster Session)**

Christian Lorenzi, Cochair

*Univ Paris Descartes, CNRS, Ecole Normale Supérieure*

Brian Moore, Cochair

*University of Cambridge*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

*Contributed Papers*

**4pPPb1. Facilitation of intensity discrimination of short stimuli in noise occurs not far from a stimuli detection threshold.** Liudmila Rinskaya-Korsakova (N. N. Andreyev Acoustics Institute, Shvernika 4, 117036 Moscow, Russian Federation, lkrk@akin.ru)

Adaptation is a main property of sensory receptors. In a basis of adaptation are properties of peripheral receptors and central neuron. There is a question: what is the auditory adaptation mechanism to short high frequency stimuli? How the auditory system can trace stimulus amplitude changes? The probable answer is: stimuli are mixed with noise of different origin. Noise adapts hearing and improves discrimination of stimulus. Simulation researches of peripheral encoding were shown the amplitude structure of short stimulus acting in isolation or in noise could be preserved when intensity are not far from thresholds of fibers. For each stimulus level there is the certain noise level, when amplitude structure can be detected without losses. In this work the validity of these statements was checked up in auditory experiments. The thresholds of intensity discrimination (ID) for short stimulus presented in silence and in conditions of simultaneous and forward masking have been estimated. In a range of average stimulus intensity and for each stimulus level there is the certain noise level when ID is better in noise but not in silence. The ID facilitation is registered near to the stimulus detection thresholds and after adaptation of hearing by noise. Results of auditory researches correspond to the simulation results.

**4pPPb2. Intelligibility of temporal fine structure speech signals with restricted FM excursion.** Gaëtan Gilbert (Equipe audition LPP UMR CNRS 8158, 29, rue d'Ulm, 75005 Paris, France, gaetan.gilbert@ens.fr), Agnes Leger (Equipe audition LPP UMR CNRS 8158, 29, rue d'Ulm, 75005 Paris, France, legeragnes@gmail.com), Christian Lorenzi (Univ Paris Descartes, CNRS, Ecole Normale Supérieure, DEC, 29 rue d'Ulm, 75005 Paris, France, lorenzi@ens.fr)

The Hilbert transform is the most common demodulation technique to derive temporal fine structure (TFS) signals. However, for speech stimuli, the Hilbert transform generally leads to results that have no clear physical meaning, generating undesired artefacts; for instance, instantaneous frequency may vary well beyond the analysis filters bandwidth. This study examined the intelligibility in quiet of TFS-coded Vowel-Consonants-Vowel signals generated with a demodulation technique minimizing these artefacts. Speech items were passed through a 16 FIR filters (750th order) filterbank. A Greenwood mapping was used to set filters bandwidth between 80-8020 Hz (approximately 2 ERBs wide). A frequency-modulation function was extracted at the output of each filter, hard limited within the analysis bandwidth and lowpass filtered. Spectral cues were removed by equating the rms across bands. Identification scores ranged between chance level (6.25%)

with no improvement across sessions for the least experienced listeners and 50% correct for the most experienced listeners. Further experiments will investigate these between-listeners differences.

**4pPPb3. Hard times for the pitch of complex sounds in reverberant environments.** Mark Sayles (Centre for the Neural Basis of Hearing, The Physiological Laboratory, Downing Street, CB2 3EG Cambridge, UK, ms417@cam.ac.uk), Ian M. Winter (Centre for the Neural Basis of Hearing, The Physiological Laboratory, Downing Street, CB2 3EG Cambridge, UK, imw1001@cam.ac.uk)

Reverberation is present in most everyday listening environments, with sound reaching our ears directly from the source(s), and indirectly after reflecting from nearby surfaces. Reverberation smears spectral transitions through time and disrupts temporal envelope modulation in harmonic complex sounds such as voiced portions of speech. Envelope modulation can provide a cue to the pitch of complex sounds and is relied upon by hearing impaired listeners. We have examined the effects of reverberation on the temporal representation of the dynamic pitch of frequency-modulated harmonic complex sounds in the responses of single units from the ventral cochlear nucleus. Without reverberation most units provide a strong representation of the modulated pitch in their short-term interspike interval distributions. Relatively mild reverberation degrades the representation based on envelope modulation, whilst more severe reverberation removes it. The representation of pitch based on fine structure is more robust to the effects of reverberation; however, this representation is smeared in time by indirect sound energy. The effects of reverberation are critically dependent on the neuron's best-frequency, source-to-receiver distance, fundamental frequency and frequency modulation rate. We observe comparable results in human listeners using the same sounds.

**4pPPb4. Effect of speech rate on speech-on-speech masking.** Jing Chen (Dept. of Machine Intelligence, Speech and Hearing Research Center, 2 Science Building, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, chenjing@cis.pku.edu.cn), Xihong H. Wu (Dept. of Machine Intelligence, Speech and Hearing Research Center, 2 Science Building, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, wxh@cis.pku.edu.cn), Xuefei F. Zou (Dept. of Machine Intelligence, Speech and Hearing Research Center, 2 Science Building, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, zouxf@cis.pku.edu.cn), Zhiping P. Zhang (Dept. of Machine Intelligence, Speech and Hearing Research Center, 2 Science Building, Peking Univ., 5

Yeheyuan Road, Haidian District, 100871 Beijing, China, zhangzp@cis.pku.edu.cn), Lijuan J. Xu (Dept. of Psychology, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, lijuan179021@yahoo.com.cn), Mengyuan Y. Wang (Dept. of Psychology, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, motoluto@163.com), Liang Li (Dept. of Psychology, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, liangli@pku.edu.cn), Huisheng S. Chi (Dept. of Machine Intelligence, Speech and Hearing Research Center, 2 Science Building, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, chi@pku.edu.cn)

Target speech can be better recognized under speech-on-speech masking conditions if certain differences between target and masker (e.g. in loudness, pitch, location) can be used as cues for streaming. This study examined whether the speech rate can be used by listeners as a cue for unmasking target speech. The rate difference between target and masking speech was manipulated by changing the rate of masking speech using the Synchronized Overlap-Add Fixed Synthesis (SOLA-FS) algorithm, and consequently, the ratio of target speech to masking speech (the speech rate ratio, SRR) was quantified. Both target and masker speech were Chinese nonsense sentences and they were co-presented with the signal-to-masker ratio of -7 dB. The results show that speech recognition was significantly increased with the SRR increase from 1 to 1.5 or the SRR decreased from 1 to 0.5. Moreover, the unmasking effect of precedence-induced perceived spatial separation on target-speech recognition was increased monotonically with the increase of the SRR from 0.5 to 1.5. These results suggest that the speech rate is one of the factors influencing both energetic masking and informational masking of Chinese speech. [Supported by the NSFC 60435010; 60535030; 60605016; 30670704]

#### 4pPPb5. Characterizing the individual ear by the "Auditory Profile".

Wouter A. Dreschler (AMC, Clinical and Experimental Audiology, 1105 Amsterdam, Netherlands, w.a.dreschler@amc.uva.nl), Thamar E. Esch Van (AMC - Dept. of Clinical and Experimental Audiology, Meibergdreef 9, 1105AZ Amsterdam, Netherlands, t.e.vanesch@amc.nl), Birgitta Larsby (Linköping Universitet, Hus Origo Campus Valla, 581 83 Linköping, Sweden, birgitta.larsby@lnr.liu.se), Mathias Hallgren (Linköping Universitet, Hus Origo Campus Valla, 581 83 Linköping, Sweden, mathias.hallgren@lnr.liu.se), Mark E. Lutman (University of Southampton, University Road, Highfield / ISVR, SO17 1BJ Southampton, UK, mel@isvr.soton.ac.uk), Johannes Lyzenga (Vrije Universiteit Medical Center, Boelelaan 1117, 1081 HV Amsterdam, Netherlands, j.lyzenga@vumc.nl), M Vormann (Hoerzentrum Oldenburg, Hoerzentrum Oldenburg, 26129 Oldenburg, Germany, matthias.vormann@web.de), Birger Kollmeier (Universität Oldenburg, Medizinische Physik, Carl-von-Ossietzky Str. 9-11, 26111 Oldenburg, Germany, birger.kollmeier@uni-oldenburg.de)

This paper describes a new approach to auditory diagnostics, which is one of the central themes of the EU-project HEARCOM. For this purpose we defined a so-called "Auditory Profile" that can be assessed for each in-

dividual listener using a standardized battery of audiological tests that - in addition to the pure-tone audiogram - focus on loudness perception, frequency resolution, temporal acuity, speech perception, binaural functioning, listening effort, subjective hearing abilities, and cognition. For the sake of testing time only summary tests are included from each of these areas, but the broad approach of characterizing auditory communication problems by means of standardized test is expected to have an added value above traditional testing in understanding the reasons for poor speech reception. The Auditory profile may also be relevant in the field of auditory rehabilitation and for design of acoustical environments. The results of an international 5-center study (in 4 countries and in 4 languages) will be presented and the relevance of a broad but well-standardized approach will be discussed.

#### 4pPPb6. Evidence of neural processing of the interaural correlation between long-duration noises in humans.

Wenjie J. Wang (Dept. of Psychology, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, wenjie@pku.edu.cn), Lingzhi Z. Kong (Dept. of Psychology, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, konglingzhi@gmail.com), Jingyu Y. Li (Dept. of Machine Intelligence, Speech and Hearing Research Center, 2 Science Building, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, lijy@cis.pku.edu.cn), Qiang Huang (Dept. of Machine Intelligence, Speech and Hearing Research Center, 2 Science Building, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, QIANG.HUANG@SPREADTRUM.COM), Xihong H. Wu (Dept. of Machine Intelligence, Speech and Hearing Research Center, 2 Science Building, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, wxh@cis.pku.edu.cn), Liang Li (Dept. of Psychology, Peking Univ., 5 Yeheyuan Road, Haidian District, 100871 Beijing, China, liangli@pku.edu.cn)

When the delay between two correlated sounds is sufficiently short, attributes of the lagging sound are perceptually captured by the leading sound, causing a fused image as coming from the leading-sound location. To investigate neural processing of the interaural correlation between long-duration sounds in humans, we recorded scalp event-related potentials (ERPs) to a transient silent gap that was inserted in long-duration broadband noises presented via headphones. The noises at the two ears were either correlated or uncorrelated. When the noises were correlated but not uncorrelated, participants mainly perceived only one gap image whose inside-head position was modulated by the interaural time difference (ITD). Compared to ERPs when the noises were uncorrelated, the amplitude of the N1/P2 component to the gap was smaller and amplitudes in the latency range of 100 to 200 ms were more imbalanced between the two hemispheres when the noises were correlated. When the ITD for the correlated noises was increased to 16 ms, which was beyond the ITD range for fusion, two gap images were perceived, and the amplitude of the N1/P2 component to the gap became larger and the sustained potentials during 300-500 ms after the gap onset became more negative. [Supported by NSFC].

## Session 4pPPc

Psychological and Physiological Acoustics: General Topics in Psychological and Physiological Acoustics VI  
(Poster Session)

Elizabeth Strickland, Cochair  
*Purdue University*

Armin Kohlrausch, Cochair  
*Philips Research Europe*

Alain De Cheveigne, Cochair  
*CNRS, Universite Paris 5, Ecole Normale Supérieure*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pPPc1. A computational model of binaural speech intelligibility level difference.** Kalle J. Palomäki (Adaptive Informatics Research Centre, Helsinki University of Technology, P.O. Box 5400, 02015 Espoo, Finland, kalle.palomaki@tkk.fi), Guy J. Brown (University of Sheffield, Dept. of Computer Science, Regent Court, 211 Portobello Street, S1 4DP Sheffield, UK, g.brown@des.shef.ac.uk)

This study addresses two questions relating to the binaural intelligibility level difference (BILD). First, we ask whether the BILD is underlain by an equalization-cancellation (EC) mechanism, in which a disparity between the interaural time difference of the target and masker is exploited within each frequency channel, rather than across channels. Second, we consider the effects of three sources of internal noise on the EC mechanism: jitter in neural delays, noise in the equalization process and nonlinearities in the auditory pathway. These issues are investigated using a computational model consisting of peripheral auditory model, binaural processor, auditory scene processor and automatic speech recognition system. The binaural model is based on EC processing, with performance limited by internal noise. The auditory scene processor groups speech harmonics by common F0 and identifies 'glimpses' in which the signal-to-noise ratio is favorable for speech. The performance of human listeners and the computational model are compared on the same speech intelligibility test (Edmonds & Culling, 2005, JASA 117 (5), 3069-3078). The BILD of human listeners can be replicated by adjusting parameters that determine the internal noise in the EC model; however, the speech reception threshold of the model is lower than that of human listeners.

**4pPPc2. Bernoulli coding on the auditory nerve and its implications for central auditory processing.** Robert A. Houde (Center for Communications Research, 125 Tech Park Drive, Rochester, NY 14623, USA, rahoude@gmail.com), James M. Hillenbrand (Western Michigan University, Dept of Speech Path & Aud., 1903 W. Michigan Ave., Kalamazoo, MI 49008, USA, james.hillenbrand@wmich.edu)

The auditory periphery is well represented as a bank of band pass filter/inner hair cell (IHC) channels, with each IHC providing half wave rectification, amplitude compression, and conversion to firing probability on the auditory nerve (AN) fibers innervating that IHC. Frequency resolution varies dramatically with sound intensity, ranging from sharp tuning near threshold to very broad at high intensities. Cochlear filtering provides a satisfactory representation of broadband characteristics such as timbre but not the fine frequency resolution required for perceptual frequency discrimination.

High resolution frequency analysis must, therefore, be provided by post-AN processes. We present a model of AN coding in which fine frequency analysis is carried out at central auditory stages. By this model the stochastic process on each AN fiber resulting from the IHC's firing probability is modeled by a Bernoulli process. As a result, the IHC output signal is transferred to the cochlear nucleus (CN) without further filtering, where it can be recovered by a simple summation over those AN fibers from the region of that IHC. We present a neurally plausible process for narrow band analysis at the CN using the regular pulses of chopper cells.

**4pPPc3. The first effect of pitch shift as a function of component spacing.** Adam Mielczarek (Acoustics Division, Wrocław University of Technology, Wybrzeże Wyspiańskiego 27A, 50-370 Wrocław, Poland, adam.mielczarek@pwr.wroc.pl)

The paper presents the results of an experiment regarding the influence of component spacing on the first effect of pitch shift. During the adjustment procedure, the listeners matched the pitch of the three-component complex to the same sensation produced by pure tone. The stimuli were composed of the 3rd, 4th and 5th harmonics of 100, 200 or 400 Hz shifted up in the frequency domain by 30 Hz. The level of each component was 50 dB SPL. The subject was presented with a 5-s sample of the test complex, and after a 500 ms break he had to define the pitch of the three-component complex using the matching tone. The results of the experiment suggest that the pitch shift phenomenon is based on the relative frequency rather than on the absolute frequency or the dominant component number.

**4pPPc4. Neuronal representation of pitch ambiguity.** Mark Sayles (Centre for the Neural Basis of Hearing, The Physiological Laboratory, Downing Street, CB2 3EG Cambridge, UK, ms417@cam.ac.uk), Ian M. Winter (Centre for the Neural Basis of Hearing, The Physiological Laboratory, Downing Street, CB2 3EG Cambridge, UK, imw1001@cam.ac.uk)

Iterated rippled noise (IRN) is produced by delaying a broadband noise by time  $d$ , multiplying by gain  $g$ , adding the delayed noise to the original, and repeating this process for  $n$  iterations. When  $g=+1$  IRN has a well-defined pitch at  $1/d$  Hz. If  $g=-1$  the pitch can be ambiguous. A gain of  $-1$  is equivalent to applying a frequency-independent phase shift  $\phi$  of  $\pi$  rads to the delayed noise ( $g=+1 \equiv \phi=0$ ). We recorded spike-trains from single units in the ventral cochlear nucleus in response to IRN with varying  $\phi$ . Units with high best frequencies represented waveform envelope modula-



tion (independent of  $\phi$ ), however, units in the phase-locking range of best frequencies represented stimulus fine structure (which varies with  $\phi$ ). Fine structure responders show a gradual transition from a well-defined peak in the interspike interval distribution at  $d$  when  $\phi=0$  to two equal-amplitude peaks flanking  $d$  when  $\phi=\pi$ , and a gradual shift back to a well-defined peak at  $d$  as  $\phi$  approaches  $2\pi$ . Within the dominance region for pitch interspike interval distributions account for psychophysical pitch matches of  $1.07/d$  and  $0.94/d$  Hz for  $\phi=\pi/2$  and  $3/2\pi$  respectively, as well as the ambiguous pitches of  $0.88/d$ ,  $1.14/d$ , and  $1/2d$  Hz heard when  $\phi=\pi$  rads.

#### **4pPPc5. The effect of regional dialect on the psychometric reliability and validity of two sets of Mandarin speech audiometry materials.**

Shawn L. Nissen (Brigham Young University, 138 TLRB, 1190 North 900 East, Provo, UT 84602, USA, shawn\_nissen@byu.edu), Richard W. Harris (Brigham Young University, 138 TLRB, 1190 North 900 East, Provo, UT 84602, USA, richard\_harris@byu.edu), Jamie Garlick (Brigham Young University, 138 TLRB, 1190 North 900 East, Provo, UT 84602, USA, jamie.garlick@gmail.com), Nathan Richardson (Brigham Young University, 138 TLRB, 1190 North 900 East, Provo, UT 84602, USA, nathan000000@yahoo.com)

Previous research has shown conflicting evidence on the effect of testing an individual's hearing acuity with speech perception materials created in a mutually intelligible, yet non-regional dialect. Thus, the aim of this study is to examine the validity and reliability of using previously developed psychometrically equivalent speech audiometry materials in Mainland Mandarin and Taiwan Mandarin to evaluate the speech perception abilities (word recognition and speech reception threshold) of regional and non-regional listeners of the presented dialects. In addition, this study will investigate whether a native speaker of one Mandarin dialect is able to accurately administer and score the results from listeners of a different regional dialect. Some aspects of the listeners' performance on materials from a non-regional Mandarin dialect were found to be significantly different statistically. However, it is unclear if such differences are large enough to make a difference in the clinical testing of speech perception. In terms of scoring accuracy, a high percentage of agreement was found between the two interpreters from different dialectal backgrounds. [Work supported by research funding from Brigham Young University School of Education]

#### **4pPPc6. Estimating the effective frequency of cochlear implant electrodes using contralateral residual acoustic hearing.**

Tim Green (UCL, Wolfson House, 4, Stephenson Way, NW1 2HE London, UK, tim.green@ucl.ac.uk), Andrew Faulkner (UCL, Wolfson House, 4, Stephenson Way, NW1 2HE London, UK, andyf@phon.ucl.ac.uk), Stuart Rosen (UCL, Wolfson House, 4, Stephenson Way, NW1 2HE London, UK, stuart@phon.ucl.ac.uk)

For some cochlear implant (CI) users a contralateral hearing aid provides significantly improved speech perception. Important factors in the bi-modal transmission of speech spectral information are likely to include the extent to which the frequency selectivity of residual hearing allows additional place-coded channels, and mismatches between frequency-to-place maps across modalities. When acoustic place coding extends above around 500 Hz an overlap of frequency coverage between acoustic and electric hearing may result in interaural conflicts. However, addressing this issue requires accurate knowledge of CI frequency-to-place maps. Effective characteristic frequencies of CI electrodes have previously been estimated using comparisons of the pitch produced by electrical stimulation with that produced by contralateral acoustic sinusoids. In the present work, the acoustic stimuli used for pitch comparisons are either sinusoids or  $1/3$  octave bands of noise. The latter minimize temporal pitch cues and may reduce differences in perceived quality between electrical and acoustical auditory sensations. Electrical stimuli are high-rate (900 pps or greater) single-electrode pulse trains. Comparisons are performed at different levels spaced over the dynamic range and both paired-comparison and adjustment tasks are used. Results will be discussed in relation to speech processing approaches for optimally combining an implant and contralateral hearing aid.

#### **4pPPc7. Performance on auditory temporal-processing tasks for speech and non-speech stimuli by young and elderly listeners.**

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Results from three auditory tasks are presented from a larger series of temporal-processing tasks completed in three sensory modalities by young and older adults. The first task measured temporal gap detection in noise bands. The second and third tasks used digitally processed vowels in four words (pit, pet, pot, put) as the stimuli. The second task required listeners to identify the order of either two- or four-vowel sequences presented monaurally or dichotically. The third task measured the identification of these four vowels when presented either before or after a noise or vowel-like masker (forward- or backward-masking tasks). Altogether, performance was obtained for 14 auditory temporal-processing measures. Young ( $N=20$ ) and older ( $N=50$ ) adults participated. Preliminary analyses (based on data from 50 of the 70 subjects) indicate that young listeners performed significantly better and with less variability than elderly listeners on all tasks. For most tasks, there was considerable overlap between the data from young and elderly listeners, indicating a modest negative impact of aging. At the individual level, correlational analyses among the older adults indicated that pure-tone thresholds were not predictive of temporal-processing performance and that performance on many of the temporal-processing tasks was moderately correlated. [Supported by NIA R01 AG022334.]

#### **4pPPc8. MEG measures of the auditory steady-state response: Sinusoidal and non-sinusoidal stimuli.**

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Human sensitivity to amplitude modulation has long been of interest to researchers, both in behavioural and neurological measures. Processing of amplitude modulation is implicated in the process of speech perception and sinusoidal amplitude modulation is used extensively to probe the mechanisms involved in encoding this information. The temporal envelope of speech is more accurately described as bursts of modulation rather than continuous modulation and the current work exposes participants to a continuum of modulation waveforms; from sinusoidal to pulsatile. Waveforms were amplitude modulated at 4 Hz and imposed upon a 500 Hz pure-tone carrier. The waveforms were generated using raised-cosine pulses with different half-durations. Half-durations of 8, 16, 24, 32, 64 and 125 ms were used (125 ms producing sinusoidal amplitude modulation at 4 Hz). Stimuli were 240 seconds in duration and responses were collected on a 248 channel whole-head MEG scanner. The frequency domain steady-state response was analysed from each condition in 14 participants, and results confirmed that the response to sinusoidal amplitude modulation was significantly lower than to modulations more representative of those found in speech signals. This suggests that non-sinusoidal stimuli may be more effective when investigating these auditory mechanisms.

#### **4pPPc9. Laboratory synthesis of industrial noise environments with predetermined statistical properties.**

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High-level nonGaussian noise is commonly found in a variety of industrial environments. Recent experiments have shown that for a given energy level, the statistical properties of a noise can have a strong effect on the extent of hearing loss produced in exposed individuals. In order to study, in an

animal model, the effects on hearing of such noise environments, the statistical properties of the noise as embodied in the kurtosis metric must be under experimental control. For a fixed value of kurtosis and energy level the following four variables will have a strong effect on hearing loss: (1) peak histogram; (2) interval histogram; (3) duration of noise transients; and (4) level of any background Gaussian noise. Simulations have shown that the relations among kurtosis and these variables are nonlinear. However, under certain restricted conditions, these relations may be linear. Accordingly, two strategies for designing controlled industrial noise exposures are presented: (1) the interval-priority model and (2) the duration-priority model. Computer simulations and measurements of actual acoustic environments showed that these two models could be effectively used to simulate a wide variety of realistic industrial noises.

**4pPPc10. Intervention for restricted dynamic range and reduced sound tolerance.** Charles Formby (University of Alabama, 700 University Boulevard East Suite 315, Tuscaloosa, AL 35487, USA, cformby@as.ua.edu), Monica Hawley (University of Maryland, 16 S. Eutaw St, Suite 500, Baltimore, MD 21201, USA, moncia@hawleyonline.net), Laguinn Sherlock (University of Maryland, 16 S. Eutaw St, Suite 500, Baltimore, MD 21201, USA, gsherlock@smail.umaryland.edu), Susan Gold (University of Maryland, 16 S. Eutaw St, Suite 500, Baltimore, MD 21201, USA, sgold@smail.umaryland.edu), Allyson Segar (University of Maryland, 0100 Lefrak Hall, College Park, MD 20742, USA, asegar@hesp.umd.edu), Christine Gmitter (University of Maryland, 0100 Lefrak Hall, College Park, MD 20742, USA, cgmitter@hesp.umd.edu), Justine Cannavo (University of Maryland, 0100 Lefrak Hall, College Park, MD 20742, USA, jcannavo@hesp.umd.edu)

Hyperacusis is an abnormal condition of sound intolerance that may cause some persons to reject amplified sound from their hearing aids. A significant secondary benefit reported for many patients receiving Tinnitus Retraining Therapy (TRT) is increased Loudness Discomfort Levels (LDLs). TRT involves both counseling and sound therapy (i.e., daily exposure to soft sound from bilateral noise generators (NGs)). We implemented a randomized, double-blind, placebo-controlled clinical trial to assess the efficacy of TRT as an intervention to improve sound tolerance in hearing-aid eligible persons with hyperacusis and/or restricted dynamic ranges. Subjects were assigned to one of four treatment groups: 1) full treatment, both counseling and NGs, 2) counseling and placebo NGs, 3) NGs without counseling, and 4) placebo NGs without counseling. They were evaluated at least monthly, typically for five months or more, on a variety of audiometric tests, including LDLs, the Contour Test for Loudness, and word recognition measured at comfortable and loud levels. Over 80% of the subjects assigned to full treatment achieved significant benefit (defined as shifts of greater than 10 dB in LDLs or the Contour Test uncomfortable level); whereas, most subjects assigned to a partial treatment group did not benefit from their treatment. [Supported by NIH].

**4pPPc11. Relationship between a visual stimulus with a feeling of depth and its equivalent sound pressure level (ESPL).** Hiroshi Hasegawa (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, hasegawa@is.utsunomiya-u.ac.jp), Hirotaka Ono (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, hasegawa@is.utsunomiya-u.ac.jp), Takumi Ito (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, hasegawa@is.utsunomiya-u.ac.jp), Ichiro Yuyama (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, yuyama@is.utsunomiya-u.ac.jp), Masao Kasuga (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, kasuga@is.utsunomiya-u.ac.jp), Miyoshi Ayama (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, nakatsue@is.utsunomiya-u.ac.jp)

This study investigated the equivalent perception between a visual stimulus and its associated sound. Experiments of an auditory-visual stimulus presentation using an audio-video clip of a man beating a drum on a road

were performed. The visual stimulus had a feeling of depth with a perspective view of the road. The visual stimulus was projected onto a screen that had the viewing angles of 43.8 deg.(W) \* 25.4 deg.(H). Four kinds of distance between the subject and the visual stimulus from 5 to 40 m, seven kinds of the delay time between auditory and visual stimulus from -8 F to 8 F (1 F = 1/30 s), where "+" indicates that the visual event preceded the sound, and nine levels of the sound stimulus from -12 dB to 12 dB of the standard sound pressure level (SPL) were combined and presented. We evaluated the sound pressure level matching with each presentation pattern (equivalent sound pressure level; ESPL). As a result, we obtained that the ESPL tended to decrease when the delay time increased (the sound was delayed). This result shows a possibility that the visual stimulus was a little shifted to the direction of the sound.

**4pPPc12. Temporal dynamics of stimulus specific processing in the human auditory cortex as revealed by electroencephalography.** Paul M. Briley (MRC Institute of Hearing Research, University Park, NG7 2RD Nottingham, UK, paul@ihr.mrc.ac.uk), Katrin Krumbholz (MRC Institute of Hearing Research, University Park, NG7 2RD Nottingham, UK, katrin@ihr.mrc.ac.uk)

When the same sound is presented repeatedly, the electrical brain response recorded over the scalp decreases in amplitude, an effect known as adaptation. Adaptation is dependent on both the similarity of the sounds and the time between them. It has been particularly well studied for a deflection of the electrical response known as the N100, which peaks about 100 ms after sound onset and receives major contributions from auditory cortical sources. Adaptation may reflect decreased sensitivity to repetitive stimuli, but could also indicate more efficient processing of familiar events. Research on adaptation has often employed an alternating tone paradigm (A-B-A-B), examining the effects of changing inter-stimulus interval (ISI) or the frequency separation between A and B tones. Decreasing the frequency separation leads to an increase in N100 adaptation, and it has been suggested that the frequency specificity of this adaptation sharpens with decreasing ISI. In contrast, some studies have used A-B pairs with long inter-pair gaps and have found an enhancement of the N100 response to the B tone at short ISIs. In order to gain a better understanding of the processes contributing to adaptation and enhancement, this study investigates the temporal dynamics and the frequency selectivity of these effects.

**4pPPc13. An Investigation of Width and Depth Perception toward a Sound Image Constructed of Multiple Variant Sound Waves Emitted from a Loudspeaker Array.** Yoko Yamakata (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, yamakata@nict.go.jp), Toshiyuki Kimura (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, t-kimura@nict.go.jp), Munenori Naoe (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, nm.s512.ex@nict.go.jp), Michiaki Katsumoto (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, katsumoto@nict.go.jp)

Many musical instruments, including violins and guitars, vibrate their resonant bodies differently over their surface when they make a sound. This paper aims to reveal the influences of such vibration variation of a soundboard surface on the width and depth perception of the sound image when listeners were in a near-field 50 cm or 1 m away from the soundboard. In this paper, a loudspeaker array mimicked the surface vibration as each loudspeaker makes a corresponding sound independently and cooperatively. Three types of sounds, synthesized single-tone, multi-tone, and instrumental, were used as sources. To know what factors affect the perception of the sound image, various test sound sets were prepared by varying an original sound set in amplitude or delay for each frequency for each loudspeaker. Eight subjects were asked to identify which sound image in a pair of test sounds was wider or farther than the other according to Scheffe's pair comparison method. The results shows that a test sound set with a delay varia-

tion, which mimics sounds emitted by bending vibrations propagating on a soundboard, obviously influences the perception of sound image width and that the amplitude variation does not have much influence.

**4pPPc14. MEG Recordings of Amplitude-modulated Noise and Tonal Stimuli in Healthy Adult Listeners.** Yang Zhang (University of Minnesota, Dept. of Speech-Language-Hearing Sci. & Center for Neurobehavioral Development, Minneapolis, MN 55455, USA, zhang470@umn.edu), Yingjiu Nie (University of Minnesota, Dept. of Speech-Language-Hearing Sci. & Center for Neurobehavioral Development, Minneapolis, MN 55455, USA, niex0008@umn.edu), Toshiaki Imada (University of Washington, Dept. of Speech & Hearing Sciences, and Institute for Learning & Brain Sciences, Box 357988, Seattle, WA 98195, USA, imada@u.washington.edu), Keita Tanaka (Tokyo Denki University, Research Center for Advanced Technologies, 270-1382 Inzai, Japan, ktanaka@rcat.dendai.ac.jp), Masaki Kawakatsu (Tokyo Denki University, School of Information Environment, 270-1382 Inzai, Japan, kawakatsu@asrl.dendai.ac.jp)

Amplitude modulation (AM) provides very important auditory information for the perception of complex sounds by normal listeners as well as cochlear implant users. The present study used a 122-channel whole-head magnetoencephalography (MEG) system to record auditory responses to amplitude-modulated pure tones and broadband noises in six healthy male adult subjects. The stimuli were presented in blocks of twenty with a brief silence in between, and the AM rates for both types of stimuli were at 20, 40, and 80 Hz. At least 80 artifact-free trials were collected for each stimulus. As expected, the MEG data showed a significant bilateral effect of AM rate in the N1m component. There was also strong evidence that the neural representations of both the unmodulated pure tone and noise stimuli in the auditory regions of both hemispheres could be significantly affected by the global context of block design stimulus presentation.

**4pPPc15. Critical-band compression method of speech enhancement for elderly people: Investigation of syllable and word intelligibility.** Keiichi Yasu (Dept. of Electrical and Electronics Engineering, Sophia University, 7-1 Kiyoi-cho, Chiyoda-ku, 102-8554 Tokyo, Japan, k-yasu@sophia.ac.jp), Hideki Ishida (Dept. of Electrical and Electronics Engineering, Sophia University, 7-1 Kiyoi-cho, Chiyoda-ku, 102-8554 Tokyo, Japan, ishida-h@sophia.ac.jp), Ryosuke Takahashi (Dept. of Electrical and Electronics Engineering, Sophia University, 7-1 Kiyoi-cho, Chiyoda-ku, 102-8554 Tokyo, Japan, ryo.t.sust@gmail.com), Takayuki Arai (Dept. of Electrical and Electronics Engineering, Sophia University, 7-1 Kiyoi-cho, Chiyoda-ku, 102-8554 Tokyo, Japan, arai@sophia.ac.jp), Kei Kobayashi (Dept. of Electrical and Electronics Engineering, Sophia University, 7-1 Kiyoi-cho, Chiyoda-ku, 102-8554 Tokyo, Japan, kei-koba@ba2.so-net.ne.jp), Mitsuko Shindo (Sophia Univ. Research Center for Communication Disorders, 7-1 Kiyoi-cho, Chiyoda-ku, 102-8554 Tokyo, Japan, shindo-m@sophia.ac.jp)

Auditory filters for the hearing impaired tend to be wider than those of normal hearing people. Thus, the frequency selectivity decreases because of increased masking effects [Glasberg and Moore, *J. Acoust. Soc. Am.*, 79(4), 1020-1033, 1986]. We have developed a method, called "critical-band compression," in which the critical band is compressed along the frequency axis [Yasu et al., *Handbook of the International Hearing Aid Research Conference (IHCON)*, 55, Lake Tahoe, 2004]. We investigated whether our method improves syllable and word intelligibility. Thirty one elderly people participated in experiments. First, we measured the auditory filter bandwidth using a notched noise method [Patterson, *J. Acoust. Soc. Am.*, 59(3), 640-654, 1976]. Second, we conducted syllable and word intelligibility tests. The compression rates of critical-band compression were set to 0% for the original, and 25%, 50%, and 75%. The results were that the percentages of correct responses were almost the same at 0%, 25% and 50% compression rates

for syllable and word intelligibility. A significant correlation was not obtained between the compression rate of processing and the auditory filter bandwidth. [Work supported by JSPS.KAKENHI (16203041) and Sophia University Open Research Center from MEXT.]

**4pPPc16. Speaker size discrimination for acoustically scaled versions of whispered words.** Yoshie Aoki (Faculty of Systems Engineering, Wakayama University, 930 Sakaedani, 640-8510 Wakayama, Japan, s085065@sys.wakayama-u.ac.jp), Toshio Irino (Faculty of Systems Engineering, Wakayama University, 930 Sakaedani, 640-8510 Wakayama, Japan, irino@sys.wakayama-u.ac.jp), Hideki Kawahara (Faculty of Systems Engineering, Wakayama University, 930 Sakaedani, 640-8510 Wakayama, Japan, kawahara@sys.wakayama-u.ac.jp), 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)

Humans can extract the message from the voices of men, women, and children without being confused by the size information, and they can extract the size information without being confused by the message. This suggests that the auditory system can extract and separate information about vocal tract shape from information about vocal tract length. Smith et al. [*J. Acoust. Soc. Am.* 117(1), 305-318 (2005)], Ives et al. [*J. Acoust. Soc. Am.* 118(6), 3816-3822 (2005)], and Aoki et al. [ARO, 31st Midwinter meeting (2008)] performed discrimination experiments with acoustically scaled vowels, syllables, and naturally spoken words, respectively, and demonstrated that the ability to discriminate speaker size extends beyond the normal range of speaker sizes. Smith and Patterson [BSA Cardiff (2005)] demonstrated that performance on the size-discrimination task is only marginally reduced when the vowels are unvoiced. We extended these size-discrimination experiments to whispered versions of naturally spoken, four-mora Japanese words. The just-noticeable-difference for the whispered words was about 6 %, which is roughly the same as that for voiced words. The results show that voicing is not required for effective extraction of the size information. Research supported by JSPS Grant-in-Aid [B18300060] and the UK-MRC [G0500221].

**4pPPc17. Effects of modality-dependent cuing and eye movements on sound localization.** Beáta Tomoriová (Laboratory of Perception and Cognition, Technical University of Košice, Letná 9, 042 00 Košice, Slovakia, beata.tomoriova@gmail.com), Rudolf Andoga (Laboratory of Perception and Cognition, Technical University of Košice, Letná 9, 042 00 Košice, Slovakia, rudolf.andoga@tuke.sk), Norbert Kopčo (Laboratory of Perception and Cognition, Technical University of Košice, Letná 9, 042 00 Košice, Slovakia, kopco@tuke.sk)

A previous study of visual and auditory hemispheric cuing in horizontal sound localization found modality-dependent effects of cuing resulting in biases in responses [Kopco, Tomoriová, Andoga, *J. Acoust. Soc. Am.* 121, 3094, 2007]. The previous study also suggested that some of the effects might be due to eye movements as eye fixation was not controlled. The goal of the current study was to isolate the attentional effects from the eye movement effects. An experiment identical to the previous one was performed, with the exception that the subjects were fixating the center of the audiovisual display. Localization performance was measured for transient auditory stimuli originating in the frontal horizontal plane. In most runs, a cue preceded the stimulus and indicated (correctly or incorrectly) the hemisphere (left vs. right) from which the subsequent target arrived. The cues differed by modality and the cue-to-target onset asynchrony. The listeners were instructed to focus their attention to the cued side. Compared to the previous study, a reduction in some effects was observed. However, modality-dependent biases in performance persisted, confirming that auditory spatial attentional control is modality dependent and operating on time scale of seconds. [Supported by the Slovak Science Grant Agency.]



**4pPPc18. Synchrony-asynchrony discrimination of audio-visual signals in auditory streaming.** Maori Kobayashi (R.I.E.C., Tohoku University, 2-1, Katahira, Aoba-ku, 980-8577 Sendai, Japan, maori@ais.riec.tohoku.ac.jp), Shuichi Sakamoto (R.I.E.C., Tohoku University, 2-1, Katahira, Aoba-ku, 980-8577 Sendai, Japan, saka@ais.riec.tohoku.ac.jp), Yôiti Suzuki (R.I.E.C., Tohoku University, 2-1, Katahira, Aoba-ku, 980-8577 Sendai, Japan, yoh@ais.riec.tohoku.ac.jp)

Temporal synchrony is a critical condition for integrating information presented in different sensory modalities. In this study, the effect of tonal organization on synchrony-asynchrony discrimination was examined. The auditory sequences were four repetitions of a triplet pattern comprising a low-frequency tone (L) and a high-frequency tone (H). The frequency difference ( $\Delta F$ ) between L and H was either approximately 1/12, 1/6, 1/3, 1/2, or 1 octave, centered at 1 kHz. Each tone was of 33.2 ms duration including rising and falling raised-cosine ramps of 5 ms. The stimulus onset asynchrony (SOA) of adjacent tones was randomized between 33.2 and 332 ms. The tone sequences were presented diotically via headphones at 65 dB SPL. The visual stimulus was a luminance-modulated Gaussian blob presented on a CRT monitor. The visual stimulus duration was 8.3 ms. Synchrony-asynchrony discrimination thresholds of visual-auditory stimulus onsets were measured using the 2IFC paradigm with a 2-up 1-down method under six  $\Delta F$  conditions. The results demonstrated that synchrony-asynchrony discrimination improved for audio-visual pulse trains at  $\Delta F$  between L and H greater than 1/3 octave, suggesting that audio-visual synchrony perception is influenced by the build-up of auditory streaming.

**4pPPc19. Voice quality of emphatics in comparison with non-emphatics in Moroccan Arabic.** Karim Shoul (Lab. LPP UMR 7018 CNRS, 19, rue des Bernardins, 75005 Paris, France, shoulkarim@hotmail.com)

Based on acoustic and physiological data, this study examines the voice quality of emphaticized (also called pharyngealized) vowels in Moroccan Arabic. The aim is to determine whether, as argued by some authors (Heath 1987, Fre woldu 1986), these vowels are creaky or glottalized. For this purpose, the /a/ vowel is considered in syllables after initial and intervocalic /t, d, s/ and their emphatic counterparts for an acoustic study as well as a physiological one (Fourcin's EGG 1974). The cues examined include F0 values, duration and amplitude of the acoustic and glottal signals, as well as the open quotient (Oq) (as seen by Henrich 2001). Results of the acoustic analysis show no significant differences between emphaticized and non-emphaticized vowels as far as F0 values, duration and amplitude of the acoustic signal are concerned. The same absence of difference is observed from the EGG experiment, which indicate that Oq represents half of the whole glottal phase. These findings suggest that emphaticized vowels, just like the corresponding non-emphaticized counterparts, are characterized by a modal voice quality. They imply that 'secondary' pharyngealization does not require a narrowing of the supra-glottic cavity which would affect the mode of vocal-fold vibrations.

**4pPPc20. Influence of multi-channel dynamic range compression on intelligibility: effect of envelope modulation bandwidth.** Michael A. Stone (University of Cambridge, Department of Experimental Psychology, Downing Street, CB2 3EB Cambridge, UK, mas19@cam.ac.uk), Christian Füllgrabe (University of Cambridge, Department of Experimental Psychology, Downing Street, CB2 3EB Cambridge, UK, cf277@cam.ac.uk), Brian Moore (University of Cambridge, Department of Experimental Psychology, Downing Street, CB2 3EB Cambridge, UK, bcjm@cam.ac.uk)

Stone and Moore [J. Acoust. Soc. Am. (in press)] showed that, as the speed and number of channels in a multi-channel compressor increased, intelligibility of noise-vocoded signals in a competing speech task decreased. The noise vocoder is often used to simulate the information conveyed by cochlear implants. However, the vocoder of Stone and Moore preserved only low-rate (<45 Hz) envelope modulations whereas some implantees show sensitivity to envelope modulation rates up to about 300 Hz. Furthermore, intrinsic fluctuations in the noise carriers affect the reception of low-rate modulations of the signal (Whitmal et al. [J. Acoust. Soc. Am. 122: 2376-2388, (2007)]). Here, a tone vocoder with N=8 or N=16 channels was used. Vocoding was preceded by N/8, N/4, or N/2 channels of com-

pression, each using one of three speeds, affecting modulation rates up to about 2, 6, or 18 Hz, respectively. The lowpass filters extracting the channel envelopes had corner frequencies of 45 or 180 Hz. Intelligibility was measured using IEEE sentences with a competing speaker. The deleterious effect of compression with increasing channel number and speed was greater for the lower corner frequency. Compression of rates below 6 Hz affected intelligibility, independent of the presence of higher modulation rates.

**4pPPc21. Characterizing lexical interferences in informational masking during speech-in-speech comprehension.** Michel Hoen (Laboratoire d'Etude des Mécanismes Cognitifs (EMC). EA 3082 CNRS, Université Lumière Lyon 2, 5, Avenue Pierre Mendès-France, 69676 Bron Cedex, France, Michel.Hoen@univ-lyon2.fr), Claire Grataloup (Laboratoire Dynamique du Langage (DDL), UMR 5596 CNRS, Université de Lyon et Lyon 2, Institut des Sciences de l'Homme - 14 avenue Berthelot, 69363 Lyon Cedex 07, France, Claire.Grataloup@pse.unige.ch), François Pellegrino (Laboratoire Dynamique du Langage (DDL), UMR 5596 CNRS, Université de Lyon et Lyon 2, Institut des Sciences de l'Homme - 14 avenue Berthelot, 69363 Lyon Cedex 07, France, Francois.Pellegrino@univ-lyon2.fr), Lionel Collet (Univ. Lyon 1 - Lab. Neurosciences, Service Pr Collet, Pavillon U, Hôpital Edouard Herriot, F-69003 Lyon, France, iakhoun@olfac.univ-lyon1.fr), Fanny Meunier (Laboratoire Dynamique du Langage (DDL), UMR 5596 CNRS, Université de Lyon et Lyon 2, Institut des Sciences de l'Homme - 14 avenue Berthelot, 69363 Lyon Cedex 07, France, fanny.meunier@univ-lyon2.fr)

Results from our former research on the characterization of informational masking effects occurring during speech-in-speech comprehension showed that phonological and lexical information create specific informational masking effects, depending on the number of speakers involved (Hoen et al., 2007). The goal of the present study was to better characterize purely lexical factors potentially participating into informational masking phenomena. We evaluated speech-in-speech comprehension performances of 40 normal hearing participants listening to isolated lexical items presented together with different speech babble sounds. Lexicality of target items was controlled by using words of variable lexical frequency, high- vs. low-frequency items, as well as pseudowords. The interaction between lexicality of target items and lexicality of words in background noise was controlled by creating babble sounds made of high- or low-frequency words. Results show that, as in silence, lexicality of target items plays a determinant role, high-frequency words being always more intelligible (70%) than low-frequency words (50%), or pseudowords (37%). Conversely, the frequency of lexical items present in the babble had an effect only on the comprehension of pseudowords, the latter being more intelligible in a background of low-frequency items. Together, these results give new precisions on the detailed informational masking effects occurring during speech-in-speech comprehension.

**4pPPc22. Perceived plausibility of a multi-modal musical performance with introduced auditory-visual spatial and temporal mismatch.** Daniel Valente (Rensselaer Polytechnic Institute, Greene Bldg., 110 8th St., Troy, NY 12180, USA, danvprod@gmail.com), Jonas Braasch (Rensselaer Polytechnic Institute, Greene Bldg., 110 8th St., Troy, NY 12180, USA, braasj@rpi.edu)

One of the biggest problems for multi-modal virtual musical performance is that of auditory-visual mismatch. Often such performances place musicians in a dry, studio-like environment and rely on acoustic room modeling techniques to place the musician's sound in an environment more conducive to musical performance. The problem, for example occurs when the musician sounds as if he/she is in a large performance space but from a visual standpoint appears to be located in a very small studio environment. This experiment aimed to reveal the impact of such mismatches on the perceived plausibility of a multi-modal musical performance. Listeners were confronted with three contrasting musical excerpts presented in a virtual environment with varying degrees of auditory visual mismatch. In the first phase, listeners were able to adjust the acoustic modeling algorithm using the salient parameters of direct to reverberant ratio and reverberation time. As the visual space volume that the performance was presented in increased, the listeners repeatedly increased reverberation time and decreased direct to reverberant ratio. During the second phase, where varying levels of spatial



and temporal mismatch were purposefully introduced, the level of repeatability and accuracy of the listening group decreased despite subjective responses indicating accurate perceived performance realism.

**4pPPc23. Haircell non-functionality and dead regions in the cochlea: an exploring study.** Bastiaan Warnaar (AMC - Dept. of Clinical and Experimental Audiology, Meibergdreef 9, 1105AZ Amsterdam, Netherlands, b.warnaar@amc.nl), Wouter A. Dreschler (AMC, Clinical and Experimental Audiology, 1105 Amsterdam, Netherlands, w.a.dreschler@amc.uva.nl)

Dead regions (DRs) refer originally to regions in the cochlea without evoked electrical potentials due to non-functional inner hair cells and/or fibers of the auditory nerve. The focus of this study is to characterize hair cell non-functionality in the cochlea by means of psychophysical tests. A battery of tests was administered to a group of 13 subjects with a steep sloping tone threshold ( $>50$  dB/octave) between 1 and 2 kHz and severe loss ( $>60$  dB HL) at high frequencies. Psychophysical Tuning Curves (PTC) and Threshold Equalizing Noise (TEN) are the classical tests to diagnose dead regions. Both use as criteria the phenomenon of off-frequency listening. Based on the results of complementary tests like Notched Noise measurements (NN) and Otoacoustic Emissions (OAE), it is argued that off-frequency listening is not necessarily connected to loss of inner hair cells and/or nerve fibers. Furthermore, combination tones produced by well functioning outer hair cells at places of severe hearing loss ( $>60$  dB HL) are found. This may be explained by the presence of a dead region, which is verified with PTC and/or TEN measurements in 3 out of 4 of the cases.

**4pPPc24. Audio-visual quality model for internet protocol television services.** Marieneige Garcia (Deutsche Telekom Laboratories, Berlin Institute of Technology, Ernst-Reuter-Platz 7, 10587 Berlin, Germany, marie-neige.garcia@telekom.de), Alexander Raake (Deutsche Telekom Laboratories, Berlin Institute of Technology, Ernst-Reuter-Platz 7, 10587 Berlin, Germany, alexander.raake@telekom.de)

This paper presents a model for predicting the perceived audio-visual quality of IPTV services. Our model follows a modular approach and audio-visual quality is deduced from the perceived audio quality, the perceived video quality, the interaction between the audio quality and the video quality and the quality of the interaction between audio and video (lip-synchronization). In its current form, the model covers H.264 video codec, Standard Definition and High Definition video resolutions, MP2 audio codec and wav audio format. Addressed degradations, which generate different visual and auditory perceptual dimensions, are compression artifacts, packet losses, reduced bandwidth and delay between audio and video. Results demonstrate a mutual influence of the perceived audio and video qualities and the predominance of the video quality on the overall audio-visual quality. We analyze the interaction between visual perceptual dimensions, like blockiness, blurriness, slicing and freezing, and auditory perceptual dimensions, like frequency content (brightness), interruptedness, and freezing. We also study the influence of the type of the degradation on the interaction between perceived audio and video qualities. At last, we examine the influence of the audio-visual content (music video, news, etc.) on the perceived audio-visual quality. An outlook highlights future model extensions.

**4pPPc25. Neural coding of envelope and fine structure in noise degraded speech.** Jayaganesh Swaminathan (Purdue University, 500 Oval Drive, West Lafayette, IN 47907, USA, jswamy@purdue.edu), Michael Heinz (Purdue University, 500 Oval Drive, West Lafayette, IN 47907, USA, mheinz@purdue.edu)

Numerous perceptual studies have revealed that envelope is sufficient for speech perception in quiet, but that temporal fine structure (TFS) is required for speech perception in noise. However, the neural correlates of these perceptual observations remain unknown. The primary focus of the present work was to develop and evaluate neural cross correlation coefficient (CCC) metrics to quantify envelope and TFS coding in auditory-nerve responses to noise degraded speech. Shuffled auto- and cross-correlogram analyses were used to compute separate CCCs to quantify stimulus-related envelope and fine structure based on neural spike train data from a computational auditory-nerve model. The neural CCCs have a wide dynamic range as revealed by near-zero values for uncorrelated conditions and near-one

values for correlated conditions based on broadband noise responses. Spectrally matched noise was systematically added to a speech sentence at different signal-to-noise ratios (SNRs). Initial analyses reveal that  $CCC\_ENV > CCC\_TFS$  for positive SNRs, whereas  $CCC\_TFS > CCC\_ENV$  for negative SNRs. Predicted effects of hearing loss on envelope and TFS coding will also be discussed. These neural metrics can be used to evaluate temporal coding of speech with implications for cochlear-implant and hearing-aid strategies. Supported by NIH-NIDCD.

**4pPPc26. The effect of masker type and word-position on word recall and sentence understanding.** Payam Ezzatian (University of Toronto Mississauga, 3359 Mississauga rd N., Mississauga, ON L5L 1C6, Canada, payam.ezzatian@utoronto.ca), Liang Li (University of Toronto Mississauga, 3359 Mississauga rd N., Mississauga, ON L5L 1C6, Canada, liang.li@utoronto.ca), Kathy Pichora-Fuller (University of Toronto Mississauga, 3359 Mississauga rd N., Mississauga, ON L5L 1C6, Canada, k.pichora.fuller@utoronto.ca), Bruce Schneider (University of Toronto Mississauga, 3359 Mississauga rd N., Mississauga, ON L5L 1C6, Canada, bruce.schneider@utoronto.ca)

Speech understanding is influenced by not only the presence, but also the specific nature of maskers. Noise maskers primarily result in energetic masking, whereas speech maskers create additional interference due to linguistic and acoustic similarities to the target. The present study examined the influence of different types of maskers and target word position on the immediate recall of words in sentences by normal-hearing younger adults. In Experiment 1, nonsense sentences with 3 keywords (e.g., A "house" should "dash" to the "bowl".) were presented against a background of speech-spectrum noise or two-talker nonsense speech. With the speech masker, accuracy increased with word position. With the speech-spectrum noise masker, performance was highest for the first word and did not vary linearly with word position. In Experiment 2, when the speech-masker was noise-vocoded to preserve envelope information while disrupting fine structure cues and minimizing semantic content, performance was similar to that found with the speech-spectrum masker. The results suggest that the ability to track a target sentence in conditions of informational masking improves as the target utterance unfolds over time.

**4pPPc27. Training-induced auditory plasticity measured using auditory steady-state responses.** Karolina Kluk (The University of Manchester, Human Communication and Deafness Research Group, Ellen Wilkinson Building, Oxford Road, M13 9PL Manchester, UK, karolina.kluk@manchester.ac.uk), Christine M. Tan (University of Essex, Department of Psychology, Wivenhoe Park, CO4 3SQ Colchester, UK, ctan@essex.ac.uk), Michael S. John (Rotman Research Institute, Baycrest Centre for Geriatric Care, 3560 Bathurst Street, Toronto, AB M6A 2E1, Canada, sjohn@rotman-baycrest.on.ca), Terence W. Picton (Rotman Research Institute, Baycrest Centre for Geriatric Care, 3560 Bathurst Street, Toronto, AB M6A 2E1, Canada, tpicton@rotman-baycrest.on.ca)

Re-mapping of the primary auditory cortex may be induced by extensive training. For example, training of monkeys to perform frequency discrimination (FD) at one carrier frequency expands the representation of that frequency region in the auditory cortex. This study was intended to demonstrate training-induced auditory plasticity using auditory steady-state responses (ASSRs) in humans. Right-handed, non-musicians underwent FD training in their left ear only at 1 kHz. ASSRs were recorded to 1- and 2-kHz amplitude modulated tones (100 % AM depth at rates of 41, 83 and 45, 87 Hz, across two conditions). ASSRs recorded at the start of the experiment were compared with the ASSRs recorded after three two-hour sessions of FD training scheduled 24 hours apart. The results revealed significant increase in the amplitude of ASSRs recorded to 41- and 45-Hz AM tones (at 1 kHz only) presented to the trained left ear. There was no significant change in the amplitude of ASSRs recorded to the 2-kHz tones or to any stimuli presented to the un-trained right ear. As expected FD training

had no effect on 83- and 87-Hz ASSRs (which are generated mainly in the brainstem). These results support the idea of training-induced reorganization of the auditory cortex.

**4pPPc28. Hearing-screening tests based on filtered sounds and on speech-in-noise intelligibility tests.** Bozena Kostek (Gdansk University of Technology, Multimedia Systems Department, 11/12 Gabriela Narutowicza Street, 80-952 Gdansk, Poland, bozenka@sound.eti.pg.gda.pl), Andrzej Czyzewski (Gdansk University of Technology, Multimedia Systems Department, 11/12 Gabriela Narutowicza Street, 80-952 Gdansk, Poland, andcz@sound.eti.pg.gda.pl), Lukasz Kosikowski (Gdansk University of Technology, Multimedia Systems Department, 11/12 Gabriela Narutowicza Street, 80-952 Gdansk, Poland, kosiq@sound.eti.pg.gda.pl), Krzysztof Kochanek (The Institute of Physiology and Pathology of Hearing, Pstrowskiego 1, 01-943 Warsaw, Poland, k.kochanek@ifps.org.pl), Henryk Skarzynski (The Institute of Physiology and Pathology of Hearing, Pstrowskiego 1, 01-943 Warsaw, Poland, h.skarzynski@ifps.org.pl)

A hearing-screening system dedicated to small-children in pre-schools and primary schools is described in the paper. It uses as a hardware a palm-top computer supplemented with a small sound calibrating device. The described application provides tests that employ automatic questionnaire analysis, audiometric test procedures, and assessment of speech intelligibility in noise. In the speech-in-noise intelligibility tests, pictures are used for young children, and the screening tests are supervised by adults. Apart from the standardized audiometric tests, the screening tests employ environmental sounds filtered in audiometric frequency bands and calibrated as to their levels. When all the testing is completed, the system automatically analyzes the results for each child examined. The decision is made automatically by the expert system taking into account the number of incorrect answers. Children whose hearing impairment is confirmed are referred to treatment in rehabilitation centers. The project presented is a part of the large-scale "I can hear..." screening tests program carried out in Poland for the last few years. This may help to increase awareness and inspire action against noise at a very early age. The methods employed for filtering and calibration environmental sounds and results achieved are presented in the paper.

**4pPPc29. The effect of masker type and word-position on word recall and sentence understanding.** Payam Ezzatian (University of Toronto Mississauga, 3359 Mississauga rd N., Mississauga, ON L5L 1C6, Canada, payam.ezzatian@utoronto.ca), Liang Li (University of Toronto Mississauga, 3359 Mississauga rd N., Mississauga, ON L5L 1C6, Canada, liang.li@utoronto.ca), Kathy Pichora-Fuller (University of Toronto Mississauga, 3359 Mississauga rd N., Mississauga, ON L5L 1C6, Canada, k.pichora.fuller@utoronto.ca), Bruce Schneider (University of Toronto Mississauga, 3359 Mississauga rd N., Mississauga, ON L5L 1C6, Canada, bruce.schneider@utoronto.ca)

Speech understanding is influenced by not only the presence, but also the specific nature of maskers. Noise maskers primarily result in energetic masking, whereas speech maskers create additional interference due to linguistic and acoustic similarities to target speech. The present study examined the influence of different types of maskers and target word position on the immediate recall of words in sentences by normal-hearing younger adults. In Experiment 1, nonsense sentences with 3 keywords (e.g., A "house" should "dash" to the "bowl") were presented against a background of speech-spectrum noise or two-talker nonsense speech. With the speech masker, accuracy increased with word position in a linear fashion. With the speech-spectrum noise masker, performance was highest for the first word but the same for the second keyword and third keyword. In Experiment 2, when the speech-masker was noise-vocoded to preserve envelope information while disrupting fine structure cues and minimizing semantic content, performance was similar to that found with the speech-spectrum noise masker. The results suggest that the ability to track a target sentence in conditions of informational masking improves as the target utterance unfolds over time.

**4pPPc30. Empirical comparisons of pitch patterns in music, speech, and birdsong.** Adam T. Tierney (UC San Diego Dept. of Cognitive Science, Neurosciences Institute, 9500 Gilman Drive, La Jolla, CA 92093-0515, USA, AdamTierney@gmail.com), Frank A. Russo (Ryerson University Department of Psychology, 350 Victoria Street, Toronto, ON M5B 2K3, Canada, russo@ryerson.ca), Aniruddh D. Patel (Neurosciences Institute, 10640 John Jay Hopkins Drive, La Jolla, CA 92121, USA, apatel@nsi.edu)

In music, large intervals ("pitch skips") are often followed by reversals, and phrases often have an arch-like shape and final durational lengthening. These regularities could reflect motor constraints on pitch production or could reflect the melodic characteristics of speech. To distinguish between these possibilities we compared pitch patterns in instrumental musical themes, sentences, and birdsongs. Patterns due to production-related constraints should be common to all three domains, whereas patterns due to statistical learning from speech should be present in speech but not birdsong. Sequences were taken from English and French instrumental classical music, sentences from 4 languages, and songs of 56 songbird families. For sentences and birdsongs each syllable/note was assigned one pitch. For each sequence, we quantified patterns of post-skip reversals, the direction of the initial and final interval, the relative duration of the final syllable/note, and the pitch contour shape. Post-skip reversals predominated in all domains, likely reflecting a shared constraint: skips frequently take melodies toward the edges of the pitch range, forcing a subsequent reversal (as suggested by Von Hippel & Huron, 2000). Arch-like contours and final lengthening were found in music and speech but not birdsong, possibly reflecting an influence of speech patterns on musical structure.

**4pPPc31. Form and content in emotional reactions to sounds.** Daniel Vastfjall (Chalmers University of Technology, Division of Applied Acoustics - Chalmers Room Acoustics Group, Sven Hultins gata 8a, 41296 Gothenburg, Sweden, daniel.vastfjall@psy.gu.se), Erkin Asutay (Chalmers University of Technology, Division of Applied Acoustics, Sven Hultinsgata 8a, 41296 Gothenburg, Sweden, erkin@student.chalmers.se), Anders Genell (Chalmers University of Technology, Division of Applied Acoustics, Sven Hultinsgata 8a, 41296 Gothenburg, Sweden, Anders.Genell@ta.chalmers.se), Ana Tajadura (Chalmers University of Technology, Division of Applied Acoustics, Sven Hultinsgata 8a, 41296 Gothenburg, Sweden, ana.tajadura@chalmers.se)

People react emotionally to auditory stimuli. Despite this fact relatively little is known how sounds can create emotional reactions in listeners. We have developed a framework, the Emotion Reaction Model (ERM), that predict that both form features (i.e. classical psychophysical attributes such as loudness, sharpness etc) and content features (i.e. psychological associations to the sound producing source). Using ERM we tested the relative contribution of form vs. content in producing emotional reactions to sounds. In a first experiment, participants rated their emotional reactions to sounds from qualitatively different categories (animals, humans, machine noise) and to same sounds with time or frequency scrambling applied (thus rendering them difficult to identify, but with retained psychoacoustical properties). Experiments 2 used the same sounds but with a priming procedure and experiment 3 assessed emotional reactions using physiological measures. Overall, content, rather than form, appeared to have the biggest impact on emotional reactions. This research may complement traditional psychoacoustical theories that focus solely on form features.

**4pPPc32. Learning to read; attaching your ear to your tongue.** Eugene Galanter (Children's Progress, Inc./ Columbia University, 460 Riverside Drive, New York, NY 10027, USA, eg53@columbia.edu)

Really early learning predicts later success in school, life, and society (1), (2). Speech fluency, measured by accuracy, rate, and prosody (3), predicts reading comprehension (4) and reading proficiency (5). Speech fluency may be improved by one-on-one teacher interactions. When properly administered they give teachers useful information, but are time-consuming and subject to teacher's expectations. The results are commonly norm-referenced; a "wait to fail" model where teachers find deficits later in "high-stakes tests" (6). These failures led us to develop "voice mirroring (VM)." (7) V-M lets children self-correct their speech by hearing it mirrored after amplification. The child hears discrepancies between their own speech and

their teacher's to self-correct. Children supported by V-M demonstrate better reading skills as estimated by their teachers; (single-blind). Well-spoken children read well

**4pPPc33. Temporal response of a simplified bidimensional numerical model of the cochlea.** Christine Lepine (Université Paul Sabatier, PHASE, 118, route de Narbonne, 31062 Toulouse cedex 9, France, salmets@aol.com), Vincent Gibiat (Université Paul Sabatier, PHASE, 118, route de Narbonne, 31062 Toulouse cedex 9, France, gibiat@cict.fr)

Within the framework of a study related to bone conduction, numerical simulations have been performed in the time domain, with the aim of comparing the cochlear partition displacement in the case of different places of stimulation. An oversimplified 2D model of the cochlea is used. It is first excited with pulses centered on various audible-range frequencies with a localisation of the source which is analogous to the position of the oval window. Secondly, new sets of calculations introduce different localisations and/or spatial extensions of the sources. An analogy with seismology being adequate to simulate the solid-fluid (cochlear partition-perilymph) coupling, a finite difference numerical simulation based upon the Virieux scheme for elastic waves propagation has been used. The movement of the simplified basilar membrane is observable when excited via air or bone conduction. Results of the propagation of a single pulse within the model will be presented and discussed through information available in literature.

**4pPPc34. Modelling of the cochlea response as a versatile tool for acoustic signal processing.** Marinus M. Boone (University of Technology Delft, Lorentzweg 1, 2628 CJ Delft, Netherlands, m.m.boone@tudelft.nl), Diemer De Vries (University of Technology Delft, Lorentzweg 1, 2628 CJ Delft, Netherlands, d.devries@tudelft.nl), Tjeerd C. Andringa (University of Groningen, Dept. Artificial Intelligence, P.O. Box 407, 9700 AK Groningen, Netherlands, t.andringa@ai.rug.nl), Anton Schlesinger (University of Technology Delft, Lorentzweg 1, 2628 CJ Delft, Netherlands, A.Schlesinger@tudelft.nl), Jasper Van Dorp Schuitman (University of Technology Delft, Lorentzweg 1, 2628 CJ Delft, Netherlands, J.vanDorpSchuitman@tudelft.nl), Bea Valkenier (University of Groningen, Dept. Artificial Intelligence, P.O. Box 407, 9700 AK Groningen, Netherlands, bea@ai.rug.nl), Hedde Van De Vooren (University of Groningen, Dept. Artificial Intelligence, P.O. Box 407, 9700 AK Groningen, Netherlands, h.van.de.vooren@ai.rug.nl)

The inner ear or cochlea processes the acoustic signals that enter the oval window into a specific time-frequency pattern. Many acoustic signal processing methods are based on this behaviour. A fundamental method is to calculate this time-frequency response by solving the differential equation of the movement of the basilar membrane, followed by a visualisation of the excitation patterns in a time-frequency plot. For that purpose Continuity Preserving Signal Processing (CPSP) is a promising method. In the presentation an overview will be given of a project that is carried out by TUD (University of Technology Delft) together with RUG (University of Groningen) being sponsored by STW (Dutch Technology Foundation). The project divides into four sub-projects which are closely related: Automatic Keyword Spotting, Machine Analysis and Diagnostics, Speech Intelligibility Enhancement for Hearing Aids and Quality Assessment of Room Acoustics. Results that have been obtained in the project will be summarised. Detailed results of the sub-projects will be presented in separate presentations.

**4pPPc35. Improving speech intelligibility based on a conjunction of multiple perceptual models.** Anton Schlesinger (University of Technology Delft, Lorentzweg 1, 2628 CJ Delft, Netherlands, A.Schlesinger@tudelft.nl), Marinus M. Boone (University of Technology Delft, Lorentzweg 1, 2628 CJ Delft, Netherlands, m.m.boone@tudelft.nl)

The enhancement of speech intelligibility is crucial to the success of hearing aids and other speech transmission systems. Such systems often have to operate in real-world situations, in which noise, reverberation and competing talkers impair the signal-to-noise ratio and therewith the degree of intelligibility. Psychoacoustic and neuropsychological models progressively explain the working principle of the auditory path. In order to technologically achieve the mammalian hearing-performance, Computational Auditory Scene Analysis (CASA), a vivid field of research in acoustics, con-

verts these models into technical approaches. Asking for neuropsychological evidence that underlies the extraction of sound-objects in reverberation and noise, two models are essential. These are echo-suppression, which is subject to the precedence effect, and modulation perception, performed by a neural mapping of time-information of the auditory input into a periodotopical organization. The models have been applied and successfully tested in enhancing the signal-to-noise ratio, respectively. In this work, we conjunct both models and obtain a temporal and lateral inhibition system, which successfully suspends interferences. An evaluation of the proposed approach verifies the consistency with psychoacoustical experiments of the precedence effect and modulation perception; the resulting speech intelligibility benefits considerably in acoustically adverse situations.

**4pPPc36. Tinnitus: one problem that can be solved.** G. Mario Mattia (Brüel Acoustics S.r.l. - EuroAcustici, Viale Cesare Pavese 304, 00144 Roma, Italy, m.mattia@euroacustici.org)

The author explains the origin of idiopathic or subjective tinnitus. After more than ten years of research (at a personal centre and at Rome University "La Sapienza" Faculty of Medicine ...) we have solved more than 80% of tinnitus problems for a thousand people. We have found that the perceived sound can be a source of stress and can be recorded in the brain as "dangerous". We illustrate that a psychological and neurological rehabilitation implemented in a multifactors approach can give health and offer a new normal quality of life. With biofeedback EEG, EMG and GSR we are implementing our therapy. The biofeedback helps people understand the effects of stress on tinnitus and how to manage this stress to optimize performance and improve health. Previous studies have shown an enhancement of human performance and faster rehabilitation when physiological measures (respiration rate, heart rate, skin conductance, temperature, and surface electromyography) were fed back in sessions of Autogenic Training.

**4pPPc37. Subjective evaluation of accelerating car interior noise using brain magnetic field.** Kenji Takami (Univ. of Hyogo, 2167 Shosha, 671-2201 Himeji, Japan, hyogo.takami@aist.go.jp), Shunsuke Ishimitsu (Hiroshima City University / National Institute of Advanced Industrial Science and Technology, 3-4-1 Ozuka-Higashi, Asa-Minami-Ku, 731-3194 Hiroshima, Japan, ishimitsu@hiroshima-cu.ac.jp), Seiji Nakagawa (National Institute of Advanced Industrial Science and Technology (AIST), 1-8-31 Midorigaoka, 563-8577 Ikeda, Osaka, Japan, s-nakagawa@aist.go.jp), Toshihiko Asami (Univ. of Hyogo, 2167 Shosha, 671-2201 Himeji, Japan, asami@eng.u-hyogo.ac.jp)

Recently, the production concept of car engine sound has been changing to the sound design from the cure against noise. Although most work has concentrated on comfortable engine sound, the psychoacoustic effects of time-varying rate of frequency in accelerating engine sound have not been sufficiently studied. Thus we investigated the effects of time-frequency-changing rate in car interior noise on auditory impression using psychological and neurophysiological methods. Harmonic complex tones which simulated accelerating car noise were used as stimuli. Frequencies of the stimuli were time-varying by 15, 25, 35, 50, and 70 Hz/s, like accelerating-engine sounds. First, subjective evaluations were examined using a SD method. Second, neuronal activities of the auditory cortex evoked by these stimuli were measured by magnetoencephalography (MEG). The results indicated that time-varying rate of frequency has significant effects on subjective impression and neuronal activities of the auditory cortex.

**4pPPc38. Subjective experiment on auditory localization for traffic alarm sounds.** Sakae Yokoyama (Chiba Institute of Technology, 2-17-1, Tsudanuma, Narashino, 275-0016 Chiba, Japan, sakae@iis.u-tokyo.ac.jp), Hideki Tachibana (Chiba Institute of Technology, 2-17-1, Tsudanuma, Narashino, 275-0016 Chiba, Japan, tachibana@acoust.cs.it-chiba.ac.jp)

When riding a motorbike with a helmet or driving a car, it is often experienced that the judgment of directions of alarm sounds like a horn and sirens of engine truck and ambulance car becomes difficult. This kind of auditory localization is seriously related to traffic safety. In this study, therefore, the effects of covering the driver's head with a helmet and sound transmission into a car on the auditory localization for traffic alarm sounds



were examined by laboratory tests. To simulate a 3-dimensional sound field, the 6-channel recording/reproduction technique was applied and the incident direction of the test sounds was varied in twelve horizontal directions.

**4pPPc39. Development of pure-tone auditory threshold in school children.** Reinhard Müller (University of Giessen, Aulweg 123, 35392 Giessen, Germany, reinhard.mueller@audio.med.uni-giessen.de)

The overwhelming majority of publications concerning hearing in children is related to diseases, but normal development of hearing attracts little attention. Normal hearing, as defined by ISO 7029, refers to persons at the age of 18. While aging effects of hearing may be estimated by the use of formulas, children are not included. A field-study at a primary school in Germany showed a notably lower hearing sensitivity for children than for young adults. First graders did not hear well, but auditory performance improved with rising age. For validating this result, the first graders of the field study were again tested 3 years later. The second tests showed the expected improvement of their hearing threshold. So the hearing sense starts not with the full capability but underlies a certain development. Maybe training effects are the key to understanding this topic, as in other human skills.

**4pPPc40. Problems in sound quality evaluation in Brazil: general or cultural ones?** Stephan Paul (Lab. of Vibrations and Acoustics, Fed. Univ. of Santa Catarina, CTC, EMC, Campus Trindade, 88040 Florianópolis, Brazil, stephan.paul.acoustic@gmail.com)

Within modern sound engineering the individual perception of sound events has to be considered. Within sound and vibration related research all over the world several instruments were already developed, but this instruments are subject to several restrictions, especially their language. When sound evaluations are to be made in Brazil adequate evaluation instruments are necessary. This article is intended to discuss some of the problems encountered by the author and its co-workers when developing assessment procedures and especially instruments for evaluation of sound quality with Brazilian subjects. Especially the work undertaken to study descriptors for sound and vibration and the problems resulting in this work will be discussed. We consider the present congress, with participants from all over the world, to be a suitable platform to discuss some of the problems and results obtained.

THURSDAY AFTERNOON, 3 JULY 2008

P2-C, LEVEL 2, 3:40 TO 5:20 P.M.

### Session 4pPPd

## Psychological and Physiological Acoustics: Auditory Perception of Sound Source Properties II (Poster Session)

John Neuhoff, Cochair  
*The College of Wooster*

Anna Preis, Cochair  
*Institute of Acoustics, Adam Mickiewicz University*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pPPd1. Auditory-guided reaching movements in the peripersonal frontal space.** Florian Dramas (IRIT, Univ. Toulouse 3 - INPT - Univ. Toulouse 1 - CNRS Equipe Diamant, Univ. Paul Sabatier, 31062 Toulouse, France, florian.dramas@irit.fr), Brian F. Katz (LIMSI-CNRS, B.P. 133, 91403 Orsay, France, brian.katz@limsi.fr), Christophe Jouffrais (IRIT, Univ. Toulouse 3 - INPT - Univ. Toulouse 1 - CNRS Equipe Diamant, Univ. Paul Sabatier, 31062 Toulouse, France, jouffrais@irit.fr)

Previous studies on auditory localization processes have shown that humans can localize sound sources accurately, including distance in certain situations. Few studies have examined auditory localization by binaural mechanisms in the peripersonal space. Numerous studies have examined auditory localization through verbal report or various pointing movements. This study examines the precision of hand "reaching" movement towards an auditory object. An experimental platform (semicircle, radius 1m) was constructed with 35 small loudspeakers placed under an acoustically transparent grid. Blindfolded subjects were seated within the platform at table height. Test protocol consisted of a brief audio stimulus presented via a single loudspeaker followed by the subject placing their index finger (preferred hand) at the location of the sound object. Optical finger tracking was used during the course of the experiment. Two test variables were investigated: different au-

dio stimuli, Gaussian noise bursts varying the number and the duration of each burst; room acoustic conditions, with and without acoustical damping for reflection suppression. Preliminary results show precision of localization does not grow indefinitely with the number of burst repetitions but reaches a limit. Azimuth precision remains accurate, even with short burst conditions, contrary to the distance perception which increases with the stimuli duration.

**4pPPd2. In-situ observation of the perceptive process linked to dashboard tapping sounds.** Francois Montignies (Renault Technocentre, FR TCR LAB 252, 1 avenue du Golf, 78288 Guyancourt Cedex, France, francois.montignies@insa-lyon.fr), Valery Nosulenko (Institute of Psychology, Russian Academy of Science, 129366 Moscow, Russian Federation, valery.nosulenko@gmail.com), Etienne Parizet (Laboratoire Vibrations Acoustique, Insa Lyon, 25 bis, av. J. Capelle, 69621 Villeurbanne Cedex, France, etienne.parizet@insa-lyon.fr)

It is well-known that in show-rooms some people might tap on the dashboard of vehicles. The aim of this study was to determine the importance of this phenomenon and to identify which properties of the vehicle are perceived through the sound thus produced. An ethomethodological observation



was conducted to collect data about the action/perception process of a subject exploring a static vehicle. The work was based on the methodology developed by Nosulenko and Saymolenko to evaluate perceived quality using free verbalisations in a comparison task. 52 naive subjects were placed in ecologically valid conditions. Their task consisted in freely exploring two vehicles and selecting their preferred one. From a qualitative analysis of audiovisual recordings, a data base was built. It linked verbalisations, operations and perceived objects, and allowed the quantification of indicators related to activity and perception. The analysis of operations validated that the tapping operation was not anecdotal. Moreover, dashboard was one of the main perceived objects linked to the auditive dimension. Finally, a significant effect of the tapping operation on the evaluation of dashboard material quality was observed, suggesting an implicate influence of sound on this perceived property.

**4pPPd3. Hemispheric Differences in the Recognition of Environmental Sounds.** Julio Gonzalez (Universitat Jaume I, Dept. Psicologia Basica, Clinica y Psicobiologia, Campus Riu Sec. Facultad CC. Humanas y Sociales, 12071 Castellon de la Plana, Spain, gonzalez@psb.uji.es), Conor T. McLennan (Cleveland State University, Dpt. Psychology, 2121 Euclid Ave. CB 175, Cleveland, OH 44115, USA, c.mclennan@csuohio.edu)

In the visual domain, Marsolek and colleagues have provided support for their claim that two dissociable and parallel neural subsystems underlie abstract and specific object recognition [Marsolek, 1999; Marsolek & Burgund, 2003]. According to their dissociable subsystems theory, an abstract-category subsystem operates more effectively in the left hemisphere (LH) and is less sensitive to the specific surface characteristics of the stimuli, whereas a specific-exemplar subsystem operates more effectively in the right hemisphere (RH) and is more sensitive to specific stimulus characteristics. In the present study, we tested this hypothesis in the auditory domain by conducting 2 long-term repetition-priming experiments on the recognition of environmental sounds. Participants attempted to identify target sounds from an initial 750 ms sound stem. Target stems were primed by either an identical or a different exemplar sound (e.g., the same or different tokens of a bagpipe). Target stems were presented monaurally in both experiments; however, in Exp. 2 white noise was simultaneously administered to the opposite ear. Our results are consistent with Marsolek's framework. In particular, in both experiments an exemplar specificity effect was obtained when the sounds were presented to the left ear (RH), but not when the sounds were presented to the right ear (LH).

**4pPPd4. Perception of speech properties from extremely brief segments.** Sue Harding (Sheffield University, Computer Science Department, Regent Court, 211 Portobello St., S1 4DP Sheffield, UK, s.harding@dcs.shef.ac.uk), Martin Cooke (Sheffield University, Computer Science Department, Regent Court, 211 Portobello St., S1 4DP Sheffield, UK, m.cooke@dcs.shef.ac.uk)

A glance at a visual scene enables observers to become rapidly aware of its most important characteristics. Here, we describe experiments using very brief segments of natural speech which demonstrate that a surprising amount of information can be determined from only a few milliseconds of the auditory signal. Segments with durations ranging from 2.5 to 80 ms were extracted from six vowels and six fricatives spoken by males and females. Listeners identified the phoneme and/or gender, or whether a vowel or consonant had been presented. While listeners' performance dropped close to chance for the 2.5 ms stimuli for most tasks, for the vowel/fricative distinction listeners obtained scores above 70% even for such short segments. Listeners performed well above chance for the 10 ms stimuli for three out of four tasks. Combining results within tasks showed that listeners also distinguished voiced from unvoiced phonemes in less than 10 ms. Threshold values from logistic fits indicate the order in which information becomes available: vowel/fricative distinction (3.0 ms), voicing distinction (6.7 ms),

phoneme identification (11.9 ms) and gender identification (15.3 ms). By exploiting the "gist" of an auditory scene, listeners may be able to deploy prior knowledge rapidly to constrain further interpretation.

**4pPPd5. Comparison of headphones and equalization for virtual auditory source localization.** David Schonstein (Arkamys, 5 rue Frédéric Bastiat, 75008 Paris, France, dschonstein@arkamys.com), Laurent Ferré (LIMSI-CNRS, B.P. 133, 91403 Orsay, France, laurent.ferre@limsi.fr), Brian F. Katz (LIMSI-CNRS, B.P. 133, 91403 Orsay, France, brian.katz@limsi.fr)

This study investigates the variation in localization performance between different headphone styles. Eight different headphones (including various in-ear, circumaural open and closed, and bone conduction headphones) were tested. In addition, the effect of headphone equalization (aiming to produce an approximately flat frequency response) was investigated. Localization was examined for 24 locations distributed on a sphere surrounding the listener. A single subject participated in the study using a single chosen non-individual HRTF set. Each location was repeated 6 times, resulting in a total of 144 localization reports. Overall, results were relatively consistent for 3 out of the 8 headphones tested. For these headphones, there was no significant difference in lateral angle error, associated with ITD and ILD cues. Polar angle errors, associated with the cone of confusion, however did vary significantly for these headphones. The headphone equalization had varying effects on localization accuracy depending on the headphone. Globally, headphone equalization showed no significant effect on localization accuracy. The results serve as a preliminary investigation, highlighting consistent results for only a select group of headphones tested for effective sound rendering in virtual auditory space. In addition, the results suggest that headphone equalization has a minimal influence on localization accuracy under these conditions.

**4pPPd6. Perception of Sound Source Distance and Loudness in a Coherent Field of a Reverberant Field.** Yoshifumi Hara (Kogakuin University, 1-24-2, Nishi-Shinjuku, Shinjuku-ku, 185-0012 Tokyo, Japan, kuro5hiron@gmail.plala.or.jp), Yoshinori Takahashi (Kogakuin University, 1-24-2, Nishi-Shinjuku, Shinjuku-ku, 185-0012 Tokyo, Japan, yoshinori@ieee.org), Hiroaki Nomura (Kure National College of Technology, 2-2-11, Aga-Minami Kure City, 737-8506 Hiroshima, Japan, hnomura@kure-nct.ac.jp), Mikio Tohyama (Waseda University, 1-3-10, Bldg. 29-7, Nishi-Waseda, Shinjuku-ku, 169-0051 Tokyo, Japan, m\_tohyama@waseda.jp), Kazunori Miyoshi (Kogakuin University, 1-24-2, Nishi-Shinjuku, Shinjuku-ku, 185-0012 Tokyo, Japan, miyoshi@cc.kogakuin.ac.jp)

Perception of reverberant sound field changes with a sound source distance (SSD). This article describes SSD perception in a coherent region close to the sound source in the reverberation field. We performed listening tests for SSD perception and loudness of speech and carried out transfer functions analysis using a reverberant room. We confirm that both SSD perception and loudness are correlated to the standard deviation of the magnitude frequency response of the transfer function in the coherent region. That is SSD perception and loudness increase as SSD becomes long in the coherent region. However loudness decreases in an incoherent region. Consequently, we surmise SSD perception in the coherent region might be due to increase in loudness.

**4pPPd7. The influence of pinna position on head-related transfer function.** Przemyslaw Plaskota (Wroclaw University of Technology, Wybrzeze Wyspianskiego 27, 50-370 Wroclaw, Poland, przemyslaw.plaskota@pwr.wroc.pl), Andrzej B. Dobrucki (Wroclaw University of Technology, Wybrzeze Wyspianskiego 27, 50-370 Wroclaw, Poland, andrzej.dobrucki@pwr.wroc.pl)

The changes of spectrum of sound at listener ear are one of the major cues for sound source localization. Head Related Transfer-Function (HRTF) describes the influence of torso, head and pinna on sound spectrum. It is possible to recognize HRTF using computational method, e.g. Boundary Elements Method (BEM). The numerical model used for calculation of HRTF is constructed by transfer geometrical shape of head and pinna into numerical domain. Important question during geometry reconstruction process is

the accuracy of shape mapping. The pinna has significant influence on HRFT. In the paper, the influence of accuracy of pinna geometry transformation and pinna position on HRFT is presented. Particularly, the pinna flare angle, pinna rotation angle and position of ear entrance were taken into considerations. Measurements have been done on numerical model with the invariable pinna and head shapes, using BEM method.

**4pPPd8. Environmental Enrichment Increases Response Strength And Paired-Pulse Depression Of Auditory Cortex Neurons.** Cherie R. Percaccio (Univ. of Texas, 800 W. Campbell Rd, Richardson, TX 75080, USA, cheriep@u.washington.edu), Autumn L. Pruette (Univ. of Texas, 800 W. Campbell Rd, Richardson, TX 75080, USA, autumn.pruette@utsouthwestern.edu), Shilpa T. Mistry (Univ. of Texas, 800 W. Campbell Rd, Richardson, TX 75080, USA, shilpa.mistry@utsouthwestern.edu), Yeting H. Chen (Univ. of Texas, 800 W. Campbell Rd, Richardson, TX 75080, USA, superhelen@gmail.com), Daniel L. Rathbun (Univ. of Texas, 800 W. Campbell Rd, Richardson, TX 75080, USA, dlrathbun@ucdavis.edu), Michael P. Kilgard (Univ. of Texas, 800 W. Campbell Rd, Richardson, TX 75080, USA, kilgard@utdallas.edu)

A wide variety of sensory gating impairments have been associated with autism. Abnormal brain development may alter patterns of interaction between the child and the environment and hinder the acquisition of critical language skills. After several months of therapy, autistic symptoms may subside as children advance to higher cognitive stages. This study modeled the physiological changes associated with therapy-related gains in children by investigating enrichment-induced plasticity in rat auditory cortex. Evoked potential response strength and paired-pulse depression were enhanced by exposure to an enriched environment and degraded by exposure to a standard environment. While neither exercise nor social stimulation, specifically, resulted in any plasticity, rats that heard the enriched environment from a distance also exhibited enhanced responses. The degree of enrichment-induced plasticity was not reduced by a substantial and persistent cholinergic deficit. The finding that enrichment increases response strength and paired-pulse depression in the auditory cortex of rats is consistent with earlier clinical observations, suggesting that proper sensory development is necessary for higher cognitive processes. In the future we will investigate if clinical gains during and after therapy are associated with increased event-related potential discrimination and hemispheric localization of speech stimuli in children with autism.

**4pPPd9. Directional loudness measurements for a multichannel system.** Densil Cabrera (University of Sydney, Faculty of Architecture, Design and Planning, NSW 2006 Sydney, Australia, densil@usyd.edu.au), Luis Miranda (University of Sydney, Faculty of Architecture, Design and Planning, NSW 2006 Sydney, Australia, lmir9852@mail.usyd.edu.au), Ian Dash (Australian Broadcasting Corporation, Technology Research & Standards, Level 11, Ultimo Building, 2001 Sydney, Australia, Dash.Ian@abc.net.au)

Loudness matching listening tests were conducted to quantify the difference in loudness level from a constant signal played from various horizontal directions. The multichannel system used for this tests was a 5-channel system, set up according to the ITU Recommendation BS.116-1 Methods for the subjective assessment of small impairments in audio systems including multichannel sound systems and the test signals were octave bands of noise with centre frequencies from 63 Hz to 8000 Hz. These tests were conducted as part of ongoing research for the ITU Recommendation BS.1770 Algorithms to measure audio programme loudness and true-peak level. The aim of this experiment is to contribute to the design of a loudness meter by providing channel weightings, and results indicate that listeners are more sensitive to the surround channels than the other channels in the mid and high frequency range.

**4pPPd10. Investigating the potential of human echolocation in virtual sonic trigonometry.** Flaithri E. Neff (University College Cork, Western Road, IRL Cork, Ireland, fn2@cs.ucc.ie), Ian Pitt (University College Cork, Western Road, IRL Cork, Ireland, i.pitt@cs.ucc.ie)

Describing a mathematical problem often involves visual diagrams. For blind students this accentuates the challenges they face. Projects such as LAMBDA have used linear speech and Braille to convey algebraic equations. However, spatial features, for example in trigonometry, are difficult to map to a linear-based system. Traditional tactile methods (e.g. German film) convey simple shapes but need Braille support and speech-tactile interfaces (e.g. NOMAD) require unconventional equipment. Cognitive issues regarding tactile interpretation of 3D shapes also persist. Blind students interact regularly with speech technology and audio games. This exposure means that the auditory system is potentially becoming accustomed to sonic interpretation of computer-based information. Some of our research has looked at expanding the sonic environment to include spatial information aimed at trigonometry. The next stage is to provide interactive user control. Our system is based on a user interface model in order to consider the cognitive issues involved. We use Microsoft's XNA/XACT environment to create our auditory scene. In this paper we discuss how to implement sonic-based user interaction while further simplifying our auditory scene. In order to achieve this, we examine the potential of human echolocation to orient within the virtual walls and corners of a triangle.

**4pPPd11. Toward synthesis tools using 'evocation' as control parameters.** Adrien Merer (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, merer@lma.cnrs-mrs.fr), Mitsuko Aramaki (CNRS - INCM and Université de Provence, 31, chemin Joseph Aiguier, 13402 Marseille, France, aramaki@lma.cnrs-mrs.fr), Richard Kronland-Martinet (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, kronland@lma.cnrs-mrs.fr), Solvi Ystad (CNRS-LMA, 31, chemin Joseph Aiguier, 13402 Marseille, France, ystad@lma.cnrs-mrs.fr)

This study addresses the design of synthesis tool controlled by high-level parameters, such as mental evocations induced by sounds. As a first approach, we considered sounds evoking motions and we addressed 3 main questions: What are the different categories of motion? What are the common acoustic features of sounds within a category? How to synthesize sounds that evokes specific motions? We gathered samples used by electro-acoustic music composers as a framework for their compositions and synthesized sounds. Then we effectuated a two-steps listening test. The first part aims at determining these different motion categories. It consisted in a free categorization task in which listeners build their own groups of sounds as function of evoked motions. The second part aims at determining a set of sounds characteristic of each of these categories. It consisted in a constrained categorization task with predefined categories represented by prototypical sounds (deduced from free categorization task). We used a feature selection method to highlight most relevant signal descriptors for each category. Finally, designing a synthesis tool implies the calibration of these descriptors (a specific range of values for each category) and their control (leading to address the inverse problem). These aspects are currently being investigated.

**4pPPd12. Human recognition by active and passive acoustic signatures.** Alexander Ekimov (University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677, USA, aekimov@olemiss.edu), James M. Sabatier (University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677, USA, sabatier@olemiss.edu)

Recognition of different sensed objects is a problem that often appears in practice. One of the solutions is based on analysis of the signatures of the specific objects. This method was applied for the acoustic detection of walking humans. Human footsteps excite envelopes of broadband acoustic signals in the air due to periodic friction forces between the foot and the ground/floor. The repetition frequency of these envelopes is equal to the footstep rate and usually lies below 3 Hz. High frequencies in these envelopes allow detection and localization of a walker using a narrowband ultrasonic receiver with a high directivity pattern. Consequently, periodic low frequency human motion results in passive ultrasonic detection using this

method. This motion has also unique Doppler signatures and is measured using ultrasonic sonar. Common analyses of passive and Doppler signatures allow the extraction of the specific cadence in human motion and recognition of a human while rejecting other moving or stationary objects. [Work supported by Department of the Army, Army Research Office contract W911NF-04-1-0190].

**4pPPd13. Strategic listener movement in a model of auditory distance perception.** Yan-Chen Lu (Sheffield University, Computer Science Department, Regent Court, 211 Portobello St., S1 4DP Sheffield, UK, y.c.lu@dcs.shef.ac.uk), Martin Cooke (Sheffield University, Computer Science Department, Regent Court, 211 Portobello St., S1 4DP Sheffield, UK, m.cooke@dcs.shef.ac.uk)

A mobile listener has the potential to exploit dynamic auditory cues to judge sound source distance. One such cue is motion parallax, which employs a sequence of azimuth estimates from interaural time differences to

triangulate sound source location. However, distortions due to reverberation and competing sources complicate matters, so it is of interest to know what active strategies listeners might adopt to arrive at robust location estimates. One hypothesis is that not all listener motion trajectories are equally-beneficial for distance estimation. Trajectories designed via certain optimisation criteria might lead to faster and more robust estimates in a wider range of environments. Eight listener motion strategies were tested, from naive approaches such as random walks and head-rotation only to more sophisticated techniques based on sequential Monte Carlo methods. In the latter case, strategies included movement towards the expected source location, or in the most informative direction, or movement reducing overall uncertainty. Evaluation in a simulated acoustic environment with single sources under both anechoic and reverberant conditions demonstrated that moving towards the most likely source location led to the most accurate estimation of distance and subsequent tracking of a moving source. Significant problems remain in estimating distance in multi-source conditions.

THURSDAY AFTERNOON, 3 JULY 2008

P2-B, LEVEL 2, 3:40 TO 5:20 P.M.

### Session 4pSAa

## Structural Acoustics and Vibration, Computational Acoustics, and EURONOISE: Efficient Boundary Element Methods II (Poster Session)

Ramani Duraiswami, Cochair

*University of Maryland Institute for Advanced Computer Studies*

Haike Brick, Cochair

*TFH Berlin - University of Applied Sciences*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pSAa1. Application of the fast multipole method for solving very large acoustic radiation problems.** Raphael Hallez (LMS International, Researchpark Z1, Interleuvenlaan 68, 3001 Leuven, Belgium, raphael.hallez@lmsintl.com), Koen De Langhe (LMS International, Researchpark Z1, Interleuvenlaan 68, 3001 Leuven, Belgium, koen.delanghe@lmsintl.com), Michel Tournour (LMS International, Researchpark Z1, Interleuvenlaan 68, 3001 Leuven, Belgium, michel.tournour@lmsintl.com), Toufic Abboud (IMACS, Ecole Polytechnique, 91128 Palaiseau Cedex, France, abboud@imacs.polytechnique.fr)

Boundary element method is well known and extensively used to solve acoustic radiation problems. It is especially appropriated for exterior radiation since the fluid domain does not need to be meshed, as opposed to the finite element method. However, the mathematical formulation leads to a dense matrix system of equations. Therefore, the size of the model increases drastically as the frequency of analysis increases and huge computer resources are required to solve complex models in the mid-frequency range. The fast multipole method can be used to extend the boundary element model and solve such problems. For a model with  $N$  nodes, this technique brings the number of operations down to  $O(N \log N)$  instead of  $O(N^3)$  for conventional boundary elements. This new methodology has been applied here to study the acoustic radiation of a complete car in the mid-frequency range. The accuracy of the results as well as the computation time demonstrate the great potential of this new method to solve very large acoustic radiation problems.

**4pSAa2. FE-Model Reduction for FE-BE Coupling with Large Fluid-Structure Interfaces.** Michael Junge (Institute of Applied and Experimental Mechanics, University of Stuttgart, Pfaffenwaldring 9, 70550 Stuttgart, Germany, junge@iam.uni-stuttgart.de), Jens Becker (Institute of Applied and Experimental Mechanics, University of Stuttgart, Pfaffenwaldring 9, 70550 Stuttgart, Germany, becker@iam.uni-stuttgart.de), Dominik Brunner (Institute of Applied and Experimental Mechanics, University of Stuttgart, Pfaffenwaldring 9, 70550 Stuttgart, Germany, brunner@iam.uni-stuttgart.de), Lothar Gaul (Institute of Applied and Experimental Mechanics, University of Stuttgart, Pfaffenwaldring 9, 70550 Stuttgart, Germany, gaul@iam.uni-stuttgart.de)

For the finite element method model, reduction techniques exist to represent the dynamic behavior of component substructures. Depending on the type of reduction method, the reduction basis contains constraint or attachment modes, which are computed for all structural degrees of freedom on an interface. The interface can either be defined by adjacent substructures or by coupling interfaces to other physical domains, as it is the case for FE-BE coupled systems. A large interface thus leads to an increased size of the reduced order model and limits standard model reduction techniques to applications with small interfaces. In this work, interface reduction methods are investigated. Here, the size of the reduced order model is decreased by reducing the number of retained interface modes, while marginally increasing the reduction error. A direct reduction method based on strain-energy considerations is presented. Additionally, an iterative reduction scheme is pro-

posed which only adds a basis vector to the reduction basis, if the spanned subspace is sufficiently enlarged. The applicability of the proposed methods is shown for an example structure.

**4pSAa3. Modeling of the acoustic eigenproblem with sound absorption using boundary element method.** Antoine Lavie (Univ. d'Artois, Faculté des Sciences Appliquées - Lab. LAMTI, Technoparc Futura, 62400 Bethune, France, antoine.lavie@univ-artois.fr), Alexandre Leblanc (ISAE, 10, av Edouard Belin, 31055 Toulouse, France, alexandre.leblanc@isae.fr), Abdelkader Haddi (Univ. d'Artois, Faculté des Sciences Appliquées - Lab. LAMTI, Technoparc Futura, 62400 Bethune, France, abdelkader.haddi@univ-artois.fr)

This paper deals with determination of resonant frequencies for absorbant 3D acoustic cavities. The behaviour of the sound absorbing boundary can be described with a Robin condition as proposed by Rajakumar et al. [Int. J. Numer. Methods Eng., 36, 3957-3072 (1993)]. This approach is inaccurate, especially for low frequencies because the absorption coefficient is assumed to be constant. We observed the acoustic admittance for foam and fibrous type materials varies linearly for low frequencies. The introduction of a new absorption coefficient allows to take into account this behavior in order to improve the accuracy in the determination of the first modes (typically less than 500 Hz in car interior). This formulation has been implemented in a boundary element program we have developed. The results are compared with those given by the finite element program ANSYS. Computations are carried out for rectangular parallelepiped and Sedan car interior.

THURSDAY AFTERNOON, 3 JULY 2008

P2-B, LEVEL 2, 3:40 TO 5:20 P.M.

## Session 4pSAb

### Structural Acoustics and Vibration and EURONOISE: Distributed Active Noise and Vibration Control II (Poster Session)

Kenneth Cunefare, Cochair  
*Georgia Institute of Technology*

Manuel Collet, Cochair  
*FEMTO-ST UMR CNRS*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

#### Contributed Papers

**4pSAb1. Active noise control in turbofan aircrafts: theory and experiments.** Ernesto Monaco (Dept. of Aerospace Engineering - University of Naples, Via Claudio, 21, 80125 Naples, Italy, ermonaco@unina.it), Leonardo Lecce (Dept. of Aerospace Engineering - University of Naples, Via Claudio, 21, 80125 Naples, Italy, leonardo@unina.it), Ciro Natale (Dipartimento di Ingegneria dell'Informazione - Seconda Università degli Studi di Napoli, Via Roma 29, I-81031 Aversa (CE), Italy, ciro.natale@unina2.it), Salvatore Pirozzi (Dipartimento di Ingegneria dell'Informazione - Seconda Università degli Studi di Napoli, Via Roma 29, I-81031 Aversa (CE), Italy, salvatore.pirozzi@unina2.it), Chris May (Laboratory of Process Automation (LPA) - Saarland University, Gebäude A5 1, D-66123 Saarbrücken, Germany, c.may@zip.uni-sb.de)

This paper presents the activities developed by the authors within the research project M.E.S.E.M.A. funded by the European Commission. A noise and vibration control system using magnetostrictive actuators has been designed and experimentally tested on a large scale (fuselage mock-up) test article, for controlling noise and vibrations between 150 - 500 Hz. The environmental noise and vibration excitation was representative of a small/medium turbofan aircraft. Firstly a numerical model of the test article has been developed in MSC/NASTRAN coupling the structural part with the interior acoustic volume. Furthermore the experimental characterisation of the test-article has been carried on. The model, updated by the mean of the experimental results, was employed to derive the required control actuators performances in order to achieve the best control predicted using a well consolidated "feed-forward" approach. Genetic Algorithms have been

employed in order to optimise the positioning of the actuators. Dedicated magnetostrictive actuators have been designed together with light power amplifiers meeting the specifications; on each actuator an optoelectronic sensor, based on Bragg grating, has been integrated to optimize the actuator performance. A two-level ANVC system has been designed and tested on a full scale fuselage mock-up. The paper present an overview of the activities developed as well as of the achieved results.

**4pSAb2. Active vibration control for the identification-based model of a circular plate.** Lucyna Leniowska (University of Rzeszow, Rejtana 16, PL35-310 Rzeszow, Poland, lleniow@univ.rzeszow.pl), Pawel Kos (University of Rzeszow, Rejtana 16, PL35-310 Rzeszow, Poland, pkos@univ.rzeszow.pl)

An active vibration control system is proposed for suppressing the small amplitude vibration of circular plate. An experimental set-up consists of a hard-walled cylinder with a thin metallic plate at one end. Primary excitation is provided by a low frequency loudspeaker installed centrally at the bottom of the cylinder. The vibrations of the plate are measured by the application of strain sensors and accelerometers. Intelligent materials such as 2-layer piezo disk elements are used as the actuators. For the considered system the OE (Output Error) method of discrete-time model identification for real-time active vibration control have been applied. The mathematical model obtained by this method identification is then employed for the linear pole placement controller design. Numerical simulations describing the attenuation effects are presented and discussed.



**4pSAb3. Adaptive predictive feedback control of circular plate vibrations.** Lucyna Leniowska (University of Rzeszow, Rejtana 16, PL35-310 Rzeszow, Poland, lleniow@univ.rzeszow.pl), Pawel Kos (University of Rzeszow, Rejtana 16, PL35-310 Rzeszow, Poland, pkos@univ.rzeszow.pl)

Adaptive predictive controller, consists of an on-line identification technique coupled with a control scheme, is used in this paper for a plate vibration suppression. It is assumed, that the system to be regulated is unknown and the control schemes presented have the ability to identify and suppress the plate vibrations with only an initial estimate of the system order. The choice of structure is motivated by its representative nature. This configuration has also been well studied by the authors both analytically and experimentally, using several kind of controllers (PID, PI2D, fuzzy, LQR). There are two fundamental steps involved in the closed-loop system. The first step is to identify a mathematical model. The second step is to use the identified model to design a controller. One drawback of this approach is that, the control signal is fed to the actuator after updates of the control law expression, which always leads to some delay. In order to align better the updating process, the authors introduce the prediction of plant output with established error convergence. The one-step ahead system output prediction is calculated from the recursive formulas of the interpolation functions chosen. Simulations are included and discussed.

**4pSAb4. Vibration control of flexible 3D robot arm with join and distributed actuators.** Ryszard Leniowski (Rzeszow University of Technology, W.Pola 2, PL 35-902 Rzeszow, Poland, lery@prz-rzeszow.pl)

An active vibration control system is proposed for suppressing amplitude vibration of flexible 3D robot arm. This system integrates control algorithms, intelligent materials and software technologies. The mathematical model of physical system is based upon the geometry and properties of an experimental set-up consisting of a Flex3D robot with a flexible joints and flexible arm. The tip of the arm is loaded by eccentric mass. The vibrations of the plate are measured by the application of a grid of strain sensors and pair of coupled gyroscope-accelerometer. Two kinds of actuators are used. The first is a grid of PZT elements which form a local segments of compensators. Second is a standard BLDC motor located in the join. For the considered system the linear and non-linear (Neural Network of Runge-Kutta type models) of discrete-time model identification for real-time active vibration control have been applied. The mathematical model obtained by this method identification is then employed for the two class of controllers: linear pole placement controller for local segments compensators and non-linear reduced model reference for servo-controller. Virtual simulations are included and discussed.

THURSDAY AFTERNOON, 3 JULY 2008

P2-B, LEVEL 2, 3:40 TO 5:20 P.M.

### Session 4pSAc

## Structural Acoustics and Vibration, Computational Acoustics, and EURONOISE: Fluid-Structure Interaction III (Poster Session)

Noureddine Atalla, Cochair  
*Univ. de Sherbrooke*

Vicente Cutanda Henriquez, Cochair  
*Institute of Sensors, Signals and Electrotechnics, University of Southern Denmark*

Stefan Schneider, Cochair  
*Laboratoire de Mécanique et d'Acoustique, UPR 7051 CNRS*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pSAc1. Modified amplitude and strouhal number scaling for correction of turbulent wall pressure fluctuations.** Thomas Galib (NUWC, Howell St, Newport, RI 02841, USA, galibta@npt.nuwc.navy.mil)

Pressure fluctuations were measured in an external turbulent boundary layer over a buoyantly propelled axisymmetric body of revolution. Data were measured for three cases, resulting in axial length Reynolds numbers of  $6.88 \times 10^4$ ,  $4.27 \times 10^4$ , and  $3.21 \times 10^4$  at the measurement locations. The fresh water measurements were made in a fully developed turbulent boundary layer, following natural transition, with a near zero (very mildly adverse) pressure gradient. The salt water measurements were made in a favorable pressure gradient following a flow trip to force transition. The momentum thickness Reynolds number was greater than 4400 for all measurements, and the data were scaled with outer variables. The turbulence data were corrected using Corcos correction factors and then further scaled in both am-

plitude and frequency by the square root of (transducer radius/displacement thickness), in an attempt to resolve spatial resolution effects. This resulted in excellent agreement among the spectra to a modified Strouhal number of 1, which was the range of validity for the data. A second data set (constant freestream velocity and increasing displacement thickness with downstream measurement location), scaled somewhat better with the ratio of transducer radius to displacement thickness.

**4pSAc2. The stability of nonequilibrium supersonic boundary layer.** Igor Zavershinsky (Samara State Aerospace University, 34 Moskow Str., 443086 Samara, Russian Federation, zav@smr.ru)

The present work looks at the influence of molecular vibrational energy nonequilibrium on the instability of boundary layer. The boundary layer loses its stability at supercritical Reynolds numbers. At small Mach numbers

$M < 2$  can be unstable only Tollmien-Schlichting waves, which phase velocity in relation of mean flow is subsonic. The recent studies were shown that in the acoustically active gas the critical Reynolds number for subsonic disturbances is decreased in comparison with equilibrium media [1,2]. At large Mach numbers  $M > 2$  in addition to Tollmien-Schlichting waves we must take into account both falling and reflecting acoustic waves (Mach waves). As a result the interaction between vortical structure and acoustic waves leads to shift of critical Reynolds number in comparison to subsonic case. Here the dependence of critical Reynolds number on thin flat plane from degree of nonequilibrium for spatial supersonic disturbances is founded. It was shown that acoustical activity of media has a large destabilizing influence. The increment of this instability is raised with growth of the degree of non-equilibrium. 1. Bertolotti F.P. J. Fluid Mech., 1998, V. 372. P. 93. 2. Knestypin V.N., Kogan E.Ya., Zavershinsky I.P. High Temperatures. 2007. V.45. N2. P.1.

**4pSac3. Coupled wavenumbers of structural acoustic waveguides: A unified asymptotic approach.** Abhijit Sarkar (Indian Institute of Science, FRITA Lab, Dept. of Mechanical Engg., Indian Institute of Science, 560012 Bangalore, India, asarkar@mecheng.iisc.ernet.in), Venkata R. Sonti (Indian Institute of Science, FRITA Lab, Dept. of Mechanical Engg., Indian Institute of Science, 560012 Bangalore, India, sonti@mecheng.iisc.ernet.in)

In the literature, the coupled wavenumbers in flexible-walled acoustic waveguides have been found mainly using numerical methods for a fixed set of parameters. These solutions, although useful, do not continuously track the coupled wavenumbers as the fluid-loading parameter is varied from small to large values. Such a continuous tracking is possible by applying the asymptotic methods to the coupled dispersion relation. Analytical formulae for the coupled wavenumbers can also be found. In this work, we present a consistent, unified and physically insightful view of structural acoustic coupling in the context of a planar structural-acoustic waveguide (with two different BCs), the axisymmetric and beam modes of a fluid-filled circular cylindrical shell and finally a fluid-filled elliptic cylindrical shell. In all the five cases, we use a single fluid-structure-coupling asymptotic parameter ( $\epsilon$ ). The regular perturbation method is used to solve the coupled dispersion relation for small and large values of  $\epsilon$ . The circular and the elliptic cylinders necessitate the use of additional asymptotic parameters. A general trend in all systems is that a given wavenumber branch transits from a rigid-walled solution to a pressure-release solution with increasing  $\epsilon$ . Also, the wavenumber curves veer where the earlier uncoupled wavenumbers intersected.

**4pSac4. Boundary element model to study sound transmission provided by a single panel in the presence of an elastic interlayer.**

Andreia Pereira (University of Coimbra, Department of Civil Engineering - Pólo II da Universidade - Rua Luís Reis Santos, 3030-788 Coimbra, Portugal, apereira@dec.uc.pt), António Tadeu (University of Coimbra, Department of Civil Engineering - Pólo II da Universidade - Rua Luís Reis Santos, 3030-788 Coimbra, Portugal, tadeu@dec.uc.pt)

In this paper the Boundary Element Method is applied to develop a numerical model which may be used to study the airborne and impact sound insulation provided by a single panel of infinite extent bounded by two fluid media, with an elastic interlayer (joint) inserted into the partition. The elastic interlayer is placed in the panel, perpendicular to its surfaces and fully occupying the panel thickness. When the interlayer assumes the properties of a resilient material, it is able to reduce the wave propagation through the elastic medium of the partition. The model is excited by a harmonic line load that acts either in the fluid medium or in the elastic medium in the direction perpendicular to the panel's surface. The model is developed following a direct frequency domain formulation which assumes full coupling between the fluid media and the elastic media. Analytical Green's functions for an elastic single layer bounded by fluid media are used to avoid having to discretize the horizontal surfaces of the partition. Numerical simulations are displayed in order to illustrate the applicability of the proposed model to the analysis of airborne and impact sound insulation when an elastic interlayer is inserted into a single partition.

**4pSac5. Numerical analysis of airborne and impact sound insulation between non-contiguous acoustic spaces using the Boundary Element Method.**

Andreia Pereira (University of Coimbra, Department of Civil Engineering - Pólo II da Universidade - Rua Luís Reis Santos, 3030-788 Coimbra, Portugal, apereira@dec.uc.pt), António Tadeu (University of Coimbra, Department of Civil Engineering - Pólo II da Universidade - Rua Luís Reis Santos, 3030-788 Coimbra, Portugal, tadeu@dec.uc.pt), Paulo Santos (University of Coimbra, Department of Civil Engineering - Pólo II da Universidade - Rua Luís Reis Santos, 3030-788 Coimbra, Portugal, pfsantos@dec.uc.pt)

In this paper a Boundary Element Method (BEM) is used to predict the acoustic behaviour provided by four two-dimensional acoustic closed spaces separated by slabs and walls, surrounded by an elastic infinite medium. The walls and slabs are modelled as single partitions. The model is excited by cylindrical loads in the form of an airborne sound source placed in the acoustic space or an impact sound source acting on the slab, perpendicular to its surface. The formulation is developed following a direct BEM approach which assumes full coupling between the fluid medium and the elastic medium. The model requires the discretization of all interfaces and allows the analysis in the low and medium frequency range. A numerical analysis is performed to study airborne and impact sound insulation between acoustic non-contiguous spaces, where the sound pressure level which is established in the receiving room is due to flanking transmission. The influence of the structure's stiffness on the sound insulation is discussed for varying thicknesses of slabs and walls. The acoustic behaviour of the structure is described by sound insulation curves and average vibration velocity level curves of the walls and slabs and the results are discussed.

**4pSac6. Influence of temperature on sound transmission through viscoelastic sandwich plates.**

Samir Assaf (ESTACA, 34 rue Victor Hugo, 92300 Levallois Perret, France, sassaf@estaca.fr), Mohamed Guerich (ESILV, 92916 Paris la Défense, France, mohamed.guerich@devinci.fr)

A numerical study to investigate the effects of temperature on the diffuse sound transmission loss (TL) of sandwich plates is presented. The numerical prediction tool used is based on a finite element formulation for the sandwich plate coupled to a boundary element method for the acoustic medium. The plate formulation is derived from Kirchhoff's theory for the elastic faces and Mindlin's theory for the core. The frequency-temperature dependence of the viscoelastic material properties are taken into account using an experimentally derived viscoelastic constitutive law. The results presented deal with a laminated glass subjected to a diffuse sound field. It is found that the dip of the TL curve at the coincidence frequency of the plate is totally removed for temperatures where this frequency is in the transition region of the used viscoelastic material. Indeed, the relative low value of the storage modulus and the high value of the loss factor in this region induce high transverse shear deformations of the viscoelastic interlayer and thus high energy losses.

**4pSac7. A transmission loss definition based on the root mean squared intensity.**

Ysbrand H. Wijnant (University of Twente, Dept. Mechanical Engineering, P.O. Box 217, 7500 AE Enschede, Netherlands, Y.H.Wijnant@ctw.utwente.nl)

Numerical simulation enables the calculation of the sound power transmitted through any structure for any arbitrary source but it can not be used to determine transmission loss, which is based on the transmitted power divided by the incident power. One can calculate the active power (= incident minus reflected power), but one generally can not determine the incident and reflected power themselves. In fact, apart from 1 dimensional analysis, transmission loss is only defined for diffuse sound fields and it only quantifies the acoustical performance of the structure for this field and not for the actual, in-situ, source. While the incident power can not be determined, the active and root mean squared power can. The latter quantity is introduced and is a measure for the total energy flow per period, i.e. the net energy flow and the energy flowing back and forth. A transmission loss definition based on the ratio between the active power and the root mean squared power is

shown to closely resemble the transmission loss in the 1 dimensional case. Results for a 2 dimensional case show that the ratio is a quantity which does characterize the transmission of the structure as it is excited by the source.

**4pSAC8. Fast solutions in FSI-Problems using CMS-Methods.** Johannes Guggenberger (Mueller-BBM, Robert-Koch-Str. 11, 82152 Planegg, Germany, johannes.guggenberger@muellerbbm.de)

Parameter studies in FSI-problems may often become quite time consuming. In most cases the fluid parameters are well defined and only the influence of the parameters of the solid is subject to investigate. Therefore it would be desirable to investigate the fluid and solid part separately and fi-

nally combine them using CMS-Methods. This approach would also provide a good physical insight into the individual and combined behaviour of the components fluid and solid. Generally for each interface DOF one constraint mode must be added to the modal base. Since in most problems in FSI the fluid-structure interface involves many DOF the general CMS approach becomes inefficient. To reduce the number of constraint modes it is proposed to use the mode shapes of each component as a load function on the other domain. The static solution provides the modal based constraint modes. Their number corresponds to the number of total component modes which is in most cases much less than in the classic approach. The application is shown in optimization, updating, and monte-carlo-simulation problems.

THURSDAY AFTERNOON, 3 JULY 2008

ROOM 250B, 2:00 TO 3:40 P.M.

## Session 4pSCa

### Speech Communication: Articulatory Modeling and Control of Speech and Singing Organs

Maureen Stone, Cochair

*Vocal Tract Visualization Lab, Depts of Biomedical Sciences and Orthodontics, University of Maryland Dental School, 650 W. Baltimore St., Baltimore, MD 21201, USA*

Phil Hoole, Cochair

*Institut für Phonetik und Sprachverarbeitung, Schellingstr. 3, München, 80799, Germany*

### Invited Papers

2:00

**4pSCa1. Issues in the acoustic modeling of the vocal tract - a progress report on APEX.** Björn Lindblom (Stockholm University, Universitetsvägen 10C, SE10691 Stockholm, Sweden, lindblom@ling.su.se), Johan Sundberg (KTH, Department of Speech, Music and Hearing, Lindstedtsvägen 24, SE-100 44 Stockholm, Sweden, jsu@csc.kth.se)

Faced with an acoustic record of the human voice an investigator is often led to wonder: How was this sample produced? Direct articulatory measurements lacking, the researcher may nonetheless find information of value by resorting to numerical models that relate cavity shapes to acoustic parameters. The classical 'three-parameter models' exemplify this type of tool. The APEX model is another example, developed to help answering questions about the acoustic consequences of articulatory movements. Based on X-ray measurements from a single speaker, APEX converts input specifications for articulatory parameters such as jaw, lips, larynx, tongue tip and tongue body into formant frequencies. The introduction of independent control of the jaw and the tongue and the fact that possible tongue shapes are specified relative to a neutral reference tongue have made significant insights into various topics possible (e.g., co-ordination jaw/tongue in singing, compensatory articulations). Recently we have been able to increase the physiological realism of APEX representations using MRI and X-ray data. We have also investigated physical models to improve the treatment of certain 3-D front cavity configurations such as raised tongue blade and spread lips. The goal of our paper is to present an overview of the most recent version of APEX.

2:20

**4pSCa2. Articulatory comparison of spoken and sung vowels based on MRI.** Pierre Badin (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, pierre.badin@gipsa-lab.inpg.fr), Nathalie Henrich (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, Nathalie.Henrich@gipsa-lab.inpg.fr), Laurent Lamalle (Plateforme régionale IRM 3Tesla, IFR n° 1, RMN Biomédicale : de la cellule à l'homme, CHU de Grenoble, BP 217, 38043 Grenoble Cedex 9, France, Laurent.Lamalle@ujf-grenoble.fr)

Understanding the differences of articulatory strategies between spoken and sung vowels is of interest to both speech and singing research. We have thus used MRI to record midsagittal images from three subjects producing sustained vowels with various characteristics. The subjects were a professional lyric soprano, a semi-professional soprano, and a semi professional bass. They were instructed to produce combinations of (1) the ten French or the five Italian oral vowels, (2) speaking, amateur singing, or professional singing modes, (3) chest or falsetto registers, (4) pitch levels varying from B2 (120 Hz) to F5 (700 Hz). Any combination that the subject would not feel comfortable with was excluded from the corpus. The midsagittal contours of the vocal organs (jaw, lips, tongue, velum, pharyngeal wall, hyoid bone, etc.) were manually traced on each image, and a number of articulatory measurements (jaw or hyoid bone height, lip aperture, tongue position, etc.) were automatically derived. Our contribution analyses the influence of the various production conditions on these articulatory characteristics, such as the jaw aperture increase related to pitch increase, or the lower position of the larynx for singing in comparison to speech. Some acoustics considerations will be discussed as well.

**4pSCa3. Resonance characteristics of the hypopharyngeal cavities.** Kiyoshi Honda (Phonetics & Phonology Laboratories, UMR-7018-CNRS & University of Paris III, 19, rue des Bernardins, 75005 Paris, France, honda@atr.jp), Tatsuya Kitamura (Konan University, Okamoto 8-9-1, Higashinada, 658-8501 Kobe, Japan, t-kitamu@konan-u.ac.jp), Hironori Takemoto (ATR Cognitive Information Science Laboratories, 2-2-2 Hikaridai, Seika-cho Soraku-gun, 619-0288 Kyoto, Japan, takemoto@atr.jp), Parham Mokhtari (ATR Cognitive Information Science Laboratories, 2-2-2 Hikaridai, Seika-cho Soraku-gun, 619-0288 Kyoto, Japan, parham@atr.jp), Seiji Adachi (Fraunhofer Institute for Building Physics, Nobelstrasse 12, 70569 Stuttgart, Germany, seiji.adachi@ibp.fraunhofer.de)

The hypopharyngeal cavities are the narrow, complex parts of the lower vocal tract that include the supraglottal laryngeal cavity and bilateral cavities of the piriform fossa. These small regions exhibit rather strong acoustic influence on vowel spectra in the higher frequencies and contribute to determining voice quality and speaker characteristics. The laryngeal cavity functions nearly as a Helmholtz resonator to generate an extra formant in the vicinity of 3 kHz, and the piriform fossa forms a pair of side-branches of the vocal tract to cause spectral zeros in the vicinity of 4-5 kHz. Vocal-tract models without employing these acoustic effects can hardly simulate natural-sounding voices of a particular speaker. Therefore, realistic acoustic models of vowel production must include the three functional components: glottal source sounds, hypopharyngeal-cavity coupling, and resonance of the vocal-tract proper. This presentation demonstrates the results from acoustic experiments on solid vocal-tract models and computer simulations of the cavities' effects based on our MRI-based visualization of the vocal tract. A possible control for singing voice qualities will be discussed based on the three-component model.

**4pSCa4. Observation of voice registers.** Ken-Ichi Sakakibara (Health Sciences University of Hokkaido, 2jo-5chome, Kita-ku, 002-8072 Sapporo, Japan, quesokis@gmail.com), Hiroshi Imagawa (Department of Otolaryngology, University of Tokyo, 7-3-1, Hongo, Bunkyo-ku, 113-8655 Tokyo, Japan, imagawa@m.u-tokyo.ac.jp), Miwako Kimura (Department of Otolaryngology, University of Tokyo, 7-3-1, Hongo, Bunkyo-ku, 113-8655 Tokyo, Japan, mkimu-ty@umin.ac.jp), Isao Tokuda (Japan Advanced Institute of Science and Technology, 1-1 Asahidai, Nomi-shi, 923-1292 Ishikawa, Japan, isao@jaist.ac.jp), Takaharu Nito (Department of Otolaryngology, University of Tokyo, 7-3-1, Hongo, Bunkyo-ku, 113-8655 Tokyo, Japan, taka.nito@nifty.com), Niro Tayama (International Medical Center of Japan, 1-21-1, Toyama, Shinjuku-ku, 162-8655 Tokyo, Japan, ntayama@imcj.hosp.go.jp)

In singing, voice register is one of the most salient aspects of voice quality, and it has therefore generated a lot of debates, acoustically, physiologically, and pedagogically. In singing, the voice registers can be physiologically classified into four categories: vocal fry, modal, falsetto, and whistle. In this study, vocal fold vibratory patterns appeared in each register were observed using high-speed images and simulated using the two-mass model. In vocal fry, three different vibratory patterns (aperiodic, subharmonic, and periodic with small open quotient) were observed. In addition, the simulation showed that transitions between the three different vibratory patterns are easy. In whistle, closure of the posterior part of glottis and rapid vibration of the anterior part were observed.

**4pSCa5. Voice production modes and vocal tract shape in South-Siberian throat-singing.** Sven Grawunder (Max Planck Institute for Evolutionary Anthropology, Deutscher Platz 6, 04103 Leipzig, Germany, grawunder@eva.mpg.de)

South-Siberian throat-singing features reinforced harmonics as carrier of sung melodies and enforced phonation modes. Available articulatory studies of throat-singers suggest that throat-singing makes use of three voice production mechanisms which result in two basic voice modes (mid tensed vs. low rough). Thus all three mechanisms share an excessive constriction of the larynx entrance i.e. approximation of the aryepiglottic folds and the epiglottis. The current study comprises acoustic data from 69 male singers. 25 singer were recorded by use of a field setting for acoustic, electroglottographic and subglottal resonance signal acquisition. Perturbation measures show dominance of individual variability over areal (cultural) factors, but strong influence of articulatory reinforcement strategies. The data also provide evidence for a model of reinforcement of harmonics by means of (1) voice source variation (closing phase, excitation strength), i.e. increased subglottal pressure, while air flow remains constant or lowered for the tensed mode; and double cycle modes involving mass bodies of upper laryngeal structures for the low mode; (2) formant merging due to multiple vocal tract constrictions including a coupling of source to the adjacent epilaryngeal tube of 1/6 vocal-tract length and bandwidth tuning as a result of adjustment of lip radiation.



## Session 4pSCb

## Speech Communication: General Topics in Speech Communication III (Poster Session)

Christine Shadle, Cochair  
Haskins Laboratories

Sharon Coffey-Corina, Cochair  
Center for Mind and Brain UC Davis

Bart De Boer, Cochair  
Amsterdam, Netherlands

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pSCb1. Feature extraction for vowel recognition using mellin transform.** Mahdi Jamaati (Technical University of Shahrood, 12345 Shahrood, Iran, mahdi.jamaati@gmail.com), Milad Lankarany (Technical University of Shahrood, 12345 Shahrood, Iran, milad.lankarany@gmail.com), Hossein Marvi (Technical University of Shahrood, 12345 Shahrood, Iran, h\_marvi@shahroodut.ac.ir)

This paper proposed a new feature extraction algorithm for vowel recognition using Mellin transform and MFCC. The scale transform is a particular restriction of the Mellin transform. The key property of the scale transform is the scale invariance. This algorithm consists of 3 main stages which are as follow. First stage contains, Extraction spectral envelope from vowels, by using cepstrum method. In the second stage, Mellin transform is used for mapping same vowels with different pitch and different vocal tract to the same waveforms. In third stage, we used a new MFCC algorithm in order to extract main parameters from the output of second stage. The new MFCC algorithm contains third and fourth order cumulant of log-mel-amplitudes. This new MFCC coefficients are robust in presence of additive white noise. Moreover, The LPC method has been used instead of MFCC method in third stage, Experimental result indicates that the new MFCC algorithm produced better result than LPC and original MFCC.

**4pSCb2. Empirical articulatory-acoustic relations for vowels.** Richard S. McGowan (CRESS LLC, 1 Seaborn Place, Lexington, MA 02420, USA, rsmcgowan@earthlink.net), Michael A. Berger (Dept. of Linguistics, Univ. of Rochester, 503 Lattimore Hall, Rochester, NY 14627-0096, USA, m.a.berger@sms.ed.ac.uk)

Vowels tokens were extracted from four talkers in the Wisconsin X-ray Microbeam Speech Production Database. The neighboring phonemes of these vowels were restricted to be non-nasal and non-liquid. The first three formant frequencies were measured using LPC analysis with manual corrections at a rate corresponding to the pellet trajectory sampling rate, thus yielding large amounts of simultaneous formant frequency and pellet position data points (between 11,000 and 20,000 for each talker.) Principal components analysis was performed for both the formant frequencies and the pellet position data, to produce three orthogonal acoustic components and four orthogonal articulatory components. A local linear regression technique, known as loess [Cleveland, W. S. and Devlin, S. J. (1988), J. Amer. Stat. Assoc., 83, 596 - 610], was applied to orthogonal components to map between the acoustic and articulatory domains. This technique permits regression slopes to vary within the domain of the independent variables. The results will be discussed in terms of optimization of loess parameters (e.g.

size of local neighborhoods), goodness of fit of the mappings, and the degree to which slopes in the mappings vary. Manually corrected formant frequencies will be compared with fully automatic Line Spectral Frequencies. [Supported by NIDCD-001247 to CRESS LLC]

**4pSCb3. Cross-language study of age perception by elderly listeners.** Kyoko Nagao (Center for Pediatric Auditory and Speech Sciences, Nemours Biomedical Research, A.I. du Pont Hospital for Children, 1600 Rockland Road, Wilmington, DE 19803, USA, nagao@asel.udel.edu), Amanda K. Riley (Indiana University, Department of Speech and Hearing Sciences, 200 S Jordan Avenue, Bloomington, IN 47405, USA, mandieriley@gmail.com)

Recent studies show that perception of speaker's age is influenced by a listener's familiarity with the speaker's language. However, these results were based on young adult listeners. The current study examines whether language familiarity influences the perception of speaker's age in elderly listeners. The vowel stimuli was prepared from the sustained vowel /i/ collected from 30 native speakers of American English and 30 native speaker of Japanese. The sentence stimuli was taken from the reading of the North Wind and the Sun by these 60 speakers (in their native language). Fifteen elderly native speakers of English (mean age 73.5 years, range from 67 to 84 years) listened to both stimuli types and estimated the age of speakers. Correlation between perceived age and chronological age was moderate for the sentence stimuli, but weak for the vowel stimuli. Correlation between perceived age and chronological age was stronger when the listeners judged sentence stimuli in their native language than when they judged sentence stimuli in the foreign language ( $r=0.74$  versus  $r=0.63$ ). The vowel stimuli did not show the effect of speaker language. The results suggest that linguistic information has a critical role in age perception, regardless of listener's age.

**4pSCb4. Acoustic tubes with maximal and minimal resonance frequencies.** Bart De Boer (Spuistraat 210, 1012VT Amsterdam, Netherlands, b.g.deboer@uva.nl)

This paper presents a theoretical derivation of acoustic tract shapes that minimize and maximize resonance frequencies. The derivation is based on a symmetry of Webster's horn equation and on Ehrenfest's adiabatic invariance hypothesis. It is shown that for minimizing formant frequencies, abrupt transitions are necessary, while for maximizing resonance frequencies, gradual transitions are needed. It is argued that this has implications for modeling human, animal and prehistoric vocal tracts. Such models should

represent the anatomic (in)ability to produce abrupt and gradual transitions correctly, otherwise they would have biases towards different sets of formant frequencies than real vocal tracts.

**4pSCb5. Rhythmic characteristics of prose and verse in varieties of Portuguese.** Zinny Bond (Ohio University, 18 Maplewood Drive, Athens, OH 45701, USA, bond@ohio.edu), Verna Stockmal (Ohio University, 384 Gordy Hall, Athens, OH 45701, USA, grannymeem@hotmail.com), Emilia A. Marks (Ohio University, Department of Modern Languages, Athens, OH 45701, USA, markse@ohio.edu), Audra Woods (Ohio University, Speech and hearing sciences, Athens, OH 45701, USA, aw365402@ohio.edu)

Whether varieties of Portuguese differ in their rhythmic classification is not entirely clear. European Portuguese is generally considered to employ stress-based rhythm whereas the rhythmic classification of Brazilian Portuguese is disputed. Acoustically-based measures of rhythm have usually employed spoken prose passages for language samples. Possibly, the apparent rhythmic characteristics of language varieties may be clarified by employing different types of language materials. We selected spoken prose and traditional verse for investigation. Five native speakers of European Portuguese and five native speakers of Brazilian Portuguese recorded a sonnet consisting of 154 syllables and a short prose passage consisting of 270 syllables. From these recordings, acoustically based rhythm metrics, as suggested by Ramus, et al. [Cognition 73, 1999, 265-292] and Ling, et al. [Language and Speech 43, 2000, 377-401] were calculated. Measures of consonantal duration variability increased from spoken prose to spoken verse, suggesting that the talkers employed slower, more clearly articulated speech in reading the poem than in reading the prose passage. However, the two tasks did not clearly distinguish between the two varieties of Portuguese.

**4pSCb6. Experimental study of turbulent flow sound production in presence of a simplified vocal tract constriction.** Olivier Estienne (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, olivier.estienne@gipsa-lab.inpg.fr), Annemie Van Hirtum (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, annemie.vanhirtum@gipsa-lab.inpg.fr), Xavier Pelorson (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, pelorson@icp.inpg.fr), Hélène Bailliet (Laboratoire d'Etudes Aérodynamiques - CNRS, Bat K, 40 avenue du recteur Pineau, 86022 Poitiers, France, helene.bailliet@lea.univ-poitiers.fr)

Sound production due to turbulence is widely shown to be an important phenomenon involved in a.o. fricatives, singing, whispering and speech pathologies. In spite of its relevance, only a few recent models are dealing with turbulence consequences during voiced sound production. The current study presents preliminary results of an experimental characterisation of the aeroacoustical effects of a turbulent flow in case of a constricted channel flow, by means of measuring the velocity fluctuations and the acoustic field downstream of the constriction. Aiming a future application in speech production, the influence of typical vocal tract shape parameters on the velocity distribution and sound field is explored: the tube shape and length as well as the degree, geometry and position of the constriction. The influence of the Reynolds number of the upstream flow is also observed. Results are discussed with respect to the upper airways and human sound production and will be exploited with respect to simplified models for velocity fluctuations and sound production.

**4pSCb7. Influences of perceived racial identity on human talker identification.** Tyler K. Perrachione (Massachusetts Institute of Technology, 77 Massachusetts Avenue, Cambridge, MA 02139, USA, tkp@mit.edu), Joan Y. Chiao (Northwestern University, 2029 Sheridan Road, Evanston, IL 60208, USA, jchiao@northwestern.edu), Patrick C. Wong (Northwestern University, 2240 Campus Drive, Evanston, IL 60208, USA, pwong@northwestern.edu)

Talker identification, the process by which human listeners recognize individuals by their voice, is one of the most poorly understood abilities of the human auditory system. Current psychological models of talker identification rely on strict analogies to face perception and the visual system, despite

differences in the objects of perception in these two modalities. Here we investigate the existence of an own-race bias in voice perception - a phenomenon which has been pivotal in the study of face perception. Our results demonstrate an own-race bias in talker identification: listeners of different ethnic backgrounds show an advantage for identifying individual voices of the same race as themselves. However, unlike in vision, the own-race bias in talker identification manifests based specifically on the perceived, but not actual, race of a talker. The influence of perceived race suggests physical (voice structural) cues do not give rise to this effect. Instead, the own-race bias in talker identification is a result of listeners' asymmetric exposure to talkers' socially-acquired manners of expression (i.e. the dynamic features of voice and speech). Such manners of expression may be stereotypically associated with a particular ethnic group, although not actually exhibited by all members of that group. [Work supported by NIH]

**4pSCb8. Signal densities and criterion variance in speech and nonspeech perception.** Luis E. Lopez-Bascuas (Universidad Complutense Madrid, Facultad Psicología, Campus Somosaguas, 28223 Madrid, Spain, lelopezb@psi.ucm.es)

The actual shape of signal densities has become an important issue when studying speech perception within the framework of Signal Detection Theory (SDT). Using an SDT model that allowed unequal criterion variances, López-Bascuas [Proc. Eurosp. 3, 2281-2283 (1995)] found that speech signals did not accommodate to the standard Gaussian assumption. However, Schouten and van Hensen [J. Acoust. Soc. Am. 104, 2980-2990 (1998)] measured response distributions directly and, assuming an interval scale, concluded that the Gaussian assumption held for both continua. Nevertheless, Pastore and Macmillan [J. Acoust. Soc. Am. 111, 2432 (2002)] applied ROC analysis to Schouten and van Hensen's data and their curves supported the Gaussian assumption for the nonspeech signals only. Later, López-Bascuas et al. [J. Acoust. Soc. Am. 115, 2465 (2004)] showed that non-linear z-transformed ROCs are not sufficient evidence for postulating non-gaussian signal densities. In this paper we try to figure out whether unequal criterion variances could underlie non-linear z-transformed ROCs by fitting a restricted Thurstonian SDT-like model to a nonspeech continuum composed by white noise and a square wave. The results indicate that unequal criterion variances is the possible cause of the deviant ROCs for nonspeech.

**4pSCb9. Perception of reduced speech: Approximated stops.** Natasha Warner (University of Arizona, MPI Nijmegen, Box 210028, Dept. of Linguistics, Tucson, AZ 85721-0028, USA, nwarner@u.arizona.edu), Tucker V. Benjamin (University of Alberta, Dept. of Linguistics, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca), Amy Fountain (University of Arizona, MPI Nijmegen, Box 210028, Dept. of Linguistics, Tucson, AZ 85721-0028, USA, avf@u.arizona.edu)

Natural, spontaneous speech often shows extreme reductions of many speech segments, to the point of apparent deletion. Where the flap allophone of /t/ and /d/ is expected in American English, one frequently sees an approximant-like or even vocalic pattern, rather than a clear flap. Still, such tokens are usually perceived as containing a /t/ or /d/ (e.g. 'needle' even with a very reduced /d/ is usually not perceived as 'kneel'). The current work identifies acoustic characteristics of reduced 'flaps' and presents phonetic identification data for continua that manipulate these characteristics. Presence vs. absence of a dip in intensity, duration of that dip, and degree of that dip in decibels are manipulated. Degree of intensity dip has the strongest effect, with a minimal dip in intensity more likely to be perceived as 'kneel' and a strong dip in intensity more likely to be perceived as 'needle.' The results indicate that all three of these characteristics do affect listeners' percept of a consonant, but not sufficiently to completely account for the percept. Listeners are sensitive to the acoustic characteristics of consonant reduction, but they are also very skilled at evaluating variability along the acoustic dimensions that realize reduction.

**4pSCb10. Amplitude modulation shape and speech intelligibility.** Garreth Prendergast (The University of York, Heslington, YO10 5DD York, UK, garreth.prendergast@yonic.york.ac.uk), Sam R. Johnson (The University of York, Heslington, YO10 5DD York, UK, sam@yonic.york.ac.uk), Gary G. Green (The University of York, Heslington, YO10 5DD York, UK, gary.green@yonic.york.ac.uk)

Research has demonstrated that low frequency amplitude modulations in speech signals are crucially important to maintaining intelligibility. The current work demonstrates a flexible way of characterising the pulsatile bursts of energy found in the temporal envelopes of sub-band filtered speech. Speech was passed through a 128-filter Gammatone filterbank and the temporal envelope of each filter extracted using the Hilbert transform. Thirty-five raised cosine pulses were fitted to model the envelope of each filter and each pulse was defined by its amplitude, half-duration and centre position. The distribution of these pulses demonstrates that the most commonly found pulse half-duration in speech is around 10 ms and few pulses have half-durations longer than 25 ms. Highly intelligible vocoded speech is generated using the extracted pulses and these measures suggest that the auditory system may signal the position in time of the amplitude modulations rather than representing low-frequency information. This method creates a flexible framework within which to further probe the mechanisms involved and allows the ability to focus on cross-channel information in the time domain.

**4pSCb11. Training English vowels for French speakers with varying English experience.** Paul Iverson (University College London, Department of Phonetics and Linguistics, 4, Stephenson Way, NW1 2HE London, UK, p.iversen@ucl.ac.uk), Melanie Preece-Pinet (University College London, Department of Phonetics and Linguistics, 4, Stephenson Way, NW1 2HE London, UK, m.pinet@ucl.ac.uk)

It is clear that high-variability phonetic training can improve speech perception for adult second-language learners, but it is uncertain what levels of processing are responsible for this change and whether this interacts with the previous experience of the learners. The present study investigated these issues by giving auditory training to French speakers with varying degrees of English experience. Listeners completed 8 sessions of high-variability identification training with clusters of minimal-pair words (e.g., sleet, slit, slight, slate) and a different speaker in each session. The pre/post tests evaluated changes in vowel perception using open-set identification in quiet and noise, a 3-interval category discrimination task, and a task in which individuals mapped their best exemplars for English vowels in a space that included formant movement and duration. The preliminary results suggest that French speakers improved substantially in their identification performance (about 20 percentage points), without having large changes in their ability to discriminate these categories. The results support a view that training improves the process of applying phonetic category knowledge, without changing the underlying categories themselves.

**4pSCb12. Computerized assessment and training of the perception of American English (AE) speech sounds by adult learners of English.** James Miller (Communication Disorders Technology, Inc., Indiana University Research Park, 501 N. Morton Street Sta 215, Bloomington, IN 47404, USA, jamdmill@indiana.edu), Roy Sillings (Communication Disorders Technology, Inc., Indiana University Research Park, 501 N. Morton Street Sta 215, Bloomington, IN 47404, USA, rsillings@comdistec.com), Charles S. Watson (Communication Disorders Technology, Inc., Indiana University Research Park, 501 N. Morton Street Sta 215, Bloomington, IN 47404, USA, watson@indiana.edu), Isabelle Darcy (Indiana University, Dept. of Second Language Studies, Memorial Hall 315, 1021 E. Third Street, Bloomington, IN 47405, USA, idarcy@indiana.edu), Kathleen Bardovi-Harlig (Indiana University, Dept. of Second Language Studies, Memorial Hall 315, 1021 E. Third Street, Bloomington, IN 47405, USA, bardovi@indiana.edu)

Fourteen volunteer students, with first languages other than English and enrolled in English pronunciation classes, used a specialized software program, the "Speech Perception Assessment and Training System (SPATS)." The software was made available in a language laboratory for seven weeks. Students used the program between 3 and 19 hours, the mean being 13 hours. The goal of SPATS is to train the perception of English words as they

are produced in naturally spoken, fluent sentences. The average student correctly identified 1723 words in 313 different sentences as spoken by 10 different talkers and also attempted to identify a total of 7530 presentations of syllable onsets, nuclei, or codas as spoken by 8 different talkers in several phonetic contexts. These syllable constituents were presented in quiet, the sentences in moderate amounts of 12-talker babble. All showed significant progress in perception of AE speech sounds, a few approaching the levels of native speakers. By extrapolation, with 20 to 30 hours of program use nearly all might approach the perceptual performance of native speakers. Those that returned post-training questionnaires viewed the program favorably and recommended its use in regular classes. Speech samples were collected. Acoustic ratings and spectrographic analyses will be presented.

**4pSCb13. Integrated magnetic resonance imaging methods for speech science and technology.** Shinobu Masaki (ATR Cognitive Information Science Laboratories, 2-2-2 Hikaridai, Seika-cho Soraku-gun, 619-0288 Kyoto, Japan, masaki@atr.jp), Yukiko Nota (National Institute of Information and Communications Technology/ATR Cognitive Information Science Laboratories, 2-2-2 Hikaridai, Seika-cho, 619-0288 Kyoto, Japan, ynota@atr.jp), Sayoko Takano (ATR Cognitive Information Science Laboratories, 2-2-2 Hikaridai, Seika-cho Soraku-gun, 619-0288 Kyoto, Japan, tsayoko@atr.jp), Hironori Takemoto (ATR Cognitive Information Science Laboratories, 2-2-2 Hikaridai, Seika-cho Soraku-gun, 619-0288 Kyoto, Japan, takemoto@atr.jp), Tatsuya Kitamura (Konan University, Okamoto 8-9-1, Higashinada, 658-8501 Kobe, Japan, t-kitamu@konan-u.ac.jp), Kiyoshi Honda (Phonetics & Phonology Laboratories, UMR-7018-CNRS & University of Paris III, 19, rue des Bernardins, 75005 Paris, France, honda@atr.jp)

This presentation introduces our integration of magnetic resonance imaging (MRI) techniques at ATR Brain Activity Imaging Center (Kyoto, Japan) toward research into speech science and technology. The first breakthrough in our application of MRI to speech research was the motion imaging of the speech organs in articulation using a cardiac cine-MRI method. It enables us to acquire information in the time-space domain to reconstruct successive image frames using utterance repetitions synchronized with MRI scans. This cine-technique was further improved for high-quality imaging and expanded into three-dimensional (3D) visualization of articulatory movements. Using this technique, we could successfully obtain temporal changes of vocal-tract area function during a Japanese five-vowel sequence. This effort also contributed to developing other techniques to overcome the limitations of MRI, such as the post-hoc inclusion of teeth images in 3D volumes or the phonation-synchronized scan for crystal-sharp static imaging. Further, a custom high-sensitivity coil was developed to visualize the fine structures of the lip muscles and laryngeal airway. The potentials of new MRI approaches such as ultra-high-resolution imaging with a higher-field scanner or real-time motion imaging during a single utterance will be discussed toward future contributions to speech science and technology.

**4pSCb14. Acoustic characteristics of solid vocal tracts modeled from ATR MRI database of Japanese vowel production.** Tatsuya Kitamura (Konan University, Okamoto 8-9-1, Higashinada, 658-8501 Kobe, Japan, t-kitamu@konan-u.ac.jp), Hironori Takemoto (ATR Cognitive Information Science Laboratories, 2-2-2 Hikaridai, Seika-cho Soraku-gun, 619-0288 Kyoto, Japan, takemoto@atr.jp), Kiyoshi Honda (Phonetics & Phonology Laboratories, UMR-7018-CNRS & University of Paris III, 19, rue des Bernardins, 75005 Paris, France, honda@atr.jp)

"ATR MRI database of Japanese vowel production" provides volumetric magnetic resonance images and speech data of the five Japanese vowels produced by a male native Japanese speaker. In this study, the database was used to evaluate acoustic characteristics of vocal tracts for five Japanese vowels; we measured frequency responses of realistic vocal tract solid models formed by a stereo-lithographic technique. The model's glottis was sealed with a plastic plane with a 1.2-mm hole. A time-stretched pulse signal generated from a horn driver unit was introduced into the solid model at the lip end. The response signals of the models were recorded at the model's glottis from the hole by a probe microphone. This method permits accurate measurement of acoustic characteristics of the vocal tract including the laryngeal cavity, which generates the laryngeal cavity resonance during closed-glottis periods of phonation [H. Takemoto et al., JASA, 120, 2228-



2238 (2006), T. Kitamura et al., JASA, 120, 2239-2249 (2006)]. The results provide a benchmark for evaluating numerical analysis methods, such as transmission line models, finite element methods, and finite difference-time domain-methods, which have been used to study vocal tract acoustics. [Work supported by SCOPE (071705001) of Ministry of Internal Affairs and Communications, Japan.]

**4pSCb15. Linguistic versus non-linguistic processing of speech prosody in dichotic listening.** Ritsu Kanamura (Graduate School of Comprehensive Scientific Research, Prefectural Univ. of Hiroshima at Mihara, 1-1 Gakuen-machi, 723-0053 Mihara, Japan, bulle.de.savon.von.von@gmail.com), Satoshi Imaizumi (The Prefectural Univ. of Hiroshima, 1-1 Gakuen-machi, 723-0053 Mihara, Japan, imaizumi@pu-hiroshima.ac.jp)

Linguistic and non-linguistic information processing of speech prosody are studied using two dichotic listening tasks, a Word task and a F0 task. During the Word task, subjects were required to identify either right- or left-ear stimulus from two-syllable homophonic words presented with different pitch accents. During the F0 task, subjects were required to identify either right- or left-ear stimulus from F0 partials extracted from the words used in the Word task. The correct percent of responses was high and RT was short for high familiarity words presented to the right ear rather than the others under the Word task, while no such differences were found under the F0 task. RT of the Word task was shorter than the F0 task. These results suggest that the processing of linguistic speech prosody under dichotic listening conditions is based on the interactive auditory and linguistic neural resources with right-ear or left-hemispheric dominance, and are faster than auditory F0 pattern identification. The tasks developed can be used to detect central auditory processing disorders.

**4pSCb16. Broader range of training voices improves performance of HMM model of phonemic identification.** J Parchment (University of Arizona, Dept. of Linguistics, Douglass Building, Room 200E, Tucson, AZ 85721, USA, jparch@u.arizona.edu)

A model proposed by Lin (2005) learns phonetic categories from wave-form input. Recorded speech from a set of male talkers is divided into training and test sets. The training set is separated into phonemes, subjected to cepstral analysis, and used as the input to a Hidden Markov Model, which clusters the phonemes into phonemic categories. After this unsupervised learning process, the model is then able to accurately identify speech segments in the test set, showing that relevant acoustic information is captured by the model. The current study explores the outcome when a model of this type is trained on a range of talkers differing in sex and vocal tract configuration. Preliminary results suggest that this approach can improve performance when testing is generalized to a wider range of new talkers. However, too wide a range of training voices reduces accurate categorization, while too narrow a range reduces generalizability. Continuing efforts seek to quantify the optimum range of training voices and to identify the variables that can predict the degree of improvement in performance on test voices. This work has implications for automatic speech recognition models as well as for issues of speaker normalization.

**4pSCb17. A Method of Co-registering Multiple Magnetic Resonance Imaged Vocal Tract Volumes for Fricatives.** Michael I. Proctor (Haskins Laboratories, 300 George St, New Haven, CT 06511, USA, proctor@haskins.yale.edu), Christine Shadle (Haskins Laboratories, 300 George St, New Haven, CT 06511, USA, shadle@haskins.yale.edu), Khalil Iskarous (Haskins Laboratories, 300 George St, New Haven, CT 06511, USA, iskarous@haskins.yale.edu)

In a study of fricative production, Magnetic Resonance Imaging was used to acquire three-dimensional tract geometries of fricatives produced in four different vowel contexts. Short scan sequences were used to image multiple productions of each fricative in a single session. Although minimizing the number of tokens, this resulted in a coarser spatial resolution in some parts of the tract, which required a different approach to assembling vocal tract volumes. Sagittal, axial and oblique-coronal imaging orientations were chosen to best resolve the tract in the mid-sagittal plane, the pharynx, and around the fricative constriction. Each fricative token was acquired using a 37 second imaging sequence. Dental impressions were imaged separately.

Three corresponding point clouds were created by sampling air-tissue boundaries in each image stack, and aligned using anatomical landmarks on the face and spinal column, creating a single, multiply-sampled volume. Area functions were estimated using the most suitable data source at each region of the tract. Vocal tract models were constructed for eight fricatives produced in four vowel contexts by four speakers of American English. The advantages of this method of tract modeling, and its automatic application to potentially variable MRI data sources will be discussed. [Funded by NIH-NIDCD-RO1-DC006705]

**4pSCb18. Acoustical analysis of Canadian and Parisian French word-final vowel productions in varying phonetic contexts.** Franz Law (CUNY Graduate Center, 365 5th Avenue, Program in Speech-Language-Hearing Sciences, New York, NY 10016, USA, flaw@gc.cuny.edu), Winifred Strange (CUNY Graduate Center, 365 5th Avenue, Program in Speech-Language-Hearing Sciences, New York, NY 10016, USA, strangein@aol.com)

Dialects can differ greatly in their phonetics and phonology, and can diverge to a great degree in vowel production. This study explores the acoustic nature of Canadian French (CF) vowels in word-final position, relative to those of Standard Parisian French (PF). CF and PF male participants were recorded producing minimal sets of words, differing in the final vowels /i y e "epsilon" a "o-bar" o u/, embedded in carrier phrases. Real-word minimal sets were constructed using words with final vowels preceded by labial, coronal, and back stops and fricatives. (Monosyllabic CV words were used whenever possible; VCV and CCV words were used when no CV word was available.) Nonsense disyllabic minimal sets were also recorded with vowels preceded by labial, coronal, and back stops in the following context: /gisCV/. Of particular interest was the stability of the /e - "epsilon"/ distinction in word-final position for Parisian French, which was maintained in preliminary results. Across-syllable boundary coarticulatory effects are also discussed. This is the first part of a larger study investigating second-language lexical and morphosyntactic CF vowel perception by English learners of French. [Work supported by NIH F31DC008075]

**4pSCb19. An MRI Study of the Effect of Vowel Context on English Fricatives.** Christine Shadle (Haskins Laboratories, 300 George St, New Haven, CT 06511, USA, shadle@haskins.yale.edu), Michael I. Proctor (Haskins Laboratories, 300 George St, New Haven, CT 06511, USA, proctor@haskins.yale.edu), Khalil Iskarous (Haskins Laboratories, 300 George St, New Haven, CT 06511, USA, iskarous@haskins.yale.edu)

To gain a better understanding of the long-observed effects of vocalic context, the articulation of fricatives was investigated using Magnetic Resonance Imaging. Five speakers of American English were imaged while producing eight fricatives in the contexts [i-a-u-schwa]. Sagittal, axial and oblique-coronal volumes were acquired for each vowel-fricative combination. Acoustic recordings were made during scans and separately in an anechoic chamber. Vocal tract models were generated by aligning and superimposing all three stack orientations. The models reveal that a variety of articulatory strategies are employed in the production of English fricatives, and that vocalic context is significant. For some subjects, tongue shape differs little with vowel context; other subjects show highly varied tongue shape differences but little difference in lip rounding. Two subjects show significant variation with vowel context for every fricative, including [θ]; two show very little difference, even for [f]. The sublingual cavity in [sh] is extensive for two subjects, but only slightly bigger than in [s] for two other subjects. Tongue grooving in and behind the constriction also varies, and is especially evident in the oblique scans. Overall, more variation and effect of vowel context was observed than expected from the literature. [Funded by NIH-NIDCD-RO1-DC006705]

**4pSCb20. F0 contour estimation based on time-varying complex speech analysis.** Keiichi Funaki (Univ. of the Ryukyus, Senbaru 1, Nishihara, 903-0213 Okinawa, Japan, funaki@cc.u-ryukyu.ac.jp)

Robust F0 (Fundamental frequency) estimation plays an important role in speech processing such as speech coding and tonal speech recognition. We have already proposed robust F0 estimation algorithm based on time-varying complex AR (TV-CAR) speech analysis for analytic signal, in



which the weighted autocorrelation function is calculated using the complex residual and then the F0 is searched as the peak sample for each frame [IE-ICE Trans. on Fundamentals, Vol.E91-A.No.3] [IEICE Trans. on Fundamentals, Vol.E90-A, No.8]. Although the algorithm can estimate more robust F0 estimation for IRS filtered speech corrupted by additive noise, the algorithm cannot perform better for non-IRS filtered speech or slightly contaminated IRS-filtered speech. In addition, the frame-based F0 estimation cannot extract the F0 trajectories in the time-domain. In order to cope with the drawbacks, this paper proposes quite simple F0 contour estimation algorithm based on the TV-CAR speech analysis, in which the F0 contour is estimated by peak-picking for the estimated time-varying spectrum that is the same manner as formant frequency estimation. The experimental results demonstrate that the proposed method leads to more accurate continuous F0 estimation than the conventional one for high-pitched speech due to the nature of analytic signal for non-IRS filtered high-pitched female speech.

**4pSCb21. Extraction of vocal tract area function from three-dimensional magnetic resonance images using digital waveguide mesh.** Kenji Inoue (Osaka Institute of Technology, 4-18-15, Tanabe, Higashiumiyosi-ku, 546-0031 Osaka, Japan, chikuwa.bushi@gmail.com), Hironori Takemoto (ATR Cognitive Information Science Laboratories, 2-2-2 Hikaridai, Seika-cho Soraku-gun, 619-0288 Kyoto, Japan, takemoto@atr.jp), Tatsuya Kitamura (Konan University, Okamoto 8-9-1, Higashinada, 658-8501 Kobe, Japan, t-kitamu@konan-u.ac.jp), Shinobu Masaki (ATR Cognitive Information Science Laboratories, 2-2-2 Hikaridai, Seika-cho Soraku-gun, 619-0288 Kyoto, Japan, masaki@atr.jp), Hirotake Nakashima (Osaka Institute of Technology, 1-79-1, Kitayama, Hirakata, 573-0196 Osaka-fu, Japan, nakas@is.oit.ac.jp)

A method is proposed in this paper to extract the vocal tract area function from the three-dimensional magnetic resonance images. The proposed method uses the digital waveguide mesh, an implementation of the finite-difference time-domain (FDTD) method, to simulate wave propagation in the vocal tract from the glottis to the lips. The dimensions of the vocal tract areas are then calculated along the traveling wavefront that emerges from the simulation. Formant analysis has been conducted for Japanese vowels to show the validity of the proposed method. The calculated formant frequencies of the area functions obtained by the proposed method and other existing methods have been compared to those measured from the acoustic utterance of the imaged person.

**4pSCb22. Increasing speech alignment through crossmodal speaker familiarity.** Rachel M. Miller (University of California, Riverside, Department of Psychology, 900 University Ave., Riverside, CA 92521, USA, rmill002@ucr.edu), Kaayumari Sanchez (University of California, Riverside, Department of Psychology, 900 University Ave., Riverside, CA 92521, USA, ksanc004@student.ucr.edu), Lawrence D. Rosenblum (University of California, Riverside, Department of Psychology, 900 University Ave., Riverside, CA 92521, USA, rosenblu@citrus.ucr.edu)

In speech alignment phenomena, individuals inadvertently imitate aspects of another talker's utterances. Recent research has shown that when asked to shadow words, subjects not only align to the speech they hear, they also align to the speech they see when shadowing words by lipreading [Miller, et al., J. Acoust. Soc. Am., 120, 5, Pt. 2 (2006)]. This research also showed that some of the dimensions to which subjects align are the same whether based on shadowing of auditory or visual speech stimuli. This might mean that subjects can align to a speaker's idiolectic dimensions available in both modalities. To examine this possibility, an experiment was conducted to see if alignment increased with exposure to the same or a different speaker, across two blocks of presentations that were: a) both auditory; b) both visual; or c) one auditory and one visual. If subjects align to amodal, idiolectic speaker style, then alignment should be comparable across presentation types in the same speaker condition. Results revealed that alignment increased when the speaker was the same over the course of

the two blocks regardless of presentation type. These results suggest that alignment can be based on amodal, idiolectic dimensions which are available across modalities.

**4pSCb23. Perception of Japanese consonants by native speakers of American English.** Takeshi Nozawa (Ritsumeikan University, College of Economics, 1-1-1 Nojihigashi, Kusatsu, 525-8577 Shiga, Japan, t-nozawa@ec.ritsumei.ac.jp)

Four native speakers of Japanese produced Japanese multi-syllable words and non-words, which include stops, fricatives, affricates, nasals and the liquid. Twelve native speakers of American English, who had never learned Japanese, heard their utterances and spelled out what they heard in English alphabet. What was of interest was whether native speakers of American English would perceive Japanese consonants in a way they are transcribed in English alphabet. The results revealed that Japanese stops were equated with generally the phonetically closest English stops, but voiceless stops were more likely to be equated with voiced stops of the same place of articulation than the other way around. Among voiced stops, /b/ and /d/ respectively were equated with /v/ and /l/. The word-initial /ts/, though words like "tsunami" have become part of English, was predominantly equated with /s/ rather than /ts/. The Japanese liquid, which is usually transcribed as /r/, was predominantly equated with /l/ rather than /r/. This agrees with results of previous research that demonstrate that English /r/ is more dissimilar from Japanese /r/ than English /l/ (See Aoyama et al. 2004).

**4pSCb24. Synchronous speech and speech rate.** Miran Kim (Sunny, Dept. Linguistics, S201, SBS building, Stony Brook, NY 11794-4376, USA, mrkim@ic.sunysb.edu), Hosung Nam (Haskins Laboratories, 300 George St., Suite 900, New Haven, CT 06511, USA, nam@haskins.yale.edu)

Synchronously read speech has shown to reduce a high degree of speaker variability of reading exhibited by speakers in laboratory recording; e.g., pause placement and duration, and speech rate. However, quantitative analysis of speech rate has rarely been found in studies on synchronous speech. This study examines Mandarin Chinese (2 dialects from Taiwan and Shanghai), which is a syllable-time language and thus expected to exhibit a relatively stable speech rate, in both read-alone and read-together speech. Consistency and variability of speech rate are compared in both reading types across repetitions within a subject, across subjects, and across dialects. The results show that speech rate is more stable in read-together than in read-alone speech, and that speech rate in synchronous reading falls on a constant value of speech rate rather than on the average between the speakers' rates in pair. This global pattern is consistent across dialects, and stylized local variation of speech rate over prosodic units (Intonational phrase) is also observed unique to each dialect. We discuss how timing in synchronous speech cannot be accounted for only by dynamic entrainment of speakers with different speech rates and how there should also be rhythmic information shared by speakers of a language.

**4pSCb25. Effects of auditory, visual, and audio-visual training on nonnative perception of English fricatives.** Yue Wang (Simon Fraser Univ., RCB 9224, 8888 Univ. Dr., Burnaby, BC V5A 1S6, Canada, yuew@sfu.ca), Dawn Behne (Norwegian University of Science and Technology, Psychology Dept, NO 7491 Trondheim, Norway, dawn.behne@svt.ntnu.no), Angela Cooper (Simon Fraser Univ., RCB 9224, 8888 Univ. Dr., Burnaby, BC V5A 1S6, Canada, akcooper@sfu.ca), Haisheng Jiang (Simon Fraser Univ., RCB 9224, 8888 Univ. Dr., Burnaby, BC V5A 1S6, Canada, hdjiang@sfu.ca), Nina Leung (Simon Fraser Univ., RCB 9224, 8888 Univ. Dr., Burnaby, BC V5A 1S6, Canada, nhleung@alumni.sfu.ca), Jung-Yueh Tu (Simon Fraser Univ., RCB 9224, 8888 Univ. Dr., Burnaby, BC V5A 1S6, Canada, jta31@sfu.ca)

This study examines the effects of auditory (A), visual (V), and audio-visual (AV) training on nonnative speech perception. Mandarin Chinese natives were trained to perceive English voiceless fricatives (in monosyllabic words and nonwords) of three visually distinct places of articulation: interdentals nonexistent in Mandarin, labiodentals and alveolars common in both languages. Participants were randomly assigned to a control group or one of three 2-week (six sessions, 40 minutes/session) training groups with a dif-

ferent input modality: A, V, or AV. In pre- and post-tests, the fricatives are presented in four ways for an identification task: A-only, V-only, AV congruent (AVc), and AV incongruent (AVi). Additionally, three generalization posttests are administered testing voiced fricatives, new real words, and a new speaker. Results show that post-training, the trainees reveal: (1) improvements corresponding to training type (e.g., the V-training group improves most for the V-only stimuli), (2) greater improvements for the familiar (but less visually distinct) alveolars than for the new interdental, (3) decreased AV-fusion for the AVi stimuli, and (4) consistent patterns in the generalization tests. Results are discussed in terms of the effects of speech input modality, experience, and L1 on L2 AV speech learning. [Research supported by SSHRC]

**4pSCb26. Comparison of the brain regions for consonant processing in Japanese and English subjects.** Yoshikazu Oya (Faculty of Systems Engineering, Wakayama University, 930 Sakaedani, 640-8510 Wakayama, Japan, s085067@sys.wakayama-u.ac.jp), Toshio Irino (Faculty of Systems Engineering, Wakayama University, 930 Sakaedani, 640-8510 Wakayama, Japan, irino@sys.wakayama-u.ac.jp), Alexis G. Hervais-Adelman (Centre for the Neural Basis of Hearing, Department of Physiology, Development and Neuroscience, University of Cambridge, Downing Site, CB23EG Cambridge, UK, alexis.hervais-adelman@mrc-cbu.cam.ac.uk), David T. Ives (Centre for the Neural Basis of Hearing, Department of Physiology, Development and Neuroscience, University of Cambridge, Downing Site, CB23EG Cambridge, UK, dti20@cam.ac.uk), Hideki Kawahara (Faculty of Systems Engineering, Wakayama University, 930 Sakaedani, 640-8510 Wakayama, Japan, kawahara@sys.wakayama-u.ac.jp), 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)

A recent fMRI study on speech-sound processing by Uppenkamp et al. [Neuroimage, 31(3), 1284-1296 (2006)] revealed that regions of the left and right, superior temporal gyri (STG) and anterior, superior temporal sulci (STS) respond preferentially to speech-like stimuli. Hervais-Adelman et al. [BSA London (2007)] extended this research to investigate the processing of consonant-vowel (CV) and vowel-consonant (VC) syllables and determine the locus of consonant-processing in the brain of English speakers. This paper reports an experiment with the same stimuli but with Japanese subjects for whom VC syllables are a novelty. In both the English and Japanese subjects, there was enhanced activity in left STS for vowels over non-speech sounds, as in Uppenkamp et al. (2006). A significant difference was observed between the responses to CV and VC syllables in left STS and planum temporale (PT). There was almost no CV-VC difference in the English subjects. The activity regions for VC syllables were larger than for CV syllables in Japanese subjects, probably because the Japanese subjects would have heard the VCs as two syllable speech sounds. Research supported by JSPS Grant-in-Aid [B18300060] and the UK-MRC [G0500221, G9900369].

**4pSCb27. Influence of music education on second language acquisition.** Barbara Pastuszek-Lipinska (Adam Mickiewicz University, al .Niepodleglosci 4, 61-874 Poznan, Poland, energin@wp.pl)

To explore the extent to which music education influences second-language acquisition, two groups of native Polish speakers, musicians and non-musicians, were asked to reproduce sentences in six languages: English, French, Italian, Spanish, Japanese, and Belgian Dutch. The speech stimuli were developed with a text-to-speech application and differed phonemically, phonostylistically, and in length. The paper includes results of a general auditory analysis of subjects' productions as well as the results of a web-based listening test with a panel of native speakers of the involved languages. All collected data were also analyzed with statistical tools. The results revealed that music education exerted a measurable impact on speech perception and production. Musicians outperformed non-musicians in the study. From the results, it appears that the influence of musical expertise extends beyond music processing to speech processing, and the strength of this influence is connected not only to auditory training. Therefore, the superior performance of the musicians in the task may be interpreted as evidence that music edu-

cation is an enabling factor in the successful acquisition of a second language. It also indicates that the impact of musical training is not a myth, but has a scientific basis.

**4pSCb28. The relation between articulatory and aerodynamic properties of single /t/, /S/ and their combination in /tS/ and /St/.** Susanne Fuchs (ZAS, Schuetzenstr. 18, 10117 Berlin, Germany, fuchs@zas.gwz-berlin.de), Laura L. Koenig (Haskins Labs. and Long Island Univ., Brooklyn, New York, NY 11201-8423, USA, koenig@haskins.yale.edu)

Intraoral pressure changes during speech production are a result of different factors, such as articulatory movements (place and manner of articulation), subglottal pressure, and laryngeal-oral coordination. This study attempts to provide a better understanding of the relation between aerodynamics and articulation in single voiceless consonants and their combinations. We gathered electropalatographic data simultaneously with intraoral pressure data for 9 native speakers of German. During voiceless aspirated stop production intraoral pressure equalizes with subglottal pressure since an oral closure is formed and the glottis is open. During voiceless fricative production intraoral pressure is lower than in stop production since the air can escape through the oral constriction. In /t, S, St/ the intraoral pressure changes are highly correlated with the articulatory behaviour. However, the affricate /tS/ shows an unusual relationship with aerodynamics and articulation. Although an oral closure is formed, pressure rises at a later point and reaches its peak during the fricative. We interpret these findings with respect to a special laryngeal - oral coordination in affricates as reported in Hoole et al. (2003) who found a relatively late glottal opening during the stop portion of the affricate.

**4pSCb29. Experimental study of the fluid-structure-acoustic interaction in a human voice model.** Stefan Kniesburges (University Erlangen-Nuremberg, Institute of Fluid Mechanics, Cauerst. 4, 91058 Erlangen, Germany, stefan.kniesburges@lstm.uni-erlangen.de), Stefan Becker (University Erlangen-Nuremberg, Institute of Fluid Mechanics, Cauerst. 4, 91058 Erlangen, Germany, stefan.becker@lstm.uni-erlangen.de), Stefan Mueller (University Erlangen-Nuremberg, Institute of Fluid Mechanics, Cauerst. 4, 91058 Erlangen, Germany, stefan.mueller@lstm.uni-erlangen.de), Antonio Delgado (University Erlangen-Nuremberg, Institute of Fluid Mechanics, Cauerst. 4, 91058 Erlangen, Germany, adelgado@lstm.uni-erlangen.de), Gerhard Link (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, gerhard.link@lse.eei.uni-erlangen.de), Manfred Kaltenbacher (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, manfred.kaltenbacher@lse.eei.uni-erlangen.de), Michael Doellinger (University Hospital Erlangen, Department of Phoniatrics and Pediatric Audiology, Medical School, Bohlenplatz 21, 91054 Erlangen, Germany, michael.doellinger@uk-erlangen.de)

A fluid-structure-coupled in-vitro model was developed for the investigation of the physical processes of the human phonation. The setup enables to simulate self-sustained vocal fold oscillation and to analyse the resulting supraglottal flow field. Measurement techniques like high-speed flow visualization, Particle Image Velocimetry (PIV) of the time-dependent flow field, unsteady pressure measurement, vibration measurement by a Laser-Scanning-Vibrometer were applied. The acoustic field was simultaneously recorded. Analysis was performed regarding correlations between the acoustic field, the flow velocity and the displacement of the vocal folds. The results support the existence of the Coanda-effect during phonation as assumed in previous work. The flow attaches to one vocal fold just past the glottis and forms a spacious vortex behind the vocal folds. This behaviour is not linked to one vocal fold and changes stochastically from cycle to cycle. The analysis indicates, that the acoustic sound is primarily produced by the pressure fluctuation caused by the pulsating flow rate due to the opening and closing process of the vocal folds. We assume, that the structural sound and the turbulence-induced sound do have minor acoustical implication.

**4pSCb30. FEM simulation of tongue deformation for /i/ with a four-cube model applied to tagged cine-MRI data.** Sayoko Takano (ATR Cognitive Information Science Laboratories, 2-2-2 Hikaridai, Seika-cho Soraku-gun, 619-0288 Kyoto, Japan, tsayoko@atr.jp), Hiroki Matsuzaki (Hokkai-Gakuen University, 1-1, Minami-26, Nishi-11, Chuou-ku, 064-0926 Sapporo, Japan, matsu@eli.hokkai-s-u.ac.jp), Kunitoshi Motoki (Hokkai-Gakuen University, 1-1, Minami-26, Nishi-11, Chuou-ku, 064-0926 Sapporo, Japan, motoki@eli.hokkai-s-u.ac.jp)

Roles of extrinsic and intrinsic tongue muscles in the production of vowel /i/ were examined using a finite element method (FEM) applied to the tagged cine-MRI data. It has been thought that tongue tissue deformation for /i/ is mainly due to the combined actions of the genioglossus muscle bundles advancing the tongue root to elevate the dorsum with a mid-line grooving. A recent study with the tagging-MRI revealed earlier, faster and greater tissue deformation at anterior top of the tongue than posterior part during /ei/ production. This result implies the contribution of the intrinsic tongue muscle (transverse anterior) with an independent hydrostat mechanism from that of the genioglossus muscle bundles. In this study, a simple four-cube model is built to examine the co-contraction effect of the genioglossus and transverse muscles using the FEM. The simulation result with the anterior transverse muscle (Ta) showed good agreement with the pattern of the tongue deformation obtained from the tagged-MRI data, suggesting that transverse anterior also plays an important role for the realization of the tongue shape for the production of vowel /i/.

**4pSCb31. Classification of normal and pathological vocal fold vibrations using Phonovibrograms.** Jörg Lohscheller (University Hospital Erlangen, Department of Phoniatrics and Pediatric Audiology, Medical School, Bohlenplatz 21, 91054 Erlangen, Germany, joerg.lohscheller@uk-erlangen.de), Daniel Voigt (Dept. Phoniatrics and Pediatric Audiology, Bohlenplatz 21, 91054 Erlangen, Germany, daniel.voigt@uk-erlangen.de), Michael Doellinger (University Hospital Erlangen, Department of Phoniatrics and Pediatric Audiology, Medical School, Bohlenplatz 21, 91054 Erlangen, Germany, michael.doellinger@uk-erlangen.de)

Clinical examination of voice disorders demands an endoscopic observation of vocal fold vibrations. High-speed endoscopy is the state-of-the-art technology for investigation of vocal fold vibrations. A novel visualization strategy is proposed which transforms the segmented contours of vocal fold edges into a set of two dimensional images, denoted Phonovibrograms (PVG). Within PVGs the individual type of vocal fold vibration becomes uniquely characterized by specific geometric patterns which can be seen as fingerprints of vocal fold vibration. The PVGs give an intuitive access on the type and degree of the laryngeal asymmetry which is essential to quantify the effects of functional and organic voice disorders. To determine the vibration characteristics within the computed PVG pattern recognition algorithms are applied. Thus, for each vocal fold the vibration type can be quantified and classified. The results of the PVG classification will be presented in 80 subjects (normal and pathological voices). It will be shown, that a classification of the vibration type can be performed very precisely even in disturbed vocal fold vibrations. The obtained PVG images can be documented and stored on a hard-disc using a lossless image data-format. The quantitative description of PVG patterns has the potential to realize a novel classification of vocal fold vibrations.

**4pSCb32. Glottal-opening and airflow pattern during production of voiceless fricatives: a new non-invasive instrumentation.** Kiyoshi Honda (Phonetics & Phonology Laboratories, UMR-7018-CNRS & University of Paris III, 19, rue des Bernardins, 75005 Paris, France, honda@atr.jp), Shinji Maeda (CNRS & ENST, 46, rue Barrault, 75634 Paris, France, maeda@tsi.enst.fr)

In the production of voiceless fricatives, the airflow passing through the vocal tract is controlled by reciprocal open-close patterns of the glottal and oral constriction. In order to observe such coordinated patterns, the authors have developed a combined method using a non-invasive (external-lighting and sensing) photoglottographic (ePGG) technique and a pressure-difference airflow mask. The former technique has the advantage of no restriction of phonetic environments (improvement from the standard PGG) and can be

further combined with instrumentation of intraoral air pressure and/or of articulatory movements. Our PGG-airflow data are examined to address a question: It has been known that the glottis opens always wider for word-initial fricatives than for word-medial ones despite no obvious difference in acoustic and physiological requirements. It will be discussed whether the degrees of glottal opening during a fricative co-vary with other aerodynamic and articulatory controls in CVCV utterances with V = nonclose vowels.

**4pSCb33. Mucosal wave analysis of a human vocal fold in the hemilarynx experiment.** Michael Doellinger (University Hospital Erlangen, Department of Phoniatrics and Pediatric Audiology, Medical School, Bohlenplatz 21, 91054 Erlangen, Germany, michael.doellinger@uk-erlangen.de), David A. Berry (The Laryngeal Dynamics Laboratory, Division of Head & Neck Surgery, David Geffen School of Medicine at UCLA, 31-24 Rehab Center, 1000 Veteran Ave., Los Angeles, 90095-1794, USA, daberry@ucla.edu), Jörg Lohscheller (University Hospital Erlangen, Department of Phoniatrics and Pediatric Audiology, Medical School, Bohlenplatz 21, 91054 Erlangen, Germany, joerg.lohscheller@uk-erlangen.de)

The mucosal wave propagation was investigated in laboratory experiments across a variety of phonatory conditions. The focus was on the medial and superior surface dynamics of the vocal fold, which quantify mucosal wave propagation, but have been relatively little studied. High-speed, digital imaging of the entire surface of the vocal fold was performed using an excised human hemilarynx setup. Surface dynamics were characterized and differentiated across a variety of phonatory conditions. During sustained, flow-induced oscillation, the local maxima of vocal fold mucosal displacements, velocities and acceleration and their particular phase delays in the glottal cycle were investigated. Statistical analysis was performed, examining the influence of applied stimulations. Increasing the airflow yielded higher values for lateral displacements as well as higher velocity/acceleration values. Elongating the vocal fold resulted in decreased lateral displacements. The mucosal wave propagation apparently increased for higher flow, elongated folds, and higher adduction forces. While an understanding of the correlation between vocal fold dynamics and phonatory physiology/pathology is still in its infancy, the data presented here help to establish such connections. The data are also useful for the development and evaluation of physical and numerical models of vocal fold vibration.

**4pSCb34. Categorical and non-categorical perception of speech: Behavioural and neural evidence.** Jack C. Rogers (MRC Cognition and Brain Sciences Unit, 15 Chaucer Road, CB2 7EF Cambridge, UK, jack.rogers@mrc-cbu.cam.ac.uk), Matthew H. Davis (MRC CBU, 15 Chaucer Rd., CB2 7EF Cambridge, UK, matt.davis@mrc-cbu.cam.ac.uk)

What effects do within- and between-category variations have on the perception of speech? Using audio-morphing and the "Straight" channel vocoder (Kawahara, 2004), we produced 320 high-quality phonetic continua varying in place, manner and voicing including word/word (blade/glade), word/pseudo (blouse/glouse), pseudo/word (bown/gown) and pseudo/pseudo (blem/glem) pairs. A 2AFC task confirmed the category boundary shift for word/pseudo and pseudo/word pairs (Ganong, 1980), equivalent for onset (bench/gench) and offset (flad/flag) pairs. This suggests that lexical influences on categorical perception are not produced on-line but rather occur post-perceptually, consistent with top-down effects. Sensitivity to within- and between-category phonological variation was investigated using sparse fMRI in a paired auditory repetition priming paradigm. Minimal pairs (48 across the 4 stimulus groups) were presented to participants who listened in the context of a semantic monitoring task. Between-category pairs with a phonological change produced a greater neural response compared to within-category same pairs with the same magnitude of acoustic difference. This response to phonologically different pairs provides a neural correlate of categorical perception in left middle temporal, inferior frontal and pre-central regions. These responses in inferior frontal regions may contribute towards top-down influences on categorical perception of speech (cf Ganong Effect).



**4pSCb35. Distinguishing Place of Consonant Articulation using the Aurora System.** Yana Yunusova (University of Toronto, 160-500 University Ave, Toronto, ON M5G 1V7, Canada, yana.yunusova@utoronto.ca), Jeff Stanley (Northern Digital Inc, 103 Randall Drive, Waterloo, ON N2V 1C5, Canada, jstanley@ndigital.com), Jordan R. Green (University of Nebraska, 3180 Barkley Center, Lincoln, NE 68583-0738, USA, jgreen4@unl.edu)

The goal of this project is to determine the effect of bite-block and speaking rate manipulation on lingual targets for consonants using a recently developed Aurora, a 3-D electromagnetic movement tracking system (NDI). Bite-block and speaking rate manipulations are common intervention techniques with speakers with dysarthria and apraxia of speech. Therefore, understanding their effect on consonant target regions is essential for predicting the outcomes of articulatory training in these clinical populations. In this project, articulatory positions of two sensors on the tongue tip and dorsum will be recorded independent of the head. Stops /t, k/ and fricatives /s, sh/ will be embedded in aCa syllables. A bite block condition will be used to eliminate jaw contribution to tongue movements. The location and size of articulatory regions associated with each consonant will be compared in bite block and no-block conditions. Additionally, the rate effect on the location and size of the consonant articulatory regions will be examined. Potential clinical implication of the finding on articulatory intervention will be considered.

**4pSCb36. Discrimination of Mandarin tone 1 vs. tone 4 in disyllables by adult speakers of English.** Shari Berkowitz (CUNY Graduate Center, Speech Acoustics and Perception Lab, 365 Fifth Avenue, New York, NY 10016, USA, shariellen@nyc.rr.com)

Previous research on cross-language perception of lexical tone has mainly used monosyllabic stimuli; however disyllables may be more difficult for non-natives to discriminate due to coarticulation and context effects (Berkowitz & Strange, 2007). Preliminary work suggests that there is an effect of context on discrimination of tone 1 vs. 4, despite the fact that this is usually considered an easy contrast when tested with monosyllabic stimuli. Disyllabic Mandarin nonsense words served as stimuli in a categorial same/different task. Pairings of tone 1 and tone 4 in initial and final position were tested in the context of all four tones. American English listeners with no background in tone languages completed the experiment without feedback. Data was scored with A' and was analyzed for effect of tone, initial vs. final position in the disyllable, and height and contour of the fundamental frequency. The results of this paradigm will be used to design a study of tone perception in preschool groups who speak Mandarin as their L1, English as their L1, and internationally adopted children who were previously exposed to Mandarin.

**4pSCb37. Articulatory features influencing regressive place assimilation across word-boundaries in German.** Marion Jaeger (Institut für Phonetik und Sprachverarbeitung, Schellingstr. 3, 80799 München, Germany, jaeger@phonetik.uni-muenchen.de), Phil Hoole (Institut für Phonetik und Sprachverarbeitung, Schellingstr. 3, 80799 München, Germany, hoole@phonetik.uni-muenchen.de)

Within current phonological theories the greater tendency of C1 nasals vs. C1 plosives to undergo regressive place assimilation is often treated as the consequence of acoustic-perceptual properties of nasality (e.g. Steriade, 2001). Little is known about the articulatory patterns underlying this asymmetry. Our current EMA study aims to test and compare the effects of manner of articulation of C1 (alveolar nasal vs. alveolar plosive), place of articulation of C2 (labial vs. dorsal plosive), vowel context (palatal /i/ vs. non-palatal vowel /a/), and word frequency upon the intra- and intergestural timing and movement magnitude of various articulators in C1C2 sequences across word-boundaries in German subjects. Our analyses of non-palatal vowel contexts in three speakers showed a greater likelihood of reduction of the tongue tip both in words with a nasal C1 and in high frequency words. For those word pairs in which tongue tip displacement was measurable, tongue tip - tongue back overlap was significantly greater in word pairs with

a nasal C1 and in word pairs with high frequency words. On the other hand, tongue tip - lower lip overlap was only significantly greater in word pairs with high frequency words.

**4pSCb38. Time-frequency detection of stridence in fricatives and affricates.** Slobodan Jovicic (School of Electrical Engineering, University of Belgrade, Bulevar kralja Aleksandra 73, 11000 Belgrade, Serby, jovicic@etf.bg.ac.yu), Silvana Punisic (Institute for Experimental Phonetics and Speech Pathology, Gospodar Jovanova 35, 11000 Belgrade, Serby, iefpgmir@eunet.yu), Zoran Saric (Institute for Experimental Phonetics and Speech Pathology, Gospodar Jovanova 35, 11000 Belgrade, Serby, sariczoran@yahoo.com)

As wheezes in abnormal breath sounds observed in patients with obstructive pulmonary diseases, the stridence in voice is manifested as excessively sharp, conspicuous, usually habitual hiss that is especially distinct with whispering. This paper reviews the articulator and acoustics features of stridence in unvoiced fricatives and affricates, and presents an algorithm for detection of stridence. Detection of stridence was based on: time-frequency representation by FFT power spectra, time-frequency representation by AR-Burg power spectra, and power trajectory of signals in characteristic frequency bands. Many features are extracted from this analysis, as: the local power spectra maximum to average surrounding power ratio, the correlation coefficient between spectral and power maxima in signal, the spectral power slope in selected frequency band, spectral entropy in selected frequency band, and phoneme duration. The extracted set of features is input to the nonlinear classifier that decides about stridence in voice and the level of pathology. The algorithm was tested with speech database of normal and pathology voices. The speakers were both sex and the different ages. The results of automatic stridence detection showed high level of coincidence with the judgment of speech therapists.

**4pSCb39. Adaptive microphone array free of the desired speaker cancellation combined with postfilter.** Slobodan Jovicic (School of Electrical Engineering, University of Belgrade, Bulevar kralja Aleksandra 73, 11000 Belgrade, Serby, jovicic@etf.bg.ac.yu), Zoran Saric (Institute for Experimental Phonetics and Speech Pathology, Gospodar Jovanova 35, 11000 Belgrade, Serby, sariczoran@yahoo.com)

The optimal microphone array includes two processing blocks - minimum variance distortionless response (MVDR) beamformer and the single-channel Wiener filter, which acts as post-filter. The main drawback of MVDR beamformer is the cancellation of the desired speech signal and its degradation in multi-path wave propagation environment. To make the adaptive algorithm robust against room reverberation and to prevent desired signal cancellation, an estimation of the unknown desired speaker's transfer function was proposed. The estimation is based on the imperfect signal and the interference covariance matrices estimated from available microphone signals during speaker activity and pause of speech respectively. As MVDR beamformer suppresses coherent interference, post-filter has to reduce diffuse acoustic noise. The post-filter proposed in this paper is developed under assumption that complex coherence function is unknown but time invariant. The additional improvement of the post-processing algorithm on low frequencies is obtained by combining a priori noise power attenuation factor for diffuse noise field with estimated one. The proposed algorithm is tested on simulated room with reverberation, and compared with some known post-processing algorithms with rather good results.

**4pSCb40. The role of source and filter cues in emotion recognition in speech.** Disa Sauter (Birkbeck College London, Henry Wellcome Building, Malet Street, WC1E 7HX London, UK, d.sauter@bbk.ac.uk), Frank Eisner (University College London, Institute of Cognitive Neuroscience, 17 Queen Square, WC1N 3AR London, UK, f.eisner@ucl.ac.uk), Stuart Rosen (Department of Phonetics and Linguistics, University College London, Gower Street, WC1E 6BT London, UK, s.rosen@ucl.ac.uk), Sophie K. Scott (University College London, Institute of Cognitive Neuroscience, 17 Queen Square, WC1N 3AR London, UK, sophie.scott@ucl.ac.uk)

In the context of the source-filter theory of speech, it is well established that intelligibility is heavily reliant on information carried by the filter, that is, spectral cues (e.g., Faulkner et al., 2001; Shannon et al., 1995). However,



the extraction of other types of information in the speech signal, such as emotion and identity, is less well understood. In this study we investigated the extent to which emotion recognition in speech depends on filter-dependent cues, using a forced-choice emotion identification task at ten levels of noise-vocoding ranging between one and 32 channels. In addition, participants performed a speech intelligibility task with the same stimuli. Our results indicate that compared to speech intelligibility, emotion recognition relies less on spectral information and more on cues typically signaled by source variations, such as voice pitch, voice quality, and intensity. We suggest that, while the reliance on spectral dynamics is likely a unique aspect of human speech, greater phylogenetic continuity across species may be found in the communication of affect in vocalizations.

**4pSCb41. Model of the configurations of intonation movements in Danish - suggesting an 'intoneme'.** Sophia Frovin (SLK, English Department, University of Aarhus, Jens Chr. Skous Vej 5, 8000 Aarhus C, Denmark, sophiafrovin@yahoo.com)

This study examines the inventory of intonation movements in Danish. The aim is to identify the micro intonation and to produce a model of the movement patterns in Danish, which would predict the intonation of synthetic speech in order to establish a close-to-natural intonation. The theoretical background is a modified version of the "close copy theory" (t Hart et al.), in which movements are tendency lines in the intonation pattern. A new movement starts when the tendency line shifts direction. The acoustic analysis comprises measurements of time, semitones and slope (semitones/second = slope) and a description of the (parts of) phones in intonation movements of actual speech. The analysis suggests that the slope of the movement is one of the most important features of intonation, and that speakers are aiming to produce one of a set of 5 standard slopes, depending mainly on the phonetic segment, suggesting an underlying unit - an "intoneme"- displaying the same relationship as that of phoneme and phone.

**4pSCb42. Influences of manner and voicing on articulatory coordination in German and French initial consonant clusters.** Barbara Kuehnert (Laboratoire de Phonétique et Phonologie - UMR 7018 CNRS - Paris 3, 19, rue des Bernadins, 75005 Paris, France, barbara.kuehnert@univ-paris3.fr), Phil Hoole (Institut für Phonetik und Sprachverarbeitung, Schellingstr. 3, 80799 München, Germany, hoole@phonetik.uni-muenchen.de), Christine Mooshammer (Haskins lab and MIT Research lab of electronics, 300 George street suite 900, New Haven, CT 06511, USA, tine@haskins.yale.edu), Lasse Bombien (Institut für Phonetik und Sprachverarbeitung, Schellingstr. 3, 80799 München, Germany, lasse@phonetik.uni-muenchen.de)

This study aims for improved understanding of whether and how coordination patterns of supraglottal gestures in complex syllable onsets are driven by competing demands of motor economy for the speaker and high recoverability for the listener. Specifically, EMA data for four German and three French speakers was acquired for C1C2 clusters where manner of articulation was varied for C2 (/l/ vs. /n/) and voicing for C1 (/p/ vs. /b/, /k/ vs. /g/). Results (1): A robust effect of less overlap of the constriction gesture for C1 and C2 when C2 is nasal. Clusters with nasal C2 may require less overlap in order to avoid compromising the acoustic characteristics of the C1 burst by early velar lowering. Interestingly, such clusters appear to be less stable diachronically and may be less favoured for the formation of complex onsets because of reduced scope for efficient parallel transmission of segmental information. Results (2): For German, a consistent effect of less overlap for voiceless compared to voiced C1. Discussion here will centre on whether possible cross-language differences between German and French can be related to differences in timing of voice onset and resulting differences in the acoustic properties of the C1-C2 transitions.

**4pSCb43. Characterisation of the velocity distribution on rigid in-vitro upper airway replicas.** Annemie Van Hirtum (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, annemie.vanhirtum@gipsa-lab.inpg.fr), Xavier Grandchamp (Département Parole & Cognition, GIPSA-lab, 46, avenue Félix Viallet, 38031 Grenoble Cedex, France, xavier.grandchamp

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Qualitative and quantitative characterisation of the velocity distribution is an important aspect in order to study respiration related flow phenomena in the upper airways as well as for human speech production. Classical phonation models exploit one-dimensional flow descriptions in order to estimate the pressure forces exerted on the vocal folds tissues. In this case, the velocity distribution is expressed as an analytical relationship depending on volume velocity airflow, geometry and pressure distribution. Therefore simplified flow models can be validated by measuring the relevant quantities. However, more detailed and quantitative velocity predictions, aiming e.g. to improve phonation modelling or to study turbulent sound production, require increased precision for both the qualitative and quantitative characterisation of the velocity distribution. The current paper presents preliminary in-vitro measurements of the velocity distribution obtained by Particle Image Velocimetry combined with hot film anemometry. Several simplified rigid geometries are assessed in order to represent different portions of the upper airway. Besides qualitative results a first quantitative comparison between simulated and measured velocities is provided and discussed.

**4pSCb44. Experimental analysis of the relationship between the glottal flow and glottal area waveforms.** Raphael Schwarz (University of South Carolina, 1621 Greene St, Williams Brice Building, 6th Fl, Columbia, SC 29208, USA, rschwarz@gwm.sc.edu), Dimitar D. Deliyski (University of South Carolina, 1621 Greene St, Williams Brice Building, 6th Fl, Columbia, SC 29208, USA, deliyski@gwm.sc.edu)

A very important but little studied aspect of human voice production is the relationship between the vocal fold vibration and the transglottal airflow. To analyze this relationship, in this study we combined high-speed videendoscopy of the glottis for determining the glottal area waveform (GAW) with inverse filtering of the acoustic signal for estimating the glottal flow waveform (GFW). The high-speed camera system, recording at 20,000 pps, and the audio recording hardware were triggered by the same quartz oscillator to achieve synchronization with an unprecedented accuracy within 25  $\mu$ s. We developed an image processing algorithm for automatic extraction of the GAW from the high-speed images. The high-speed video samples and the corresponding acoustic signals were obtained from 12 normophonic individuals (6 male, 6 female) for different voicing conditions: register (pulse, modal, falsetto); adductory adjustment (loose, normal, pressed); longitudinal tension within modal register (low, comfortable, high pitch); non-stationary phonation (variation in pitch and loudness). To compare the resulting GAWs and GFWs, the waveforms were parameterized concerning their temporal and spectral features. It is shown, that the revealed relationships between the vocal fold vibrations and the transglottal flow are comprehensible by accounting the different phonation conditions. [Work supported by NIH.]

**4pSCb45. Pitch production by bilingual Mandarin-Min speakers and monolingual Mandarin speakers in speaking Taiwan Mandarin and Min.** E-Chin Wu (Graduate Institute of Linguistics, National Taiwan Univ, Grad. Inst. of Linguistics, National Taiwan Univ., 1, Roosevelt Rd. Sec. 4, 106 Taipei, Taiwan, r95142002@ntu.edu.tw), Janice Fon (Grad. Inst. of Linguistics, National Taiwan Univ., 1, Roosevelt Rd. Sec. 4, 106 Taipei, Taiwan, jfon@ntu.edu.tw)

In this study, Mandarin produced by bilingual Mandarin-Min speakers and monolingual Mandarin speakers were compared to see whether the presence of Min would pose influence on the pitch range utilized by bilingual Mandarin-Min speakers in speaking Mandarin. The results showed that bilingual Mandarin-Min speakers and monolingual Mandarin speakers did indeed differ in their pitch values when speaking Mandarin. Intriguingly, however, the difference did not go in the same direction for males and females.

It is proposed here that the difference found in the male group was due to influence of Min, while the difference found in the female groups was due to social factors. The results of this study indicate that both language experiences and social constraints are important factors that influence a person's use of pitch.

**4pSCb46. Probing the independence of formant control.** Ewen Macdonald (Queen's University, 62 Arch St, Dept. of Psychology, Humphrey Hall, Kingston, ON K7L 3N6, Canada, ewen.macdonald@queensu.ca), Bryan P. Burt (Queen's University, 62 Arch St, Dept. of Psychology, Humphrey Hall, Kingston, ON K7L 3N6, Canada, bryan.burt@queensu.ca), Kevin G. Munhall (Queen's University, 62 Arch St, Dept. of Psychology, Humphrey Hall, Kingston, ON K7L 3N6, Canada, kevin.munhall@queensu.ca)

Previous experiments in speech motor learning have demonstrated that acoustic feedback is used to control formant frequencies (Houde and Jordan 1998; Purcell and Munhall 2006; Villacorta, Perkell, and Guenther 2007). In these studies, the formants of a vowel were shifted using a real-time signal processing system. When subjects spoke a vowel, they heard themselves saying a different vowel. The talkers spontaneously compensated for this auditory feedback perturbation by producing formants in the opposite direction in frequency to the perturbation. The purpose of the present study was to investigate if compensating for a perturbation in either the first or second formant also leads to a change in production of the other formant. A between-subjects experiment was conducted where half the participants had a perturbation applied only to the first formant and the other half had a perturbation applied only to the second formant. As in previous experiments, both groups compensated for the perturbed formant. In compensating for the perturbed formant, the production of the other, unperturbed, formant was also affected. However, the change in frequency was quite small. The results will be discussed in terms of the independent variables of speech motor planning

**4pSCb47. Pitch Tracking using the Generalized Harmonicity Indicator.** Darren Haddad (Air Force Research Lab, 525 Brooks Road, Rome, NY 13441, USA, darren.haddad@rl.af.mil), Andrew Noga (525 Brooks Road, Rome, NY 13441, USA, Andrew.Noga@rl.af.mil), Tappan Sarkar (323 Link Hall, Syracuse University, Syracuse, NY 13244-1240, USA, tksarkar@syr.edu)

For many audio applications, a process is required to obtain an accurate estimate of the fundamental and harmonics of periodic sections of the audio signal. The Generalized Harmonicity Indicator (GHI) is to determine, assess and track the fundamental and harmonic frequencies of consecutive time segments of a speech signal using the Matrix Pencil (MP) technique[i]. Various methods of fundamental and harmonic frequency tracking have been proposed and developed, but most have been based on other low resolution techniques such as FFT and cepstral analyses. This is opposed to using a super-resolution frequency estimation technique as provided by the MP. The prior art in the area of super-resolution speech fundamental determination consists of the "super resolution pitch determinator" (SRPD)[ii] and the "enhanced SRPD" (eSRPD) methods. Because these prior methods do not explicitly process a spectral representation or decomposition of the input audio signal, they are not considered to be in the same class as the MP GHI, although they provide a baseline for comparing different estimation techniques. [i] Haddad, D.M., Sarkar, T.K., Noga, A.J.; "Speech Compression Using the Matrix Pencil Technique"; IEEE 12th DSP Workshop; Sept. 2006; Page(s):218-221. [ii] Y. Medan, E., Yair, D., Chazan, "Super Resolution Pitch Determination of Speech Signals," IEEE Trans. On Signal Processing, ASSP-39(1):40-48, 1991.

**4pSCb48. Intraglottal pressures in a static physical model of the converging glottis: entrance loss coefficients, exit coefficients, Bernoulli effects, and viscous effects.** Lewis P. Fulcher (Bowling Green State University, Department of Physics and Astronomy, Bowling Green, OH 43403, USA, fulcher@bgsu.edu), Ronald C. Scherer (Bowling Green State University, Department of Physics and Astronomy, Bowling Green, OH 43403, USA, ronalds@bgsu.edu)

Pressure distributions were obtained with a static physical model (M5) at diameters  $d = 0.005, 0.01, 0.02, 0.04, 0.08, 0.16$ , and  $0.32$  cm for converging angles of  $5, 10, 20$ , and  $40$  degrees. At each diameter and angle, transglottal pressures typically ranged from  $3$  to  $15$  cm  $H_2O$ . For each angle, diameter, and transglottal pressure, the measured pressure at the glottal entrance was used to calculate an entrance loss coefficient, and the measured pressure near the glottal exit was used to determine an exit coefficient. Previous work with the uniform glottis, where the only important physical effect was viscosity, found linear fits to the intraglottal pressures to be excellent approximations. Since the widening channel of the converging glottis produces Bernoulli effects as well as viscous effects, a parabolic form for intraglottal pressures is explored. Such an analytic form for the intraglottal pressures when accompanied by tables of entrance loss and exit coefficients would be a useful tool for researchers needing expressions easily included in numerical models of phonation. The validity of the new analytic treatment will be assessed by comparing the calculated pressures with the observed M5 pressures. [Work supported by NIH R01DC03577.]

**4pSCb49. Mitigation of Nonlinear Distortion in Speech Signals Using Histogram Matching.** Brett Smolenski (RADC, 2433 Forest Lane, Schwenksville, PA 19473, USA, bsmolens@gmail.com)

Nonlinear distortion is a common artifact in audio communication equipment. In addition, it is well known that the normalized amplitude distribution of speech signals converges to approximately a gamma distribution after a few seconds. Hence, the transfer function of any memoryless nonlinearity distorting the speech signal can easily be estimated, provided one has buffered enough data. This research shows how both parametric and non-parametric histogram matching algorithms can be employed to remove the effects of these types of distortions. Further, the improvement these algorithms have on speaker identification performance is also studied. This approach represents a radical departure from the traditional approach taken in speech enhancement. The traditional approach has been to first acquire a model or representation of the distortion, noise, or interference that is corrupting the signal, and then attempt to remove this from the signal, usually introducing other forms of distortion in the process. With this approach a model of clean undistorted speech is used that the distorted speech is then matched to.

**4pSCb50. Fricative synthesis investigations using the transmission line matrix method.** Athanasios Katsamanis (National Technical University of Athens, School of Electrical and Computer Engineering, Zografou campus, 15773 Athens, Greece, nkatsam@cs.ntua.gr), Petros Maragos (National Technical University of Athens, School of Electrical and Computer Engineering, Zografou campus, 15773 Athens, Greece, maragos@cs.ntua.gr)

We investigate the potential of properly applying the Transmission Line Matrix method to simulate the 3D acoustic field in the vocal tract especially for the synthesis of fricatives. For fricatives, we are mainly interested in the higher end of the spectrum where the planar wave propagation assumption that is accepted in the one-dimensional simulation cannot provide accurate results. This is the main reason why we explore 3D acoustic field simulation. Proper incorporation of noise sources is considered to account for frication. Their placement and acoustic properties are investigated. Motivated by measurements of the airflow during fricative production, we also explore the influence of mean flow to acoustics in the tract. Sound propagation in moving medium is considered for this purpose. The applied vocal tract geometry is determined from 3D MRI images. We present computational considerations for the analyzed framework in parallel to potential benefits compared to the one-dimensional vocal tract simulation.

**4pSCb51. Acquisition of the production of 'new' and 'similar' vowels: the case of /u/ and /y/ in French by Japanese-speaking learners.** Takeki Kamiyama (Laboratoire de phonétique et phonologie (UMR 7018), 19, rue des Bernardins, 75005 Paris, France, takekik@phiz.c.u-tokyo.ac.jp)

French /y/ (F2/F3 close around 2000 Hz for males) does not have an equivalent phoneme in Japanese and English, whereas /u/ (F1/F2 close < 1000 Hz) has a phonemic counterpart (high back) in both Japanese and English, but its phonetic realization is different from French /u/, with a higher F2. Flege (1987) found out that it was easier for American English

speakers to acquire a native-like production of French /y/ ("new" phone) than that of /u/ ("similar" phone). Our recording of two groups of adult Japanese-speaking learners of French (JLF) shows a similar tendency. We measured the formants of isolated vowels embedded in a carrier sentence read 3 times by JLFs. In the first group, consisting of 3 elementary learners who volunteered for the task, 2 learners pronounced /y/ with close F2/F3, but none of them produced a low F2 (< 1000 Hz) for /u/. In the second group, consisting of 50 JLFs in 4 different classes and levels (elementary to upper intermediate), only 4 of them, who had all lived in a French-speaking country except one, produced /u/ with a low F2 (< 1000 Hz), whereas a dozen of them pronounced /y/ with close F2/F3.

**4pSCb52. On the influence of vocal fold collision on phonation.** Juergen Neubauer (School of Medicine, University of California, 31-24 Rehab Center, 1000 Veteran Ave., Los Angeles, CA 90095-1794, USA, jneubauer@mednet.ucla.edu), Zhaoyan Zhang (School of Medicine, University of California, 31-24 Rehab Center, 1000 Veteran Ave., Los Angeles, CA 90095-1794, USA, zyzhang@ucla.edu)

For laryngoscopic observations show that the closed time of the glottis is significantly long, especially for low-frequency vocal fold vibrations in the vocal fry and chest register. Vocal fold contact appears to be a major part of the phonatory cycle and may play an important role for self-sustained vocal fold oscillations. Using a 2D, finite element, self-oscillating model of the coupled vocal fold-glottal flow system, we studied the influence of the mechanical impact on phonation onset mechanisms and vocal fold vibratory behavior. The air flow was assumed to be laminar and the compressible Navier-Stokes equations were solved for the flow domain. For fixed values of the Young's modulus of the vocal fold we found that vocal fold contact significantly increased the vibration frequency as compared to the case of no contact. The changed total pressure forces on the vocal fold in medial-lateral (lift force) and inferior-superior (drag force) directions resulted in different phonation threshold pressure values and vocal fold vibration patterns. The increase in phonation frequency due to contact will be discussed based on the theory of impact oscillators.

**4pSCb53. ERPs to words correlate with behavioral measures in children with Autism Spectrum Disorder.** Sharon Coffey-Corina (Center for Mind and Brain UC Davis, 267 Cousteau Pl, Davis, CA 95618, USA, sccorina@ucdavis.edu), Denise Padden (I-LABS, University of Washington, Box 357920, Seattle, WA 98195, USA, dpadden@u.washington.edu), Patricia K. Kuhl (University of Washington, Dept. of Speech & Hearing Sciences, and Institute for Learning & Brain Sciences, Box 357988, Seattle, WA 98195, USA, pkkuhl@u.washington.edu)

Children with Autism Spectrum Disorder (ASD) participated in a research study that involved both electrophysiological and behavioral measures. Event related brain potentials (ERPs) were recorded during auditory presentation of known and unknown words. Behavioral measures of language/cognitive function and severity of autism symptoms were also collected at the time of ERP testing and again one year later. In general, higher functioning children with ASD exhibited more localized brain effects for differences between known and unknown words. Lower functioning children with ASD had more diffuse patterns of response to the different word classes and also exhibited a stronger right hemisphere lateralization. That is, they showed differences between known and unknown words at many electrode sites and larger differences in the right hemisphere. In addition, significant correlations were obtained between specific brain wave measurements for both known and unknown words and the various behavioral measures. Patterns of ERPs effectively predicted later behavioral scores.

**4pSCb54. Magnetoencephalography as a tool to study speech perception in awake infants.** Toshiaki Imada (University of Washington, Dept. of Speech & Hearing Sciences, and Institute for Learning & Brain Sciences, Box 357988, Seattle, WA 98195, USA, imada@u.washington.edu), Alexis N. Bosseler (University of Washington, Dept. of Speech & Hearing Sciences, and Institute for Learning & Brain Sciences, Box 357988, Seattle, WA 98195, USA, bosseler@u.washington.edu), Samu Taulu (Elekta-Neuromag Oy, 22 Elimäenkatu, 00510 Helsinki, Finland,

samu@squid.neuromag.fi), Elina Pihko (BioMag Laboratory, Helsinki University Central Hospital, 00029 Helsinki, Finland, pihko@biomag.hus.fi), Jyrki Mäkelä (BioMag Laboratory, Helsinki University Central Hospital, 00029 Helsinki, Finland, jyrki.makela@hus.fi), Antti Ahonen (Elekta-Neuromag Oy, 22 Elimäenkatu, 00510 Helsinki, Finland, antti.ahonen@elekta.com), Patricia K. Kuhl (University of Washington, Dept. of Speech & Hearing Sciences, and Institute for Learning & Brain Sciences, Box 357988, Seattle, WA 98195, USA, pkkuhl@u.washington.edu)

Magnetoencephalography (MEG) provides a safe, noninvasive method for studying the developing brain by offering reliable localization of the brain regions activated during speech processing. However technical challenges make recording awake infants difficult. The small size of the infant head in the adult-sized helmet results in a low signal-to-noise ratio. Head and limb movement, which is typical of young infants, produces signal artifact that is difficult to overcome during signal processing. This study used MEG to study phonetic processing in awake, non-sedated typically developing infants from 5 to 16 months. The recordings were made using the Elekta Neuromag® 306-channel instrument at BioMag Laboratory, Helsinki University Central Hospital, Finland. Infants listened to speech syllables produced by a loudspeaker inside the magnetically shielded room. Newly developed signal processing methods and behavioral entertainment greatly improved the quality of the data, producing 29 successful infant recordings out of 35 attempts. We describe the methods, as well as removal of movement-modulated artifacts, efficient interference suppression, and movement compensation during data analysis. Whole-head MEG recordings in awake babies a few months old are now feasible.

**4pSCb55. Adapting second language phonemic perception training to common instructional situations: A progress report.** Thomas R. Sawallis (Univ. of Alabama, English Dept., Box 870244, Tuscaloosa, AL 35487, USA, tsawalli@bama.ua.edu), Michael W. Townley (Univ. of Alabama, English Dept., Box 870244, Tuscaloosa, AL 35487, USA, townl003@bama.ua.edu)

Although current L2 pedagogy de-emphasizes phoneme-level pronunciation training, laboratory experiments demonstrate benefits from training non-natives in perception of difficult target-language phonemic contrasts. Specifically, evidence shows that: learners' perceptual performance improves (Jamieson & Morosan, 1986; Flege, 1995), improvements generalize to new talkers and words (Lively, Logan, & Pisoni, 1993), perceptual training triggers production improvements (i.e., without production training, Bradlow et al., 1997), and both perceptual (Lively et al., 1994) and production improvements (Bradlow et al., 1999) are maintained over several months. We are adapting perceptual training methods from such studies for use in common instructional situations, starting with Japanese students learning the English /l-r/ contrast. This paper addresses three practical concerns. First, studies show that the perceptual training needs tokens with multiple kinds of variation. We discuss phonological variation in corpus design, sociolinguistic variation in talker recruitment, and inducement of within-talker variations. Second the training must be usable both independently and in classrooms. We discuss the design of sessions short enough for inclusion as part of daily classroom activities. Third, the training must be computer-controlled. We discuss necessary functions, available programs, and our choice. Finally, we briefly demonstrate the training, and give a sketch of interim results.

**4pSCb56. "Polyaural" array processing for robust automatic speech recognition in noisy and reverberant environments.** Richard M. Stern (Carnegie Mellon University, Department of Electrical and Computer Engineering and Language Technologies Institute, 5000 Forbes Avenue, Pittsburgh, PA 15213, USA, rms@cs.cmu.edu), Evandro B. Gouvea (Carnegie Mellon University, Department of Electrical and Computer Engineering and Language Technologies Institute, 5000 Forbes



Avenue, Pittsburgh, PA 15213, USA, egouvea@cs.cmu.edu), Kshitiz Kumar (Carnegie Mellon University, Department of Electrical and Computer Engineering and Language Technologies Institute, 5000 Forbes Avenue, Pittsburgh, PA 15213, USA, kshitizk@cs.cmu.edu)

It is well known that human binaural processing is very useful for separating incoming sound sources as well as for improving the intelligibility of speech in reverberant environments. In this paper we present a new method of signal processing for robust speech recognition using multiple microphones. The method, loosely based on the human binaural hearing system, consists of passing the speech signals detected by multiple microphones through bandpass filtering and nonlinear halfwave rectification operations, and then cross-correlating the outputs from each channel within each frequency band. These operations provide rejection of off-axis interfering signals. These operations are repeated (in a non-physiological fashion) for the negative of the signal, and an estimate of the desired signal is obtained by combining the positive and negative outputs. We demonstrate that the use of this approach provides substantially better recognition accuracy than delay-and-sum beamforming using the same sensors for target signals in the presence of additive broadband and speech maskers, and it provides substantial improvements in specific reverberant environments as well. [Supported by NSF and DARPA]

**4pSCb57. Coupled 2D-Fluid-Structure-Acoustic Simulation of the Human Voice.** Gerhard Link (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, gerhard.link@lse.eei.uni-erlangen.de), Manfred Kaltenbacher (Univ. Erlangen-

Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, manfred.kaltenbacher@lse.eei.uni-erlangen.de), Michael Doellinger (University Hospital Erlangen, Department of Phoniatrics and Pediatric Audiology, Medical School, Bohlenplatz 21, 91054 Erlangen, Germany, michael.doellinger@uk-erlangen.de), Reinhard Lerch (Univ. Erlangen-Nuremberg, Dept. of Sensor Technology, Paul-Gordan-Str. 3/5, 91052 Erlangen, Germany, reinhard.lerch@lse.eei.uni-erlangen.de)

The human voice is a key factor in social life. If the phonation process is disturbed due to a disease as e.g. hoarseness, communication and social life are strongly affected. Therefore, it is necessary to enhance therapies in order to minimize affliction caused by a disease. There exist different approaches to improve therapies. Experimental field studies of different physical parameters, like acoustic pressure or vocal fold displacements are one possibility. Another promising approach to advance insight into laryngeal dynamics is given by numerical simulations. Due to the growing computing power the complexity of numerical phonation models is steadily increasing and full fluid-structure-acoustic interacting models are now feasible. Therefore, we have developed a two-dimensional numerical model for the human phonation process, which includes the complete fluid-structure-acoustic interaction. As discretization method the finite-element method was applied for all possible three physical fields. The fluid-structure and the structure-acoustic interactions are based on general continuum mechanical principles; the fluid-acoustic interaction is based on Lighthill's acoustic analogy. Therefore, the analysis of all sound mechanisms, which consist of the eddy-induced, the volume-induced, and the mechanical-induced sound can be performed. First simulation results will be presented and discussed.

THURSDAY AFTERNOON, 3 JULY 2008

P3-C, LEVEL 3, 3:40 TO 5:20 P.M.

### Session 4pSPa

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

Sean Lehman, Cochair  
*Pleasanton, CA*

Christian Pichot, Cochair

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

### Contributed Papers

**4pSPa1. Detection and classification using the Estimated Ocean Detector.** Richard Lee Culver (ARL Penn State, PO Box 30, State College, PA 16804, USA, rlc5@psu.edu), Colin W. Jemmott (ARL Penn State, PO Box 30, State College, PA 16804, USA, cwj112@psu.edu), Brett E. Bissinger (ARL Penn State, PO Box 30, State College, PA 16804, USA, beb194@psu.edu), Nirmal K. Bose (ARL Penn State, PO Box 30, State College, PA 16804, USA, bkn@engr.psu.edu)

We have developed the Estimated Ocean Detector, a likelihood ratio receiver with an estimator-correlator structure, and applied it to detection and classification of underwater acoustic signals. The receiver requires that the

noise probability density function (pdf) to belong to the exponential class but need not be Gaussian. A composite hypothesis is employed in order to incorporate knowledge (or predictions) of the signal parameter statistics. Previously, receiver performance was demonstrated for Gaussian noise and sinusoidal signals with known frequency and phase and whose amplitude pdfs were predicted using knowledge of the ocean environment and an acoustic propagation program. In order for the receiver to be useful operationally, it must be able to accommodate unknown signal phase and incorporate numerical estimates of noise and signal parameter pdfs. Progress toward satisfying these requirements is reported in this talk. Work supported by the Office of Naval Research Undersea Signal Processing.



#### 4pSPa2. Fractal analysis of signals of the seismic acoustic emission.

Alexander V. Glushkov (Odessa University, P.O.Box 24a, 65009 Odessa-9, Ukraine, glushkov@paco.net), Andrey A. Svinarenko (Odessa University, P.O.Box 24a, 65009 Odessa-9, Ukraine, glushkov@paco.net), Yaroslav I. Lepikh (Odessa University, P.O.Box 24a, 65009 Odessa-9, Ukraine, glushkov@paco.net)

Paper is devoted to analysis fractal properties of signals of the seismic acoustic emission in periods between earthquakes. Earlier it has been carried out an analysis of natural data on distribution of the earthquake hypercenters (Mukhamedov, 1992; Salimi et al, 1993). To reveal the fractal properties for signals of the seismic acoustic emission the joint wavelet analysis is carried out by using the non-decimated wavelet transform (Glushkov et al, 2004, 2005). We present the fractal processing data for envelopes of signal of the seismic acoustic emission in different ranges of frequencies. The dependences of the Herst indicator and fractal dimension curve on the lengths of considered intervals are presented. It takes a place crossover upon a behaviour with the Herst indicator  $H=0.4-0.6$ . It is given the physical interpretation of the seismic acoustic emission. References: Mukhamedov V.A., *Izv. Russian Acad.Sci.* 3, 39 (1992). Salimi M., Robertson M., Sammis S., *Phys. Rev.Lett.* 70, 2186 (1993). Glushkov A.V. et al, *Nonlinear Processes in Geophys.* 11, 285 (2004). Glushkov A.V. et al, *Atmospheric Res. (Elsevier)* 77, 100 (2005).

**4pSPa3. A torpedo detection and 3-D tracking system.** Nico Roosnek (Roosnek Research & Development, Vlaskamp 82, 2592 AC The Hague, Netherlands, nico@roosnek.nl)

Ship torpedo defence systems are ideally based on 3D information of incoming torpedoes. Therefore a Torpedo Detection and Location (TDL) sonar system has been designed based on a transmitting transducer, two receiving arrays and optimal signal processing with 3D-tracking capabilities with optimal estimation. The transmitted signal is a chirp. The signal processing with intrinsic beamforming is based on the phase of the signal and on optimal estimation (Kalman filtering). For testing the capabilities of such a design a test system on detection and tracking of a torpedo-like object in the atmosphere has been developed. Some figures about its capability, which can be extrapolated to underwater situations, will be given.

#### 4pSPa4. Chirplet Transform Analysis for Ultrasonic Inspection of Composite Materials.

Abdessalem Benammar (Image and signal processing laboratory. Welding and NDT Centre, Route de Dely-Ibrahim, BP 64, Chéraga, 16035 Alger, Algeria, Abs\_benammar@yahoo.fr), Redouane Draï (Image and signal processing laboratory. Welding and NDT Centre, Route de Dely-Ibrahim, BP 64, Chéraga, 16035 Alger, Algeria, drai\_r@yahoo.fr), Ahmed Kechida (Image and signal processing laboratory. Welding and NDT Centre, Route de Dely-Ibrahim, BP 64, Chéraga, 16035 Alger, Algeria, Abs\_benammar@yahoo.fr), Abderrezak Guessoum (Image and signal processing laboratory. Welding and NDT Centre, Route de Dely-Ibrahim, BP 64, Chéraga, 16035 Alger, Algeria, Abs\_benammar@yahoo.fr)

In this work, a successive parameter estimation algorithm based on the chirplet transform is presented. The chirplet transform is used not only as a means for time frequency representation, but also to estimate the echo parameters, including the amplitude, time-of-arrival, center frequency, bandwidth, phase, and chirp rate. We initially apply this method to simulated signals with additional structural noise. These signals contain several echo defects, closer between them. This stage permits to see the robustness of the

developed algorithm. Thereafter, we validate all simulated results by experimental results obtained on composite material with and without delamination defects.

**4pSPa5. Source localisation on a single hydrophone.** Grégoire Le Touzé (GIPSA-lab, dep. DIS, 961, rue de la Houille Blanche, 38402 St Martin d'Hères, France, gregoire.letouze@lis.inpg.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)

The aim of the study is to localise an underwater Ultra Low Frequency (ULF) source in a shallow water environment. The acoustic signal is recorded on a single hydrophone and the source, which has to be short in time, is unknown. To perform the localisation, we have developed modal filters based on time-frequency techniques. Different localisation techniques are proposed: - Conventional Matched Mode Processing: results are good for range estimation but contain error on source depth estimation. We show that those errors are due to the ULF band. - Source depth estimation technique based on mode amplitude estimation: this estimation is precise but presents ambiguities. - Range estimation technique using mode phase estimation: we recently developed this method which also estimates mode signs. The source range is precisely estimated and mode sign estimation avoids ambiguity on source depth estimation combined with the previous method. These methods are validated on real data coming from the North Sea.

#### 4pSPa6. Analyze effects of the flow on the vocalic reduction and the coarticulation in sequences CV of pharyngal Arabic.

L Falek (Université des sciences et de la technologie Houari Boumédiène (USTHB), Laboratoire de communication parlée et de traitement du signal, Faculté d'Electronique et d'Informatique, BP 32, El Alia, Alger, Algeria, lilalcppts@yahoo.fr), O. Bouferroum (Université des sciences et de la technologie Houari Boumédiène (USTHB), Laboratoire de communication parlée et de traitement du signal, Faculté d'Electronique et d'Informatique, BP 32, El Alia, Alger, Algeria), A. Djeradi (Université des sciences et de la technologie Houari Boumédiène (USTHB), Laboratoire de communication parlée et de traitement du signal, Faculté d'Electronique et d'Informatique, BP 32, El Alia, Alger, Algeria)

The degree of coarticulation and the vocalic reduction (RV) are indices related to good engine control (Gay 1978). Fowler (1998) explains why locus equation (LE) is used to characterize, at the same time, the place of articulation and the degree of coarticulation between consonants and vowels: a strong slope ( $m=1$ ) indicates a maximum coarticulation between consonants and vowels (i.e. minimal resistance of the coarticulation), while a weak slope ( $m=0$ ) indicates absence of coarticulation between consonants and vowels (maximum resistance of the coarticulation). The bond between the degree of coarticulation and the RV can be explained according to the linear relation between F2onset and F2milieu: the modifications of values of F2milieu will affect those of F2onset and consequently those of the slopes. In this study, the analysis of the vocalic reduction and slopes of the equations of locus, carried out on CV (extracts starting from sentences) in standard Arabic pronounced by speakers having different mother tongues (near to Arabic standard and very far away from standard Arabic), and at speed of variable elocution, revealed a vocalic reduction and a variation of the slope of the locus equation, specific to each speaker, who seems to be related to his mother tongue. El Tamimi (2006) carried out a similar study with normal flow, in dialectical Arabic and in French, with normal flow, an influence of the mother tongue showed on the vocalic reduction and the slope of the equation of locus.

## Session 4pSPb

**Signal Processing in Acoustics: Advances in Acoustic Sensors and Networks for Defense Applications II  
(Poster Session)**

Brian Ferguson, Cochair  
*Defence Science and Technology Organisation*

Pierre Naz, Cochair  
*ISL*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

**Contributed Papers**

**4pSPb1. Aircrafts localisation and tracking with arrays of microphones.** Gaetano Caronna (Università La Sapienza - Dept. Fisica Tecnica, Via Eudossiana 18, 00184 Roma, Italy, gaetano.caronna@uniroma1.it), Pierluigi Testa (Università La Sapienza - Dept. Fisica Tecnica, Via Eudossiana 18, 00184 Roma, Italy, pierluigi.testa@uniroma1.it)

University of Roma "La Sapienza" will participate in a project, financed by the Italian Ministry of Research, aiming to detect and to track aircrafts near the airport using the acoustic emissions of the aircraft and the technique of acoustic beam-forming. Although the principle is well known and some applications are reported in literature and explored also by our group in a previous European FP6 project ("Safe Airport"), the final performance and the engineering value are not assessed. In this research effort the problem is again raised aiming to create a more realistic model including wind and temperature gradient that, in particular conditions, can drastically impair the validity of the results, estimating the related errors. A simulation software was written considering the steering drive on the microphones outputs, being the mechanical rotation of the array prohibited, due to the aerodynamic noises induced. At least two arrays, opportunely located, are requested in order to perform a triangulation and detect the source of the acoustic wave emitted by the aircraft. The precision in the localisation of the acoustic source was estimated in a simulation where real aircraft acoustic emissions and environmental noise were used as input.

**4pSPb2. Acoustic Source Localization via Distributed Sensor Networks using Tera-scale Optical-Core Devices.** Neena Imam (Oak Ridge National Laboratory, 1 Bethel Valley Road, Oak Ridge, TN 37831-6015, USA, imamn@ornl.gov), Jacob Barhen (Oak Ridge National Laboratory, 1 Bethel Valley Road, Oak Ridge, TN 37831-6015, USA, barhenj@ornl.gov)

For real-time acoustic source localization applications, one of the primary challenges is the considerable growth in computational complexity associated with the emergence of ever larger, active or passive, distributed sensor networks. The complexity of the calculations needed to achieve accurate source localization increases dramatically with the size of sensor arrays, resulting in substantial growth of computational requirements that cannot be met with standard hardware. One option to meet this challenge builds upon the emergence of digital optical-core devices. The objective of this work was to explore the implementation of key building block algorithms

used in underwater source localization on optical-core digital processing platform recently introduced by Lenslet Inc. We investigate key concepts of threat-detection algorithms such as Time Difference Of Arrival (TDOA) estimation via sensor data correlation in both time and frequency domains with the purpose of implementation on the optical-core processor. We illustrate our results with the aid of numerical simulation and actual optical hardware runs. The major accomplishments of this research, in terms of computational speedup and numerical accuracy achieved via the deployment of optical processing technology, should be of substantial interest to the acoustic signal processing community.

**4pSPb3. Performance of high-resolution sensor array processing algorithms in the localization of acoustic sources.** Joseph Lardies (FEMTO-ST Applied Mechanics, 24 chemin de l'épitahe, 25000 Besançon, France, joseph.lardies@univ-fcomte.fr), Hua Ma (University of Franche-Comté, Institute FEMTO - LMARC, 24 rue de l'Epitahe, 25000 Besançon, France, rosyhorse@hotmail.fr), Marc Berthillier (University of Franche-Comté, Institute FEMTO - LMARC, 24 rue de l'Epitahe, 25000 Besançon, France, marc.berthillier@univ-fcomte.fr), Emmanuel Foltete (FEMTO-ST Applied Mechanics, 24 chemin de l'épitahe, 25000 Besançon, France, emmanuel.foltete@univ-fcomte.fr)

The localization of noise sources from a specified direction may often be accomplished with an array of sensors. One commonly used processor consists of delay and add networks: a conventional beamformer, however its spectrum suffers from the Rayleigh resolution and its performance is highly degraded, specially in lower frequency range. In the communication, the performance of some typical high-resolution sensor array processing algorithms: Minimum Variance, MUSIC, Mini-Norm algorithms are investigated for wideband source location. Their performances are compared with a new source localization algorithm which is based on a sparse representation of sensor measurements with an overcomplete basis composed of samples from the array manifold. The key of the method is the use of the SVD for data reduction and the formulation of a joint multiple-sample sparse representation problem in the signal subspace domain. Increased resolution and improved robustness to noise is obtained with this algorithm applied to various numerical examples.

## Session 4pSPc

## Signal Processing in Acoustics: Filter Design, Detection, and Estimation II (Poster Session)

Michael Roan, Cochair  
Virginia Tech

Jorge Quijano, Cochair  
NEAR Lab-Portland State University

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pSPc1. Averaged Lagrange Method for interpolation filter.** Jonathan Andrea (LIEN - BP 239, Université Henri Poincaré, 54506 Vandoeuvre, France, jonathan.andrea@lien.uhp-nancy.fr), Frederic Coutard (LIEN - BP 239, Université Henri Poincaré, 54506 Vandoeuvre, France, frederic.coutard@lien.uhp-nancy.fr), Patrick Schweitzer (LIEN - BP 239, Université Henri Poincaré, 54506 Vandoeuvre, France, patrick.schweitzer@lien.uhp-nancy.fr), Etienne Tisserand (LIEN - BP 239, Université Henri Poincaré, 54506 Vandoeuvre, France, etienne.tisserand@lien.uhp-nancy.fr)

This paper presents a new method for Lagrange interpolation for reducing distortions without introducing any complexity. The aim is to improve the linearity of the phase and the gain responses of the interpolation filter by an averaging method. A first FIR interpolation filter of the second order computes the values between three successive samples of the input signal. At the same time, a second filter, identical to the first, computes the values between  $(x_{k-2}, x_{k-1}, x_k)$ . Finally, the common values between  $(x_{k-2}, x_{k-1})$  provided by the two filters are averaged two by two. This double interpolation can be simply done with a single third order filter and with a Farrow structure filter. Compared to the usual Lagrange's third order interpolation filter, the behaviour of the filter we propose is more regular especially in the high frequencies of the Nyquist band. More, the filter coefficients are easier to calculate. The designed filter is tested on a FPGA of Altera. The results shows that the method significantly reduces distortions and improves quality of the frequency response.

**4pSPc2. Application of digital filters for measurement of nonlinear distortions in loudspeakers using Wolf's method.** Rafal Siczek (Wrocław University of Technology, Wybrzeże Wyspiańskiego 27, 50-370 Wrocław, Poland, rafal.siczek@pwr.wroc.pl)

The design of digital filters used for measurement of nonlinear distortions in loudspeakers by the Wolf's method is presented in the paper. The Wolf's method has been developed in 1953 and originally the analog filters was applied. This method requires particularly steep slopes of both band-reject filters (at the transmitting part of the measurement system) and band-pass filters (at the receiving part). The digital filters allows for achievement of such steep slopes with a relatively small cost in comparison with analog ones. The very high attenuation in the barrage bands can be achieved. In the very narrow transition band any irregularities do not appear. The design procedures using MATLAB package have been described. Then, the digital filter allow for achievement of a high accuracy of measurement.

**4pSPc3. Objective quality measurement of the excitation of impact sounds in a source/filter model.** Mathieu Lagrange (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, mathieu.lagrange@mcgill.ca), Bruno L. Giordano (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, bruno.giordano@music.mcgill.ca), Philippe Depalle (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, depalle@music.mcgill.ca), Stephen McAdams (Centre for Interdisciplinary Research in Music Media & Technology (CIRMMT) - Schulich School of Music - McGill Univ., 555 Sherbrooke Street West, Montreal, QC H3A1E3, Canada, smc@music.mcgill.ca)

For the modeling of percussive (non-sustained) sounds, the excitation signal can be estimated from an original sound in several ways, usually by a time-domain deconvolution process. The source signal obtained by such a process cannot be compared with the original excitation because it is usually unknown. Hence in most of the approaches available in the literature, the validation of the deconvolution process is quantified in terms of spectral flatness, i.e. a source signal is considered as a good estimation of the excitation when most of the resonant content has been removed. However, the excitation signal is usually a percussive burst, the time domain properties of which are known to be very important, at least perceptually speaking. To evaluate the time domain properties of the estimated excitation, we propose in this paper to compare the estimated excitation to the recording of the acceleration of the hammer hitting a plate. In the recordings considered, the evolution of the acceleration of the hammer has a specific pattern with several peaks due to the bouncing of the hammer on the suspended plate. This specific pattern allows us to propose a metric that can be useful for objectively measuring the quality of the estimation process.

**4pSPc4. Detecting Scenes in Lifelog Videos based on Probabilistic Models of Audio data.** Kiichiro Yamano (Hosei University, 3-7-2 Kajinocho, 184-8584 Koganei, Japan, n04k1035@cis.k.hosei.ac.jp), Katunobu Itou (Hosei University, 3-7-2 Kajinocho, 184-8584 Koganei, Japan, itou@hosei.ac.jp)

Lifelog videos are recorded every activity in everyday lives. To utilize them efficiently, it is required to be indexed automatically. To index continuous shots of the lifelog, significant scenes are detected automatically. For detection, many researches employ image features such as color and edge, however, the accuracy is insufficient. In this study, we propose

probabilistic models for scene detection from lifelog video. In this method, mel-frequency filter bank output of audio tracks of the lifelog videos is modeled statistically with hand-labeled training data. We tested the proposed method to use train station scene. We collected 11 hour sound data for such scenes. To analyze them, we defined seven categories, such as stopping trains, passing trains, starting trains, waiting, and so on. Our method achieved to 100% for waiting scene and 18.4% in average.

**4pSPc5. Estimation of reflection location by the correlation coefficient function.** Hideo Shibayama (Shibaura Institute of Technology, 3-7-5, Toyosu, Koto-ku, 135-8548 Tokyo, Japan, sibayama@sic.shibaura-it.ac.jp), Takeshi Araya (Hitachi Information & Communication Engineering, Ltd, 393 Totsukamati Totsuka-ku, 244-8502 Yokohama Kanagawa, Japan, takeshi.araya.tx@hitachi.com), Yoshiaki Makabe (Hitachi Information & Communication Engineering, Ltd, 393 Totsukamati Totsuka-ku, 244-8502 Yokohama Kanagawa, Japan, yoshiaki.makabe.hd@hitachi.com), Eiji Okarura (Hitachi Information & Communication Engineering, Ltd, 393 Totsukamati Totsuka-ku, 244-8502 Yokohama Kanagawa, Japan, eiji.okamura.mh@hitachi.com)

Gas is supplied to households through gas pipes that branch out from a main pipe. It is essential to carry out maintenance of these branch pipes to ensure the safe supply of gas. However, there are many cases in which the state of buried gas pipes and the connection condition of a large number of pipes are unknown because no detailed information is included in piping drawings or only plan views are available. If the piping arrangement can be estimated on the basis of response waves originating from an acoustic wave travelling in a gas pipe emitted by a loudspeaker, which is placed at one end of the pipe, this method will be important for the maintenance of branch pipes. To realize this, we report a method of estimating the length of gas pipes using correlation coefficient between a driving signal and measured waves. And, we show the experimental results with high accuracy.

**4pSPc6. An experiment for signal identification of the MIMO communication by sonic waves.** Daisuke Hayashi (Shibaura Institute of Technology, 3-7-5, Toyosu, Koto-ku, 135-8548 Tokyo, Japan, m107077@sic.shibaura-it.ac.jp), Yasukazu Maeda (Shibaura Institute of Technology, 3-7-5, Toyosu, Koto-ku, 135-8548 Tokyo, Japan, m107092@sic.shibaura-it

.ac.jp), Toru Itakura (Shibaura Institute of Technology, 3-7-5, Toyosu, Koto-ku, 135-8548 Tokyo, Japan, m106009@sic.shibaura-it.ac.jp), Hideo Shibayama (Shibaura Institute of Technology, 3-7-5, Toyosu, Koto-ku, 135-8548 Tokyo, Japan, sibayama@sic.shibaura-it.ac.jp)

Recently, the communication method by MIMO works in a field of the wireless communication. We do research on application to the MIMO system in an acoustic field by sonic wave. As the transmission method, we use the space division multiplexing (SDM) that is the method for the purpose of the improvement of the transmission rate that accepted the number of the transmission elements by sending plural signals at the same time. It is important to identify each signal from several different signals that are transmitted at the same frequency band. And, we study method for detecting each signal. This paper describes influence of the signal detection for the different conditions of the multi-path propagation.

**4pSPc7. Transverse vectorization of fast Fourier transforms on multicore architectures.** Travis Humble (Oak Ridge National Laboratory, 1 Bethel Valley Road, Oak Ridge, TN 37831-6015, USA, humblets@ornl.gov), Jacob Barhen (Oak Ridge National Laboratory, 1 Bethel Valley Road, Oak Ridge, TN 37831-6015, USA, barhenj@ornl.gov), Michael Traweck (Office of Naval Research, 875 North Randolph Street, Arlington, VA 22203, USA, Mike.Traweck@navy.mil)

Single-instruction, multiple-data (SIMD) multicore computing architectures, such as the IBM Cell Broadband Engine Architecture, offer new opportunities for quickly and efficiently calculating the 1D-FFT of acoustic signals, as time-sampled data arrays can be naturally partitioned across the multiple cores on which vectorized implementations of the FFT operate. Building on this parallel pipeline model, we consider the case that  $M$  data arrays of length  $N$  reside within each core. Whereas the cost of sequentially executing these  $M$  FFT's conventionally scales as  $\alpha M N \log_2 N$ , we demonstrate a transverse vectorization solution whose cost scales as  $\alpha \beta N \log_2 N$ , where  $\alpha$  and  $\beta$  are constant scaling factors. Our approach makes use of the SIMD instruction set and large vector register file inherent to each core of the IBM Cell in order to calculate the FFT of  $M$  data arrays simultaneously. By efficiently using all the available vector registers in performing the FFT, this transverse SIMD vectorization solution further reduces the computational complexity of the conventional parallel pipeline model.



## Session 4pUWa

## Underwater Acoustics and ECUA: Image and Signal Processing

Ivars Kirsteins, Cochair

NUWC, 1176 Howell St, Newport, RI 02841, USA

Ioannis Koukos, Cochair

Hellenic Naval Academy, Telecommunications Lab, Terma Hatzikyriakou Street, Piraeus, 18539, Greece

## Contributed Papers

2:00

**4pUWa1. Physics-based signal processing methodologies for separating target echoes into their constituent elastic and geometric components.**

Ivars Kirsteins (NUWC, 1176 Howell St, Newport, RI 02841, USA, kirsteinsip@npt.nuwc.navy.mil), Alessandra Tesei (NATO Undersea Research Centre, Viale San Bartolomeo 400, 19126 La Spezia, Italy, tesei@nurc.nato.int)

Echoes from even simple canonical symmetric shapes such as solid elastic spheres are highly complex, consisting of a superposition of specular, elastic, and geometric diffractive components closely spaced in time and frequency. Although measurement of the individual elastic and geometric components is important for analysis of scattering physics and object identification, in practice individual components are often hard to measure because of mutual interference between components. When an object interacts with a boundary, such as the sea floor, additional interfering echoes make the object signature even more complicated to interpret. Motivated by this problem, a frequency-domain technique is developed for separating an echo into its constituent elastic and geometric components, and for isolating specular reflections from a possibly interacting boundary. It is based on modelling the echo components in the frequency domain as amplitude-modulated piecewise-linear chirps, utilizing physically-inspired group delay models and constraints, and then using a modified Costas' residual signal analysis (RSA) algorithm in combination with chirp transform analysis to decompose the echo. The RSA scheme is demonstrated on actual echoes collected in the NURC EVA-06 sea trial from spherical and cylindrical target shapes and is shown to work well with the separated components verified against modelling predictions.

2:20

**4pUWa2. Two Dimensional Wavelet Coefficient Statistics for Sea Bottom Classification.**

Ioannis Koukos (Hellenic Naval Academy, Telecommunications Lab, Terma Hatzikyriakou Street, 18539 Piraeus, Greece, jalex\_14u@hotmail.com), Theodoros Mavroidis (Hellenic Naval Academy, Telecommunications Lab, Terma Hatzikyriakou Street, 18539 Piraeus, Greece, tmavroid@tellas.gr), Georgios Vardoulis (Hellenic Naval Academy, Telecommunications Lab, Terma Hatzikyriakou Street, 18539 Piraeus, Greece, ggvard@gmail.com)

In this paper we examine the classification of different seafloors based on the analysis of images obtained by side-scan sonar. For this purpose, we apply various two dimensional multilevel wavelet decomposition schemes on images obtained from three different seafloor types, i.e., sand ripples, rocks and sands, and then we examine the statistics of the corresponding wavelet coefficients. The observed Probability Density Functions (pdf) are modeled using various theoretical distributions such as the Alpha Stable, Sum of Gaussians, Log-normal. The parameters of the fit are subsequently used to classify the side scan sonar images according to well known cluster analysis techniques. The use of the energy of the wavelet coefficients as a tool for side scan sonar image classification is also evaluated. A new unsupervised classification scheme based on the pdf fitting parameters is proposed.

2:40

**4pUWa3. Computer Vision Techniques Applied for Reconstruction of Seafloor 3D Images from Side Scan and Synthetic Aperture Sonars Data.**

Krzysztof Bikonis (Gdansk University of Technology, Department of Geoinformatics, Narutowicza 11/12, 80-952 Gdansk, Poland, binio@eti.pg.gda.pl), Andrzej Stepnowski (Gdansk University of Technology, Department of Geoinformatics, Narutowicza 11/12, 80-952 Gdansk, Poland, astep@pg.gda.pl), Marek Moszynski (Gdansk University of Technology, Department of Geoinformatics, Narutowicza 11/12, 80-952 Gdansk, Poland, marmo@eti.pg.gda.pl)

The Side Scan Sonar and Synthetic Aperture Sonar are well known echo signal processing technologies that produce 2D images of the seafloor. Both systems combine a number of acoustic pings to form a high resolution image of seafloor. It was shown in numerous papers that 2D images acquired by such systems can be transformed into 3D models of seafloor surface by algorithmic approach using intensity information, contained in a grayscale image. The paper presents the concept of processing the Side Scan Sonar and Synthetic Aperture Sonar records for detailed reconstruction of 3D seafloor using Shape from Shading techniques. Shape from Shading is one of the basic techniques used in computer vision for the objects reconstruction. The algorithms proposed in the paper use the assumed Lambert model of backscattering strength dependence on incident angle and utilize additionally the information from shadow areas for solving obtained set of equations. The idea was verified by simulation study. The obtained results of 3D shape reconstruction are presented and the performance of the algorithms is discussed.

3:00

**4pUWa4. Underwater vehicle attitude estimation using Hough transformation.**

Hisashi Shiba (Radio Application Division, NEC Corporation, 1-10, Nisshin-cho, Fuchu, 183-8501 Tokyo, Japan, h-shiba@aj.jp.nec.com)

In these decades various imaging sonar systems have been developed. They are very effective for underwater investigations such as geologies, fisheries, resource surveys and securities. To extend these system abilities we studied new imaging sonar applications and found they are also useful in underwater vehicle navigations. In this presentation I propose a new method of estimating vehicle attitude parameters relative to the ocean surface or the seafloor by the onboard imaging sonar. The ocean surface or the seafloor is often obtained as a plane in sonar images. However, the images are sometimes too noisy to identify the plane locations because of ambient noise and reverberations by texture patterns on the planes. The new method describes these planes with plane equation parameters connected to attitude parameters, and searches the most appropriate parameter sets using a voting procedure which is a three dimensional expansion of the Hough transformation widely applied in line detections on noisy two dimensional images. The method provides the plane location and attitude parameters simultaneously without other sensors. After the algorithm explanation multi-beam imaging sonar simulations are shown and accuracies are evaluated.

3:20

**4pUWa5. Adaptive coding/modulation for shallow-water UWA communications.** Sanjay Mani (Arizona State University, Dept. of Electrical Engineering, Tempe, AZ 85287-5706, USA, sanjay.mani@asu.edu), Tolga M. Duman (Arizona State University, Dept. of Electrical Engineering, Tempe, AZ 85287-5706, USA, duman@asu.edu), Paul Hursky (HLS Research, Inc., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA 92037, USA, paul.hursky@hlsresearch.com)

We consider adaptive modulation and coding techniques for Phase Shift Keying (PSK) transmission schemes over underwater acoustic (UWA) channels. Adaptive modulation and coding can be an effective means of obtaining higher data rates while retaining acceptable error levels by exploiting

knowledge of the channel state. This is particularly significant in the UWA scenario, where attained spectral efficiencies are critical since usable bandwidth is a severe limitation. We examine the use of channel capacity and post-equalization signal to noise ratio (SNR) as adaptation metrics. We illustrate the ideas both through simulations and using the results of a recent experiment (AUVfest 2007). In the experiment, in order to evaluate effectiveness of adaptive coding/modulation for UWA communications, we have transmitted a bank of signals with varying spectral efficiencies (obtained by changing the number of transmit elements, modulation scheme, and code rate) back to back. Using the received signals, we illustrate the relevance of the channel condition metrics under consideration, that is, we evaluate the proposed metrics for various scenarios, and provide indications of the transmission rates that could have been achieved reliably.

THURSDAY AFTERNOON, 3 JULY 2008

AMPHI BORDEAUX, 3:00 TO 5:40 P.M.

### Session 4pUWb

## Underwater Acoustics and ECUA: High Frequency Variability II

Marcia Isakson, Cochair

*Applied Research Laboratories, University of Texas, PO Box 8029, Austin, TX 78713-8029, USA*

Thomas Folégot, Cochair

*NATO Undersea Research Center, Viale San Bartolomeo 400, La Spezia, 19126, Italy*

### Invited Papers

3:00

**4pUWb1. Examples of high frequency variability in underwater acoustic systems.** Peter Stein (Scientific Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063, USA, pstein@scisol.com)

Invited paper for the structured (special) session UW09 High Frequency Variability In this paper we will look at examples of high frequency variability encountered by the author during the design and implementation of underwater acoustic systems over the last 20 years. These systems include those for radiated noise measurements, short baseline tracking, tomographic correction of numerical ocean models, marine mammal detection, and diver detection. The examples will highlight the different physical sources of variability and their effects on system performance. Methods used for mitigating the effects both in the physical design of the sensor system and in the signal processing will also be discussed.

3:20

**4pUWb2. Gaussian beam tracing for high-frequency acoustics.** Michael B. Porter (HLS Research, Inc., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA 92037, USA, michael.porter@hlsresearch.com), Martin Siderius (HLS Research, Inc., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA 92037, USA, martin.siderius@hlsresearch.com), Paul Hursky (HLS Research, Inc., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA 92037, USA, paul.hursky@hlsresearch.com)

Gaussian beam tracing is an approach that constructs full-wave beams around the skeleton of conventional ray theory. This Gaussian beam approach leads to a very simple algorithm and provides remarkable accuracy and speed. For high-frequency, broadband applications, Gaussian beams are often the only practical approach, as the standard full-wave modeling alternatives are often thousands of times slower. Gaussian beams are derived using high-frequency asymptotics, and therefore fit naturally to certain current areas of interest in HF acoustics, such as acoustic communications. However, what often surprises people who are not familiar with the technique is that it works quite well at lower frequencies, depending on the water depth. That fact, is really a consequence of 20 years of continued advances in the Gaussian beam method. We will review those developments in the context of HF variability, considering effects of boundary, volume, and source/receiver dynamics.

3:40-5:00 Posters

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

## Invited Papers

5:00

**4pUWb3. Field-calibration: exploiting high-frequency mobile platform transmissions for source localization at lower frequency with arrays.** Paul Hursky (HLS Research, Inc., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA 92037, USA, paul.hursky@hlsresearch.com), Michael B. Porter (HLS Research, Inc., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA 92037, USA, michael.porter@hlsresearch.com)

Multipath arrivals in an ocean waveguide with a reflective enough bottom can be used as a fingerprint for source range and depth and this has been demonstrated by many researchers using matched field processing. However, MFP relies upon acoustic propagation models to produce the Green's functions to be used as steering vectors for this processing. We have proposed to instead directly measure these Green's functions using wideband acoustic comms signals from mobile platforms such as AUVs that are increasingly part of naval applications. Thus we measure the multipath arrival pattern at 8-16 kHz from the many locations AUVs are visiting, and then apply such measurements for locating other sources at low frequency. A key factor in this process is being able to capture the essential features of the impulse response function at high frequency, where fluctuations are much more severe than at low frequency. We have previously presented experimental results of applying this technique using a single hydrophone receiver. Here we will report on a continuation of that work, using vertical arrays and data from the RADAR '07 experiment.

5:20

**4pUWb4. The effects of sediment variability on reflection coefficient measurements.** Marcia Isakson (Applied Research Laboratories, University of Texas, PO Box 8029, Austin, TX 78713-8029, USA, misakson@arlut.utexas.edu), Nicholas P. Chotiros (Applied Research Laboratories, University of Texas, PO Box 8029, Austin, TX 78713-8029, USA, chotiros@arlut.utexas.edu)

The statistical distribution of specularly scattered acoustic energy, commonly known as the reflection coefficient, is an important parameter when developing models for shallow water propagation and acoustic communications. The distribution of measured reflection coefficient data from 5 to 50 kHz and 10 to 70 degrees grazing angles was taken from a sea bottom recently perturbed by hurricane off the coast of Florida at the Sediment Acoustic Experiment 2004 (SAX04). The width and shape of the distributions are attributed to varying sediments in the experimental area and interface roughness. These distributions are analyzed to determine the scattering from different types of sediments including sand and mud. The effects of both roughness scattering and sediment variability on the mean

value and distribution of the measured reflection coefficient will be explored. [Work sponsored by ONR Ocean Acoustics.]

THURSDAY AFTERNOON, 3 JULY 2008

P3-C, LEVEL 3, 3:40 TO 5:20 P.M.

## Session 4pUWc

### Underwater Acoustics and ECUA: Tank Experiments II (Poster Session)

Raymond Soukup, Cochair  
*U.S. Naval Res. Lab.*

Jean-Pierre Sessarego, Cochair  
*Laboratory for Mechanics and Acoustics CNRS*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

## Contributed Papers

**4pUWc1. Acoustical and optical measurements on a mixture of air microbubbles in water.** Vincent Duro (LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, vincent.duro421@univ-lehavre.fr), Dominique Rajoana (LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, raphael.rajoana@univ-lehavre.fr), Dominique Decultot (LOMC FRE 3102 CNRS Groupes Ondes Acoustiques, Université du Havre (IUT), Place Robert Schuman, 76610 Le Havre, France, dominique.decultot@univ-lehavre.fr), Gerard Maze (LAUE, Université du Havre, Place Robert Schuman, F-76610 Le Havre, France, gerard.maze@univ-lehavre.fr)

An important challenge for current naval research is the modernization of battleships. Their target detection system must be increasingly efficient and they must be increasingly undetectable. Due to turbulent flows and bubbles, ship wakes are a detectable acoustic signature and ship bow waves disturb sonar detection. In this work, we study sound propagation through bubble clouds in water. We have developed an experimental set-up which permits us to acquire, in synchronization, acoustical signals and optical images. The phenomenon of bubble monopole resonance in very low frequency, related to bubble size, provokes effects of strong sound damping and sound speed dispersion. These experimental results, related to theoretical results, permit to estimated sizes and concentrations of bubbles. The ac-

quired bubble images permit to know the real bubble sizes and concentrations, in order to correlate with the experimental acoustical results. Air bubbles are generated with a high pressure water jet introduced into the host liquid medium. A hydrodynamic study is done to characterize the bubble jet. We present in this work, theoretical results establishing a complex effective wave number characterizing the sound propagation in an effective medium. All results are discussed and compared with results of others papers on this subject.

**4pUWc2. Acoustic methods in extremely shallow water: validity of the far field approximation.** Fantina Madricardo (CNR-Istituto di Scienze Marine, Riva Sette Martiri - Castello 1364/a, 30122 Venice, Italy, fantina.madricardo@ismar.cnr.it), Silvano Buogo (CNR-Istituto di Acustica 'O.M.Corbino', Via del Fosso del Cavaliere, 100, 00133 Rome, Italy, silvano.buogo@idac.rm.cnr.it), Paola Calicchia (CNR-Istituto di Acustica 'O.M.Corbino', Via del Fosso del Cavaliere, 100, 00133 Rome, Italy, paola.calicchia@idac.rm.cnr.it), Emiliano Boccardi (CNR-Istituto di Scienze Marine, Riva Sette Martiri - Castello 1364/a, 30122 Venice, Italy, E.Boccardi@lse.ac.uk)

In this paper we present some experimental results concerning the possibility of using traditional echosounder in extremely shallow water environments. In the framework of the Echos Project, a collaboration between ISMAR and IA, a wide area of the Venice Lagoon has been explored with a traditional echosounder. Since the explored area was extremely shallow (up to 50 cm), it became necessary to check experimentally whether the far field approximation was still valid in such conditions. In this work the acoustic field of the echosounder ELAC LAZ72 used for the sub bottom investigation is experimentally characterized and the validity of the far field approximation is verified.

**4pUWc3. Acoustic measurements of bubbles in the wake of ship model in tank.** Alexander Sutin (Stevens Institute of Technology, Castle Point on Hudson, Hoboken, NJ 07030, USA, asutin@stevens.edu), Alexander Benilov (Stevens Institute of Technology, Castle Point on Hudson, Hoboken, NJ 07030, USA, Alexander.Benilov@stevens.edu), Heui-Seol Roh (Stevens Institute of Technology, Castle Point on Hudson, Hoboken, NJ 07030, USA, Heui-Seol.Roh@stevens.edu), Young In Nah (Agency of Defense Development, P.O.Box, ChinHae, 645-600 Kyung Nam, Republic of Korea, yinah@add.re.kr)

The interest to bubble generation by moving ships is connected with large area of bubble wake reaching several kilometers that can be used for ship detection. We developed acoustic system for the measurements of bubble density and conducted measurements in 100 m long towing tank. The developed system measured attenuation of ultrasound in wide frequency band from 50 to 800 kHz between two acoustic sensors placed on the distance 20cm. The attenuation of sound produced by bubbles was observed during several minutes after model of ship passed the point of measurement. The attenuation was recalculated to the bubble size distribution for bubbles from 4 to 65 microns using theory of resonance bubble attenuation. The measured bubble size distribution can be interpolated by power dependence  $n(R) \sim R^{-3.5}$  that is typical for bubbles at sea subsurface layer. The dependencies of bubble concentration of model ship speed and type of propeller were investigated. The generation of bubbles was observed when the model speed exceeds definite threshold that can be connected with the cavitation threshold. The theory describing dynamics of wake turbulence based on the shear-free turbulent wake was developed. The measured bubble concentration was in good agreement with the developed theory.

THURSDAY AFTERNOON, 3 JULY 2008

P3-C, LEVEL 3, 3:40 TO 5:20 P.M.

## Session 4pUWd

### Underwater Acoustics and ECUA: Determination of Acoustic Properties of Materials for Sonar Applications II (Poster Session)

Kenneth Foote, Cochair  
*Woods Hole Oceanographic Institution*

Stephen Robinson, Cochair  
*National Physical Laboratory*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

#### Contributed Paper

**4pUWd1. Experimental study on resistance and noise reduction of low functional surfaces hydrophobic coatings.** Lin Zhang (Harbin Engineering University, 31st Building Wenmiao Street, 150001 Harbin, China, zhanglin8624@yahoo.com.cn), Ting Jiang (Harbin Engineering University, 31st Building Wenmiao Street, 150001 Harbin, China, jianfly@163.com)

The object moves in viscous fluid will accepting two kinds of resistance, one kind is the friction drag that the fluid applied to the object directly, another kind is the pressure drag that because of separating the fluid and taking from, and this drag force can create the flow noise. On theoretical aspect, this article analyses the relation between physical properties of the material

and flow resistance and flow noise. The conclusion is that the surface material has the effect of reducing the flow resistance and noise because of its experimental on reducing the mutual action between fluid and solid surface under given conditions. On experimental aspect, use the gravitational low noise water tunnel test condition and adopt torpedo model, many times have the test on the drag and noise reduction of the torpedo model under diversity flow rate, have coating and not coating situation. Indicate by analysis that the low surface energy dewatering coatings used in experiment have the certain drag and noise reduction effect under the high flow rate, and with the flow rate power-up, the effect of the drag and noise reduction becomes better and better.



## Session 4pUWe

**Underwater Acoustics and ECUA: General Topics in Underwater Acoustics (Poster Session)**

David L. Bradley, Cochair  
*Pennsylvania State University*

Kevin LePage, Cochair  
*Naval Research Laboratory*

All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

*Contributed Papers*

**4pUWe1. A solution to the problem of simultaneous classification and localization of underwater objects from their acoustic field.** Andrey I. Mashoshin (Central Scientific and Research Inst. 'Elektropribor', 30, Malaya Posadskaya Street, 197046 St. Petersburg, Russian Federation, amashoshin@eprib.ru)

This paper presents one of the solutions to the problem of simultaneous classification and localization of underwater objects from its acoustical field. It was shown that both problems (classification and localization) have some common features that allow to consider them as one mutual problem which can be solved by maximum likelihood method. The solution takes into account: - the fundamental physical representations of the transmission, propagation, reception of signals, and the measurement of their parameters against the noise background; - the target parameters vector describing features of its noise radiation and motion; - the parameters vector characterizing the observation conditions; - the errors vector of the signal parameter measurements. The theoretical solution is illustrated by solving a relatively simple but practically important problem of classification and ranging of an underwater object from passive underwater acoustic observation data.

**4pUWe2. On acoustic tomography method physical advantages in long range ocean inhomogeneities control.** Andrew Semenov (Acad. N.N. Andreev's Acoustics Institute, Russ. Acad. of Sci., 4 Shvernik Street, 117036 Moscow, Russian Federation, asemen@akin.ru)

An acoustic tomography method based on observation of signals transecting an inhomogeneity is discussed. While for various inhomogeneities (ships, underwater objects, sea mammals etc.) short range monitoring of back scattered sonar reflections could be used, their monitoring at longer ranges is still a problem. This is due to the fast decay of sonar signals with distance, especially in unfavorably absorbing shallow water regions where there is functional distance dependence of optimum frequency. New perspectives on ocean climate change measurement (ATOC) require monitoring of extra long (500 - 1000 km) tomography traces, where ocean vortices and icebergs could hamper the progress. The physical limits (optimum frequencies and efficiencies) for two tomographic methods for deep and, most unfavorable in the author's view, shallow ocean regions are compared. For several types of inhomogeneities tomographic measurement optimization estimates are discussed. For ranges up to 500 km the acoustic tomography method has substantial advantages with respect to conventional sonar monitoring even for unfavorable shallow water regions at least up to 300 km distance and in all deep water regions. Tomographic measurements are also indispensable in long-range ocean vortices monitoring.

**4pUWe3. A hydrophone calibration centre for the Mediterranean area.** Giovanni Bosco Cannelli (CNR-Istituto di Acustica 'O.M.Corbino', Via del Fosso del Cavaliere, 100, 00133 Rome, Italy, giovannibosco.cannelli@idac.rm.cnr.it), Silvano Buogo (CNR-Istituto di Acustica 'O.M.Corbino', Via del Fosso del Cavaliere, 100, 00133 Rome, Italy, silvano.buogo@idac.rm.cnr.it), Paola Calicchia (CNR-Istituto di Acustica 'O.M.Corbino', Via del Fosso del Cavaliere, 100, 00133 Rome, Italy, paola.calicchia@idac.rm.cnr.it)

The Underwater Acoustics Laboratory of the Institute of Acoustics "O. M. Corbino" in Rome, Italy, has recently achieved accreditation for hydrophone calibration. It can offer hydrophone calibration services operating under a management system compliant with UNI EN ISO/IEC standard and traceability of measurements to NPL primary standards. Precise, traceable measurements are needed today in all applications of underwater acoustics, where a faithful response of measuring devices is essential for a correct evaluation of the system performance and for a rigorous assessment of the environmental impact of man-made activities at sea. To attain these requirements the Laboratory has a water tank 6 m long, 4 m wide and 5.5 m deep, equipped with a motorized two-trolley positioning system capable of handling loads of up to 100 kg. A variety of test and measurement equipment is available, controlled by a computer acoustic calibration system, and standalone instruments for signal generation, acquisition and analysis. Besides, two windows are provided on one of the tank walls for combined acoustical and optical measurements, which can be performed using the available Laser Doppler Vibrometer (up to 100 kHz) and the 16-mm film rotating prism high-speed camera (up to 10 000 frames per second).

**4pUWe4. Characteristics of sound pressure field focused by an acoustic aplanat lens.** Toshiaki Nakamura (National Defense Academy, 1-10-20 Hashirimizu, 239-8686 Yokosuka, Japan, toshiaki@nda.ac.jp), Yuji Sato (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, yuji@aclab.esys.tsukuba.ac.jp), Ayano Miyazaki (National Defense Academy, 1-10-20 Hashirimizu, 239-8686 Yokosuka, Japan, g44044@nda.ac.jp), Kazuyoshi Mori (National Defense Academy, 1-10-20 Hashirimizu, 239-8686 Yokosuka, Japan, kmori@nda.ac.jp)

In this paper, we describe the characteristics of an acoustic aplanat lens which can eliminate both spherical and coma aberrations. A singlet aplanat lens was designed using a ray theory in paraxial area. Sound pressure fields of the bi-concave aplanat lens and a spherical lens with 160 mm in diameter for the frequency of 500 kHz were evaluated by a three-dimensional finite difference time domain (3-D FDTD) method. The aplanatic lens showed better convergence characteristics than spherical lenses. Two bi-concave lenses of aplanat and spherical lenses were made by acrylic resin and water tank experiment was conducted to compare the on-axis characteristics and

beam patterns at the focus of them. As a result, the aplanat lens could focus higher sound pressure than the spherical one within the incident angle of 10 degrees.

**4pUWe5. Echo analysis of objects on the seafloor.** Xiukun Li (College of Underwater Acoustic Engineering, Harbin Engineering University, 150001 Harbin, China, xiukun\_li@yahoo.com.cn), Huiguang Chi (College of Underwater Acoustic Engineering, Harbin Engineering University, 150001 Harbin, China, xiukun\_li@yahoo.com.cn)

Echoes from elastic objects on the seafloor comprise two kinds of acoustic components as well as bottom reverberation. One is elastic scattering echoes, and the other is geometric echoes. These echoes are called highlights, which can be used for identification of objects. Because time delay among highlights from small objects is very short, neither the wavelets resolved highlights structure nor suppressed bottom reverberation according to research. In order to obtain highlights structure and decay reverberation, the Hilbert Huang Transform (HHT), which is a new idea for analyzing nonlinear and nonstationary is applied in the paper. The HHT can describe the data from the instantaneous frequency and energy rather than the global frequency and energy defined by the Fourier spectral analysis, and the adjustable window Fourier spectral analysis defined by the wavelet. The results from both extensive simulations and real data show that the HHT has the highest resolution in time and frequency domain, and may prove to be a vital method for identification of objects on the sea floor.

**4pUWe6. Tidal Effect of Reciprocal Sound Propagations at the Experiment in Hashirimizu Port.** Hanako Ogasawara (National Defense Academy, 1-10-20 Hashirimizu, 239-8686 Yokosuka, Japan, ogasawar@nda.ac.jp), Toshiaki Nakamura (National Defense Academy, 1-10-20 Hashirimizu, 239-8686 Yokosuka, Japan, toshiaki@nda.ac.jp), Kazuyoshi Mori (National Defense Academy, 1-10-20 Hashirimizu, 239-8686 Yokosuka, Japan, kmori@nda.ac.jp), Koichi Mizutani (Tsukuba Univ., Tsukuba Science City, 305-8573 Ibaraki, Japan, mizutani@esys.tsukuba.ac.jp)

Acoustical monitoring method could monitor wide area such as ocean because it can spatially measure the measurement object with a few sensors. It is important to monitor ocean structure changes for understanding global climate changes and for controlling aquatic resources. We investigated sound propagation characteristics using sound propagated data measured at Hashirimizu port of Yokosuka, Japan in August 2006 and 2007. The experimental area was very shallow water with the average depth of 5 m and travel distance of 110 m. M-sequence signal alternately propagated every 30 seconds from each bank side. Water depth change caused by tide affected received signal amplitude traveled with the frequency of 12 kHz and 80 kHz.

It was also confirmed from simulated sound pressure by FDTD method through the change of assumed water depth. Furthermore, water temperature fluctuation could be confirmed from the travel time changes.

**4pUWe7. Reduced Scale Experiment of Frequency Dependence of Single Spherical Biconcave Acoustic Lens for Ambient Noise Imaging.** Kazuyoshi Mori (National Defense Academy, 1-10-20 Hashirimizu, 239-8686 Yokosuka, Japan, kmori@nda.ac.jp), Hanako Ogasawara (National Defense Academy, 1-10-20 Hashirimizu, 239-8686 Yokosuka, Japan, ogasawar@nda.ac.jp), Toshiaki Nakamura (National Defense Academy, 1-10-20 Hashirimizu, 239-8686 Yokosuka, Japan, toshiaki@nda.ac.jp)

By the numerical analysis results using the Finite Difference Time Domain method in our previous studies, it was supposed that the spherical biconcave lens with an aperture diameter of 2.0 m has sufficient directional resolution (for example, the beam width is 1 deg at 60 kHz) for realizing the Ambient Noise Imaging (ANI) system. In this study, to confirm the directional resolution of the lens in a wide frequency band of 20-100 kHz, we performed a reduced scale experiment of one-fifth space in a water tank. The lens, made of acrylic resin, has an aperture diameter of 400 mm and a radius of curvature of 500 mm. A burst pulse of 25 cycles at 100, 200, 300, and 500 kHz, in which the frequency increases 5 times, was radiated from the sound source to the lens. The results show that the -3 dB area, whose pressure is 3 dB lower than the maximum at the image point, does not overlap each other at 300 and 500 kHz. It is supposed that the lens of 2.0 m aperture has the fine resolution over 60 kHz in the original scale.

**4pUWe8. Suppression of side lobe level on the cone characteristics of the directivity pattern of an antenna as an important factor in its directivity index and effective aperture.** Zvonimir M.S. Milosic (MORH, Trg kralja Petra Kresimira IV, HR-10000 Zagreb, Croatia, zvonimir.milosic@morh.hr)

This paper will present a universal procedure for precise de-embedding of the directivity index and effective aperture dependence of the suppressed side lobes on measured cone directivity patterns of sonar antennas. The procedure is derived on an idealized model of cone characteristics and directivity patterns of antennas. It is universally applicable to any contemporary sonar or hydroacoustic communication system with a cone directivity pattern. In accordance with the given expressions of directivity index and effective aperture of circle baffled pistons, this paper will present substitute analytical functions for hypothetical circle antennas with suppressed minor lobes. The presented graphs of functional behavior in a two-dimensional system, and in a three-dimensional coordinate system, are an excellent base for standardization, unification and quality evaluation of sonar or radar systems and also medical scanners. Key words: directivity index of sonar antenna, idealized model of cone directivity pattern, level of suppressed side lobes, effective aperture

**Session 4pUWf****Underwater Acoustics and ECUA: Sound Propagation in 3-Dimensional Environments I (Poster Session)**

David Calvo, Cochair  
*U.S. Naval Res. Lab.*

Michael Taroudakis, Cochair  
*University of Crete & FORTH/IACM*

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***Contributed Paper***

**4pUWf1. On a new approach to numerical modeling of a low-frequency underwater sound in 2 and 3-dimensional oceanic waveguides.** Oleg E. Gulin (Il'ichev Pacific Oceanological Institute of Far-East Branch of Russian Academy of Sciences, 43, Baltiyskaya st., 690068 Vladivostok, Russian Federation, gulinoe@rambler.ru)

A new method to compute underwater sound fields in irregular planar waveguides is proposed. It realizes the full two-way propagation approach and exploits an idea of a problem solution dependence on a certain variable parameter that is the position of a boundary of the irregular region. With respect to this parameter for waveguide modes, an initial value problem can

be formulated in the horizontal plane that is completely equivalent to the boundary value problem for the original wave equation (Helmholtz equation). This fact allows simulation of sound fields in waveguides based on ordinary differential equations with traditional approximations and for arbitrary source distance from the irregular region and the degree of irregularities. Examples of a simulation for a 2-D irregular waveguide model with an upslope rigid or absorbing penetrable bottom are presented for low frequencies and shallow sea conditions. They illustrate the strong difference between our solution and approximate solutions that arise due to both mode coupling and considerable backscattering within the considered irregular waveguide models

**Session 4pUWg****Underwater Acoustics and ECUA: Nonlinear Acoustic Methods in Searching for Buried Objects**

Benjamin R. Dzikowicz, Cochair  
*Naval Surface Warfare Center, Panama City Division, Code HS-11, 100 Vernon Ave., Panama City, FL 32407, USA*

Eugeniusz Kozaczka, Cochair  
*Univ., Narutowicza 11/12, Gdansk, 80-952, Poland*

***Invited Papers*****5:20**

**4pUWg1. Parametric sonars in searching of buried objects.** Eugeniusz Kozaczka (Univ., Narutowicza 11/12, 80-952 Gdansk, Poland, kozaczka@pg.gda.pl)

The subject of this paper is the description of usage of the parametric sonar for searching of the objects that are on the surface of the seabed or very close to it. Searching of underwater objects, especially these ones buried in the seabed has a very practical meaning. Presently more often mass destruction weapon is placed in the very difficult way to find it. Moreover searching of the objects of the archaeological character at sea requires usage of devices those have possibility of penetration of sediment which covers the searched object. In this case the most useful of acoustic means are parametric sonars that due to their specific features are good tools for underwater searching. The principle of detection of buried objects is similar to detection in the case of usage of the classical sonars. The dispersion of the sound field enables to track the object and sometimes enables to define the shape of the searched object. The measurement equipment can be connected to the Global Positioning System (GPS) and complementary equipment that is necessary in such kind of investigations. There will be presented experimental results that were obtained during the trials in the Gdansk Bay. Also will be shown the typical images for chosen objects.

**4pUWg2. Observation of sub-bottom sediments in the Southern Baltic by means of nonlinear acoustic method.** Grazyna Grelowska (Polish Naval Academy, Smidowicza 69, 81-103 Gdynia, Poland, ggrel@wp.pl), Eugeniusz Kozaczka (Univ., Narutowicza 11/12, 80-952 Gdansk, Poland, kozaczka@pg.gda.pl), Ignacy Gloza (Polish Naval Academy, Smidowicza 69, 81-103 Gdynia, Poland, i.gloza@amw.gdynia.pl)

The aim of the paper is to present results of preliminary experimental investigation using parametric echosounder in natural conditions for profiling the subbottom sediments in the Gulf of Gdansk. The ability to predict seabed properties: seafloor roughness, sub-bottom structure and discrete scatterers laying on the seafloor or buried into sediments, from remotely sensed data is important especially in regions that need permanent monitoring. Precise determination of seabed structure or localisation of buried objects in the sand requires the use of a low frequency signal to penetrate the sediment and a narrow beam to provide high-resolution data. Both requirements can be achieved with a parametric (nonlinear) technique. The special experimental setup has been arranged allowing penetration of bottom sediments as well as precise positioning and following a given route. Measurements were conducted exactly along given routes, for that geological profiles have been taken by means of another method. It allows to compare detailed data obtained by means of parametric echosounding to ones given at geological map. In all investigations the primary frequency of the array was of 100 kHz, whereas the secondary frequency changed in range 5 kHz - 15 kHz.

6:00

**4pUWg3. Transmission of low frequency sounds from ships into water environment.** Ignacy Gloza (Polish Naval Academy, Smidowicza 69, 81-103 Gdynia, Poland, i.gloza@amw.gdynia.pl)

The main goal is presentation of the broad experimental research associated with the transmission of acoustic energy generated by moving objects into the water. I present the methodology of evaluation of the transmission of low frequency vibration from the mechanisms of ship into the water environment. For that purpose the research of vibration distribution over ship's hull were conducted along with the hydroacoustic field. The underwater sound measurements were performed both for sailing condition and during anchorage. The hull is source of complex surface shape inducing waves of different frequencies depending on the working parameters of ship's machinery. As a result of mechanical incitement of the hull's surface a complex distribution of vibration is created depending on vibration spectra of the primary sources as for the main propulsion units, generators or other mechanisms. This research allowed to determine the transmission coefficient for the mechanical vibration energy going through the ship's hull and the evaluation of influence on the hydroacoustic field of a ship. The values signaling the changes of the technical condition of machinery and propulsion system.

6:20

**4pUWg4. Detection of a resonant target buried in sediment using iterative time reversal: mid-frequency pond experiments.** Benjamin R. Dzikowicz (Naval Surface Warfare Center, Panama City Division, Code HS-11, 100 Vernon Ave., Panama City, FL 32407, USA, benjamin.dzikowicz@navy.mil), Zachary J. Waters (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, zjwaters@bu.edu), R. Glynn Holt (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, rgholt@bu.edu), Ronald A. Roy (Boston University, Dept. of Aerosp. and Mech. Eng., 110 Cummington St., Boston, MA 02215, USA, ronroy@bu.edu)

Iterative time reversal techniques developed at smaller scales, [Waters et al., J. Acoust. Soc. Am. 122, 3023 (2007)], are applied to the detection of a 15 cm diameter stainless steel shell buried in sandy sediment at the acoustic test pond at the Naval Surface Warfare Center - Panama City Division. A mid-frequency, directional projector is located 1.5 m above the sediment and directed normally to it. A hydrophone is located midway between the sediment and the projector. This system gives a response between 20 kHz and 200 kHz. A calibration filter is designed using the direct path response between the projector and hydrophone. This filter is applied at each time reversal iteration to prevent the time reversal technique from converging to the transducer resonance. Application of iterative time reversal allows the detection of the target at greater depths than otherwise possible due to the resonance scattering of the target. Additional experiments explore the application of filters, and the effect of different window sizes. Also, comparisons with similar laboratory experiments and comments on real world applications are discussed. [Work supported by the Office of Naval Research.]

6:40

**4pUWg5. Sonar detection of targets buried under seafloor ripple at shallow grazing angles.** 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), 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)

This paper summarizes results from modeling and measurement efforts investigating shallow grazing angle reverberation levels from a rippled bottom and subcritical detection of targets buried under such interfaces. The focus of this work is associated with frequencies less than 10 kHz where evanescent transmission is important. Measurements were performed in a 13.7-m deep, 110-m long, 80-m wide test-pool with a 1.5-m layer of sand on the bottom. Rippled contours were artificially formed with the aid of a sand scraper. A parametric sonar that generated difference frequency signals in the 1 to 20 kHz frequency range was placed onto a rail system permitting acquired data to be processed and displayed similar to that of a side scan sonar. The buried target was a solid aluminum cylinder. The seabed roughness was measured to assess ripple fidelity and to estimate the small-scale roughness spectrum which was used in scattering models to calculate the backscattered signal levels from the target and bottom. Acoustic backscatter data obtained for various ripples parameters (wavelengths, heights, orientation, etc.) were compared to model predictions based on perturbation theory. [Work supported by the Office of Naval Research and the Strategic Environmental Research and Development Program, USA].



**4pUWg6. Changing of scattering properties of underwater objects covered by elastic shell.** Grazyna Grelowska (Polish Naval Academy, Smidowicza 69, 81-103 Gdynia, Poland, ggrel@wp.pl), Ignacy Gloza (Polish Naval Academy, Smidowicza 69, 81-103 Gdynia, Poland, i.gloza@amw.gdynia.pl)

In certain situations it is of a great importance to decrease reflection properties of underwater objects positioned into water, deployed on a seabed or buried into sediments. This effect could be achieved by means of covering shell changing acoustic parameters of considered object. The aim of the paper is to present the idea of performing the covering material for underwater objects as well as results of experimental investigation of its scattering characteristics. Acoustic scattering measurements were performed in laboratory condition on solid air-filled objects of sphere or ellipsoidal shape with or without covering shell. The objects were insonified by acoustic source at frequency in range of 60 kHz - 150 kHz.

### *Contributed Paper*

7:20

**4pUWg7. Experimental study of parametric transmission: Simultaneous generation of two beams.** Maud Amate (GESMA, BP42, 29240 Brest Armées, France, maud.amate@dga.defense.gouv.fr), Pierre Cervenka (Institut Jean le Rond D'Alembert, UMR 7190, 2 Place de la Gare de Ceinture, 78210 Saint Cyr L'Ecole, France, cvk@ccr.jussieu.fr), Jacques Marchal (Institut Jean le Rond D'Alembert, UMR 7190, 2 Place de la Gare de Ceinture, 78210 Saint Cyr L'Ecole, France, jmarchal@ccr.jussieu.fr)

In order to respond to the buried mines threat, an approach of GESMA (Groupe d'Etudes Sous-Marines de l'Atlantique) is to take advantage of the

parametric transmission in sonar imagery. This technique allows producing narrow beamwidths at low frequencies. Such a parametric transmitter has been designed and calibrated in collaboration with an academic laboratory of Paris VI, and the targeted benchmark is fulfilled:  $2^\circ$  @ 20 kHz, with sufficient source level. The design of the complete sonar system is based on a sequential multibeam transmission associated with a synthetic aperture technique at receive. However, the shortfall of the sweeping technique lies in the timing limitation. One solution to increase the coverage rate is the simultaneous generation of several parametric beams in different directions. Experimentations have been conducted in a tank with different signals: CW, LFM, Ricker. This presentation addresses the results of these experimentations in the context of buried mines detection and classification.

THURSDAY AFTERNOON, 3 JULY 2008

ROOM 362/363, 5:20 TO 7:40 P.M.

### **Session 4pUWh**

## **Underwater Acoustics, Computational Acoustics, and ECUA: Numerical Methods in Underwater Scattering**

Andrew F. Seybert, Cochair

*University of Kentucky, Dept. of Mech. Engineering, 151 RGAN Bldg., Lexington, KY 40506, USA*

Martin Ochmann, Cochair

*Technische Fachhochschule Berlin, Univ. of Applied Sciences, Luxemburger Str. 10, Berlin, 13353, Germany*

Bodo Nolte, Cochair

*Forschungsanstalt der Bundeswehr für Wasserschall und Geophysik, Klausdorfer Weg 2-24, Kiel, 24148, Germany*

### *Invited Papers*

5:20

**4pUWh1. BIE and coupled BIE-FEM methods for scattering from underwater objects.** Ilkka Karasalo (FOI, Gullfösgatan 6, SE 16490 Stockholm, Sweden, ilkka.karasalo@foi.se)

Two techniques are described for computation of acoustic scattering by a smooth elastic object in a fluid space or a fluid layer of a layered fluid-solid medium. Both techniques use a frequency-domain boundary integral equation (BIE) method for the exterior layered medium, with the Green's function computed numerically by adaptive high-order transform integration and exact finite elements. The surface of the object is parameterized by a smooth map of the unit sphere, and the BIE is discretized by B-splines, point collocation and numerical integration in polar-like coordinates centered at the collocation point. Two approaches for handling the interior dynamics of the object are described. In the first, applicable to objects structured into homogeneous layers separated by closed smooth interfaces, the interior dynamics is formulated as a system of BIEs with displacements and tractions at the interfaces as unknowns. In the second, the interior is discretized by a general-purpose finite element (FEM) method, and an impedance matrix connecting tractions and displacements on the exterior surface is computed by eliminating interior degrees of freedom by Gaussian elimination. Computational results on test cases of the 2006 target strength benchmark workshop at NURC are presented.

**4pUWh2. Calculation of the acoustic target strength of elastic objects based on BEM-BEM-coupling.** Ralf Burgschweiger (Technische Fachhochschule Berlin, Univ. of Applied Sciences, Luxemburger Str. 10, 13353 Berlin, Germany, burgi@tfh-berlin.de), Martin Ochmann (Technische Fachhochschule Berlin, Univ. of Applied Sciences, Luxemburger Str. 10, 13353 Berlin, Germany, ochmann@tfh-berlin.de), Bodo Nolte (Forschungsanstalt der Bundeswehr für Wasserschall und Geophysik, Klausdorfer Weg 2-24, 24148 Kiel, Germany, bodonolte@bwb.org)

Based on a BEM-BEM-coupling method, the scattered pressure from elastic objects placed in water and partially buried in the sediment is calculated. For this reason, a special variant of the boundary element method (BEM) is implemented. It contains a pre- and a postprocessor with 3D visualization, in order to define the geometry of the scattering objects in the interface layer between fluid and sediment and the parameters needed for characterizing the fluid and the elastic material. The solver is able to perform numerical calculations in a multiple parallel manner. For the solution of the underlying system of linear equations, we use different kinds of approximate and direct solution techniques. Simple acoustical exterior problems, for example, the sound scattering by elastic solid cylinders and spheres placed in a fluid are treated by the BEM-BEM-coupling method. The results will be compared with analytical solutions or solutions obtained from other numerical methods.

### Contributed Papers

6:00

**4pUWh3. Modeling of scattering from targets in an oceanic waveguide using Kirchhoff/diffraction method.** Keunhwa Lee (Seoul National University, San 56-1, Silim, Kwanak, 11111 Seoul, Republic of Korea, nasalkh2@snu.ac.kr), Woojae Seong (Seoul National University, Room. 306, Bd. 34, San 56-1, Sillim-dong, Kwanak-gu, College of Engineering, Dept. of Naval Architecture and Ocean Engineering, 151-744 Seoul, Republic of Korea, wseong@snu.ac.kr), Yongtaek Joo (Daewoo shipbuilding & marine engineering CO.,LTD., Aju-dong, Geosje city, 11111 Geosje, Republic of Korea, yhsaga@dsme.co.kr)

The target scattering model in an oceanic waveguide is presented. The target scattered pressure field is formulated using the generalized Green's function method [F. Ingenito, J. Acoust. Soc. Am. 82, 2051-2059 (1987)]. The concept of Kirchhoff/diffraction method is introduced in order to simplify the Fredholm integral equation. In numerical analysis, complex target is divided into numerous polygon facets, whose analytic solution for scattered field is derived based on the waveguide solution by the ray or normal mode theory. This solution is used in constructing the target scattered field for complex target. Comparison between ray and normal mode based target scattering model is shown. Finally discussion for conditions of the source/receiver and target which improves the numerical efficiency is given.

6:20

**4pUWh4. Modeling ocean reverberation under short pulse conditions.** Henry Weinberg (Alion Science Incorporated, 23 Colonial Drive, Waterford, CT 06385, USA, chic@ct.metrocast.net)

The Comprehensive Acoustic System Simulation (CASS) is a standard model for predicting ocean reverberation. However, the current version, CASS V4.1, is known to have theoretical and numerical difficulties when investigating short pulse lengths. This is due to two model requirements: (1) the time increment for sampling reverberation should not exceed the pulse length; and (2) the range increment for sampling the environment should not exceed half the sound speed-pulse length product. Unless these requirements are met, certain phenomena, such as time splitting, may not be accurately modeled. In addition, the predicted results may have an unrealistic step function appearance. On the other hand, very small time and range increments often lead to excessive computational requirements. A simple modification to CASS V4.1 appears to have relaxed the current increment requirements substantially. The range increment must still be small enough to sample environmental features, but not to the extent dictated by small pulse lengths. Although the modification is based on a well known mathematical method for accelerating convergence, its success in modeling reverberation was unexpected.

6:40

**4pUWh5. Comparison of two monostatic reverberation models based on ray theory.** Youngmin Chu (Seoul National University, Room. 306, Bd. 34, San 56-1, Sillim-dong, Kwanak-gu, College of Engineering, Dept. of Naval Architecture and Ocean Engineering, 151-744 Seoul, Republic of Korea, sks655@snu.ac.kr), Woojae Seong (Seoul National University,

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Two monostatic reverberation models applicable for range dependent environment are presented. Considering the roughness of both the sea surface and bottom, we calculate the reverberation signal, based on the ray propagation model and scattering model using the composite-roughness theory and/or other empirical formulae. First model computes the reverberation time signal using two-way eigenray searching between source/receiver position along the discrete scattering area (dA). The other model uses one-way eigenray searching along the discrete ray angle (d $\theta$ ) which generates a non-uniform scattering area providing less accurate solution at a reduced computational burden. The time series calculated from both models are shown and compared with the solutions presented in the Reverberation Modeling Workshop (Nov. 2006, Austin, TX).

7:00

**4pUWh6. A novel modelling approach for sound propagation analysis in a multiple scatterer environment.** Bert Van Genechten (K.U.Leuven - Dept. of Mechanical Engineering, Celestijnenlaan 300B - bus 2420, 3001 Heverlee, Belgium, Bert.VanGenechten@mech.kuleuven.be), Bart Bergen (K.U.Leuven - Dept. of Mechanical Engineering, Celestijnenlaan 300B - bus 2420, 3001 Heverlee, Belgium, Bart.Bergen@mech.kuleuven.be), Bert Pluymers (K.U.Leuven - Dept. of Mechanical Engineering, Celestijnenlaan 300B - bus 2420, 3001 Heverlee, Belgium, Bert.Pluymers@mech.kuleuven.be), Dirk Vandepitte (K.U.Leuven - Dept. of Mechanical Engineering, Celestijnenlaan 300B - bus 2420, 3001 Heverlee, Belgium, Dirk.Vandepitte@mech.kuleuven.be), Wim Desmet (K.U.Leuven - Dept. of Mechanical Engineering, Celestijnenlaan 300B - bus 2420, 3001 Heverlee, Belgium, Wim.Desmet@mech.kuleuven.be)

In the past decades element-based numerical modelling techniques have become a commonly used and invaluable tool for the analysis of acoustic radiation and scattering problems. However, the pollution errors associated with the element discretisation inherent to these methods increase with the increasing size of the problem domain. As a result, the applicability of these methods for radiation and scattering problems in which the source and receiver positions are located far from each other is often prohibited. The Wave Based Method (WBM) is an alternative deterministic prediction method for the analysis of steady-state acoustic problems. It is based on an indirect Trefftz approach in that wave functions, which are exact solutions of the underlying differential equation, are used to describe the dynamic response. The enhanced computational efficiency of the WBM as compared to the element based methods has been shown already for the analysis of both finite and (semi-)infinite acoustic problems. This paper introduces a novel WBM-based methodology to model the acoustic source-receiver transfer path functions in a multiple scatterer environment. A sound propagation validation case illustrates the potential of the proposed approach.

**4pUWh7. The basis choice and the reconstruction of combined refractive-kinetic inhomogeneities in the problems of ocean acoustic tomography.** Andrey Shurup (Department of Acoustics, Physics Faculty, Moscow State University, Leninskie Gory, 119991 Moscow, Russian Federation, burov@phys.msu.ru), Sergey Sergeev (Department of Acoustics, Physics Faculty, Moscow State University, Leninskie Gory, 119991 Moscow, Russian Federation, sergeev@su29.ru), Valentin Burov (Department of Acoustics, Physics Faculty, Moscow State University, Leninskie Gory, 119991 Moscow, Russian Federation, burov@phys.msu.ru)

The characteristics of the acoustical signal in ocean are determined by both the refractive inhomogeneities (the perturbation of sound velocity) and the presence of ocean currents. The methods of acoustical tomography can

be applied to the simultaneous reconstruction of both refractive and kinetic inhomogeneities. In this paper the problem of the combined refractive-kinetic inhomogeneities reconstruction by the tomography methods is considered. For the realization of proposed scheme so called band basis consisted of a number of intersected stripes is applied. The advantage of the band basis in tomographic applications is conditioned by the simplicity of solving the direct problem and the possibility of describing all types of inhomogeneities in a unified manner. The comparison of the band basis with the commonly used basis composed of nonoverlapping squares is considered. The results of the reconstruction of model kinetic and combined refractive-kinetic inhomogeneities in band basis are presented. The possibility of complete tomography reconstruction of two-dimensional flows based on the scattering data only is illustrated.