Session 1aID

Interdisciplinary: Plenary Lecture: Studying the Sea With Sound

N. Ross Chapman, Chair
School Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3065, Victoria, BC V8P 5C2, Canada

Chair's Introduction—7:55

Invited Paper

8:00

1aID1. Studying the sea with sound. Stan E. Dosso and Jan Dettmer (School of Earth & Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

Because electromagnetic radiation is strongly attenuated in seawater while sound propagates efficiently to long (even global) ranges, scientists and engineers have devised many ingenious methods to use acoustics in the ocean in place of light, radio, and microwaves. Myriad underwater acoustic applications include remote sensing, remote control, communications, navigation, and source detection/localization. This talk will present a semi-historical overview of the use of sound to study the sea (including the seabed), from philosophical musings of Aristotle, through the Renaissance, two world wars, and into the modern era of advanced measurement technologies and computer analysis. A final emphasis involves on-going research to estimate seabed geophysical properties and quantify their uncertainty and variability using a variety of ocean acoustic measurements and probabilistic inversion theory.

MONDAY MORNING, 3 JUNE 2013 513ABC, 8:55 A.M. TO 12:00 NOON

Session 1aAAa

Architectural Acoustics and Signal Processing in Acoustics: Advanced Analysis of Room Acoustics: Looking Beyond ISO 3382 I

Boaz Rafaely, Cochair
Dept. of Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva 84105, Israel

Samuel Clapp, Cochair
Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180

Michael Vorländer, Cochair
ITA, RWTH Aachen Univ., Neustr. 50, Aachen 52066, Germany

Chair's Introduction—8:55

Invited Papers

9:00

1aAAa1. An objective measure for the sensitivity of the room impulse response. Rok Prislan (Faculty of Mathematics and Phys., Univ. of Ljubljana, Jadranska 19, Ljubljana 1000, Slovenia, rok.prislan@gmail.com), Jonas Brunskog, Finn Jacobsen, and Cheol-Ho Jeong (Acoust. Technol., Dept. of Elec. Eng., Tech. Univ. of Denmark, Lyngby, Denmark)

This study is relevant for a number of important acoustic measurements in reverberation rooms such as measurement of sound transmission and measurement of sound power levels of noise sources. From a pair of impulse responses measured in a room differing only in the position of the sound source, it might be possible to quantify the sensitivity of the room due to changes in initial conditions. Such changes are linked to mixing. The proposed measure is the maximum of the absolute value of the cross-correlation between the time windowed sections of the two impulse responses. By integrating this quantity normalized by the energy of the impulse response of the room, a single number rating is obtained. The proposed measure is examined experimentally, and the results are discussed. The results indicate that the number of absorbers and diffusers in the room influences the proposed measures systematically.

9:20

1aAAa2. Room acoustic transition time based on reflection overlap. Cheol Ho Jeong, Jonas Brunskog, and Finn Jacobsen (Acoust. Technol., Tech. Univ. of Denmark, Oersteds Plads, Bldg. 352, Lyngby 2800, Denmark, chj@elektro.dtu.dk)

A transition time is defined based on the temporal overlap of reflected pulses in room impulse responses. Assuming specular reflections only, the temporal distance between adjacent reflections, which is proportional to the volume of a room, is compared with the characteristic width of a pulse at time t, which is mainly controlled by the absorption characteristics of the boundary surfaces of the room. Scattering, diffuse reflections, and diffraction, which facilitate the overlapping process, have not been taken into account. Measured impulse responses show that the transition occurs earlier in a room with nonuniform absorption and furniture than in a room that satisfies the underlying assumptions.

9:40

1aAAa3. What is "clarity," and how it can be measured? David H. Griesinger (Research, David Griesinger Acoust., 221 Mt Auburn St #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Human hearing did not evolve to detect reflections and reverberation. It is the ability to detect the direct component of a sound field that allows us to separate simultaneous signals, determine their direction, their timbre, their distance, and their importance for our attention. "Clarity" is perceived when we can separately detect direct sound. But ISO3382 measures concentrate on reflections, and are either blind to the direct component of a sound field or misinterpret its significance. They fail to predict sound quality in individual seats. We will demonstrate the vital importance of clarity through demonstrations of "clear" and "muddy" in speech and music. We will then present three physiologically based methods that measure the degree of clarity in a particular acoustic environment. The first, LOC, uses a simple nerve firing model to analyze an impulse response for the build-up of reflected energy at the onsets of sounds. The second method measures the degree of phase randomization above 1000 Hz caused by a particular impulse response. The third measure—based on a computer model of human hearing—measures clarity directly from binaurally recorded speech. All three measures predict perceived clarity with useful accuracy.

10:00

1aAAa4. Perceptual limits for detecting interaural-cue manipulations measured in reverberant settings. Stefan Klockgether (Acoust. Group, Univ. of Oldenburg, Carl-von-Ossietzky-strasse 9-11, Oldenburg 26129, Germany, steven.van.de.par@uni-oldenburg.de), Jasper van Dorp Schuitman (Philips Res. Europe, Eindhoven, Netherlands), and Steven van de Par (Acoust. Group, Univ. of Oldenburg, Oldenburg, Germany)

In this study, binaural room impulse responses (BRIRs) were manipulated to determine the just noticeable differences in the interaural time delay (ITD), interaural level difference (ILD), and interaural cross-correlation (ICC) in reverberant settings. The BRIR were split in two sections, the first 75–150 ms of the BRIR was found to be direction dependent, and for this first part either an extra ITD or ILD was applied. These manipulations were expected to change the perceived direction of the sound source. Changes in the ICC were applied to the remaining part of the BRIR, which was expected to change the overall spatial impression, but not the perceived location. Each of these three differently manipulated BRIRs was convolved with an anechoic musical instrument, and the just noticeable change in ITD, ILD, or ICC was determined in a listening experiment. Due to the convolution with a temporally varying musical instrument stimulus, a complex spectrotemporal pattern of binaural cues is created. An analysis of these cues will be presented and it will be compared to the listening test results. This analysis will be based on a model of human auditory processing, which predicts perceptual cues related to room acoustic perception.

10:20–10:40 Break

10:40

1aAAa5. Accuracy in the acoustical parameters evaluation according to ISO-3382. Miguel Arana, Ricardo San Martin, Abel Arregui, and Jorge Machin (Physics, Public Univ. of Navarre, Campus de Arrosadia, Pamplona, Navarre 31006, Spain, marana@unavarra.es)

An exhaustive characterization of the new auditorium of the Navarre Conservatory of Music (Pamplona, Spain) has been carried out. All monaural acoustic parameters in all seats (375) have been measured for three source positions on the stage. For acoustic characterization, countless results can be obtained in accordance (in all cases) with the views of the ISO-3382 for the presentation of the results. The spatial dispersion for each source position and combinations thereof will be shown. Finally, the accuracy on the acoustic evaluation of the room will be discussed from a statistical point of view.

11:00

1aAAa6. On the effects of pre-processing of impulse responses in the evaluation of acoustic parameters on room acoustics. Andrea Venturi (Univ. of Bologna, Via delle Scienze, Parma 43100, Italy, andrea.venturi78@gmail.com), Angelo Farina, and Andrea Venturi (Univ. of Parma, Parma, Italy)

The evaluation of room acoustics characteristics in rooms has been thoroughly described in several papers since 1960s. Moreover, the ISO 3382 standard describes several acoustic parameters and their measurements. However, there are only a few information about the methods of pre-processing the impulse responses that are required before calculating those acoustic parameters. In this paper, the main processing methods (based on Luneby, Chu, and Hirata methods) are analyzed. Moreover, they are compared with the Schroeder (background integrated) methods. In a further step, these methods are applied in some acoustic measurements employed in some opera houses in Italy. Finally, after a full discussion about the uncertainties that is beyond these methods, the acoustic parameters are compared with the JND that is actually accepted in the evaluation of the mono-aural, binaural, and spatial acoustic parameters.

11:20

1aAAa7. The influence of noise on monaural room acoustic parameters utilizing different evaluation methods. Martin Guski and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustraße 50, Aachen 52066, Germany, mgu@akustik.rwth-aachen.de)

Besides clarity and definition, the reverberation time is the most common room acoustic parameter. The latter is also an essential quantity for acoustic measuring techniques (i.e., sound insulation, scattering, or diffuse absorption). The unavoidable occurring noise in every measurement is the most significant factor that causes incorrect evaluated parameters. Therefore, it is important to respond to the effects caused by noise. In this study, different noise compensation methods are compared theoretically and based on measurements. At first all methods are investigated theoretically utilizing a simple parametric model impulse response. As a second step, long-term measurements have been conducted in an auditorium to analyze the performance of the different techniques under realistic conditions. Therefore, the excitation signal has been varied in volume to obtain measurements with different noise levels, and the evaluated room acoustic parameters are examined as a function of peak-signal to noise ratio. Theoretical and measured results coincide with each other for each analyzed method. The performances of the examined evaluation methods differ clearly. In particular, the three methods defined by ISO 3382 show different behaviors. The advantages and the limitations of each noise compensation method are presented.

11:40

1aAAa8. Including directivity patterns in room acoustical measurements. Martin Pollow, Johannes Klein, Pascal Dietrich, and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustraße 50, Aachen 52056, Germany, mpo@akustik.rwth-aachen.de)

Room acoustical measurements according to ISO 3382 require source and receiver to be of omnidirectional sensitivity. Therefore, radiation patterns of natural sources and receivers (although audible) are not accounted for when using the obtained room impulse responses (RIRs) for room acoustic analysis or even auralization. In order to include this spatial information in the RIR, it is necessary to measure the RIR for each pair of radiation patterns of source and receiver. This could be done by electronic beamforming during the measurement using array systems or by mechanical modification of the transducer (as, e.g., a dummy-head with its corpus). In this contribution, an alternative approach is shown, using the superposition of a set of sequential measurements done with a spherical sound source. At the cost of longer measurement times, the obtained data can be used universally to synthesize RIRs of arbitrary directivity up to a certain maximal spatial resolution, as long as the room is considered as a linear and time-invariant system during the measurement. The measurement device, obtained results, and a study of the validity of the superposition approach are presented in this talk. Based on this representation of the RIR, more advanced spatial room acoustic analysis accounting for arbitrary sets of source and receiver directivity becomes possible.

MONDAY MORNING, 3 JUNE 2013 — 513DEF, 9:00 A.M. TO 12:00 NOON

Session 1aAAb

Architectural Acoustics and Noise: Cultivating the Sustainable in Architectural Acoustics

Jesse Ehnert, Chair
Arpeggio, LLC, 1060 Mercer St., SE, Atlanta, GA 30316

Invited Papers

9:00

1aAAb1. Achieving the acoustical credit within the Leadership in Energy and Environmental Design (LEED®) for healthcare green building rating system and changes within the forthcoming 2014 Guidelines. Daniel M. Horan (Cavanaugh Tocci Assoc., Inc., 327 F Boston Post Rd., Sudbury, MA 01776, dhoran@cavtocci.com) and Jean-François Latour (Acoust. and Vibr., SNC-Lavalin Inc., Longueuil, QC, Canada)

The 2009 LEED® for Healthcare green building rating system includes a total of two possible points to be earned by satisfying the requirements of the Indoor Environmental Quality Credit 2 (IEQ Credit 2: Acoustic Environment). This credit references criteria that are defined by the Facility Guidelines Institute's 2010 *Guidelines for Design and Construction of Health Care Facilities* (2010 FGI Guidelines). The acoustical design requirements of the 2010 FGI Guidelines will be summarized, as well as a brief history of the FGI document itself. The FGI Guidelines document is currently being revised for the 2014 edition. Proposed changes in the 2014 edition will also be briefly discussed, as we anticipate that future versions of the LEED® rating system will reference the 2014 FGI Guidelines. The presentation will be made in English by Mr. Horan, secretary of the FGI's Acoustical Working Group (AWG). Mr. Latour, a French-Canadian member of the AWG, will be available to help respond to any French-speaking audience members during Q&A.

9:20

1aAAb2. Achieving Leadership in Energy and Environmental Design acoustical requirements in a commercial office project. Ethan Salter (Charles M. Salter Assoc., Inc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, ethan.salter@cmsalter.com)

This paper will discuss a recently completed a Leadership in Energy and Environmental Design Commercial Interiors (LEED-CI) office renovation project in San Francisco, California. The project had multiple aims: office space, test laboratory for new materials and technologies, and teaching tool for clients and students. The owner, an acoustical consulting firm, decided on LEED-CI certification because of the belief that acoustic comfort could be achieved while simultaneously meeting other LEED requirements (e.g., design, products and materials, construction methods, and operations). Recently, acoustical requirements have been adopted into various LEED rating systems because occupant acoustic comfort in many LEED-certified buildings has been poor. The organization responsible for LEED, the US Green Building Council, is taking steps to more comprehensively adopt acoustical standards throughout their portfolio. By using the LEED Innovation in Design (ID) Pilot Credit Library, projects can attempt to achieve a wider range of potential credits. One of those credits, Pilot Credit 24, addresses acoustic comfort, including sound isolation, speech privacy, background noise, and reverberation time. The project is one of the first to achieve Pilot Credit 24 requirements. This paper will discuss the project design objectives, Pilot Credit 24 requirements, and how the project achieved those requirements.

Contributed Papers

9:40

1aAAb3. Straw bale sound insulation: Blowing away the chaff. Stephen Dance and Paul Herwin (Urban Eng., London South Bank Univ., Borough Rd., London SE1 0AA, United Kingdom, dances@lsbu.ac.uk)

Popular opinion states that straw bale walls are good at isolating sound. Cheap load bearing straw bale houses could contribute substantially to low carbon sustainable construction. However, literature on the subject was found to be highly anecdotal. The paper presents a summary of nine laboratory and field sound insulation test reports and two especially commissioned tests. Data were compared to European party wall sound insulation criteria, and it was found that straw walls could perform as well as, but sometimes worse than, conventional constructions, due to poor performance at low frequencies. Better performance could help to promote the use of straw bales in multi-unit housing. It was found that by adding a plasterboard layer on studs to just one side of a plastered straw bale wall would allow the construction to pass all of the criteria reviewed.

10:00–10:20 Break

10:20

1aAAb4. From felt to fungus: New materials and applications—Focus on sustainability. Dawn Schuette and Scott Pfeiffer (Threshold Acoust. LLC, 53 W Jackson Blvd, Ste. 815, Chicago, IL 60604, spfeiffer@thresholdacoustics.com)

A two-part presentation of new materials for use in architectural acoustics. This presentation emphasizes new materials or new applications of standard products that provide acoustic benefit in a highly sustainable context. The companion session is presented in "New Materials for Architectural Acoustics." Current trends in architecture are bringing more organic approaches to the use of natural materials. Exploiting these trends with approaches that have definable acoustic behavior leads to more flexibility in architectural design and yields acoustical application of materials that are not traditionally part of the acoustical treatment vocabulary. Case studies will be presented featuring new materials and/or methods being employed for sustainable acoustic solutions.

10:40

1aAAb5. Development of an ecological, smooth, unperforated sound absorptive material. Seda Karabulut (R&D, MEZZO Studyo LTD., METU,R&D, MEZZO Studyo LTD, Ankara, Turkey, sedakarabulutt@gmail.com) and Mehmet Çalışkan (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

Material selection for acoustically comfortable environments is a very important issue especially for rooms for speech and music as well as for large volumes like shopping centers and foyers. Energy efficient and sustainable materials are devised in construction industry for healthy environments; hence, ecological sound absorbing materials for acoustically sensitive environments are being preferred to get credits for international certification procedures like LEED and BREAM. Nevertheless, most of the acoustic materials in construction industry are perforated with mineral wool based absorption materials behind and have great effects on design of the building environments. Architects usually prefer seamless unperforated materials to avoid changes in the appearance of design environments for acoustical requirements. This article is about development of an ecological unperforated acoustical material, which is made of reed and pumice stone. Different layers of pumice stone and reed glued with an ecological binder are evaluated according to frequency range they are effective, and an optimization is done to create an acoustic material that is effective especially in medium frequencies. Acoustical performance of the material is justified with measurements of sound absorption coefficient in Kundt Tube.

11:00

1aAAb6. Design, optimization, and testing of door-grille silencers. Vivek V. Shankar and Murray Hodgson (Acoust. & Noise Res. Group, Univ. of British Columbia, 2054-6250 Appl. Sci. L., Vancouver, BC V6T1Z4, Canada, vivekvs1@gmail.com)

It is a common practice to install doors that have openings in them to improve cross airflow through horizontal ventilation. However, excessive outdoor noise and poor noise privacy are known associated issues. Grilles are often installed in these door openings to address this issue. While they may reduce the noise level slightly, they have proven not to be very effective. Effective silencers would be too thick to be installed in doors. This work investigates the design and development of a novel door silencer that reduces the sound transmission to acceptable limits without compromising the airflow. A model of the silencer has been designed and tested—using the Acoustics module of the COMSOL Finite Element software—in a diffuse field environment, and validated with STC ratings. The airflow was modeled using the COMSOL CFD module. The dimensions of the ventilation opening and its position in the door have also been optimized. A real prototype of the model has then been built and its performance tested. Various design guidelines have then been proposed for the design of these doors.

11:20

1aAAb7. The furniture industry needs a new evaluation standard for evaluation of sound absorption. Klas G. Hagberg (Acoust., WSP, Box 13033, Goteborg 41526, Sweden, klas.hagberg@wspgroup.se) and Delphine Bard (Acoust., dBA R&D, Neuchatel, Switzerland)

Since decades the standard ISO 11654 are prevailing for evaluating sound absorption of products. The standard is developed and fully adapted to ceiling manufacturer, in particular mineral wool ceiling manufacturer. However, the standard is used independently of which interior product it is applied to, casing a lot of "misuse" and confusion amongst many manufacturer. In particular, when it comes to evaluation of various types of office screens, the ISO 11654 becomes a problem. There is no need, and probably not even possible, to calculate absorption factor for an office screen correctly. Therefore, Sweden decided to develop a new standard, SS 25269—"Acoustics—Evaluation of sound absorption of single objects." Hence office screens should be treated as single objects to cover a wide range of variety. The standard specifies an evaluation method of the sound absorption using only sound absorption area for each object tested. It will simplify the evaluation and minimize risk for errors since there is no need to state the product area in order calculate absorption coefficient. Furthermore, the area does not have to be stated yet again when performing calculation of room characteristics when using the same products in the finished room.

11:40

1aAAb8. Audio and acoustic design of the University of Sydney's Indoor Environmental Quality Laboratory. Densil Cabrera, Robert Crow, Luis Miranda, and Richard de Dear (Faculty of Architecture, Design & Planning, The Univ. of Sydney, G04, Sydney, NSW 2006, Australia, densil.cabrera@sydney.edu.au)

The quality of indoor environments such as commercial offices is affected by many factors, including temperature, humidity, air movement,

illumination, ambient sound, and room acoustics. In 2012, a new laboratory was established at the University of Sydney to examine how such factors affect human occupants. In terms of sound, the design of the laboratory has three components: the acoustic design of the testing rooms; the audio system design (for introducing artificial soundscapes); and the design of generic soundscapes to support experimental work in the laboratory. Acoustic design considerations of the laboratory allow for the testing rooms to be configured as high grade office environments. The laboratory has a 24-channel audio system for introducing realistic and potentially complex sound fields in to the testing rooms, both from within and outside the rooms. Parametrically controlled soundscapes have been developed for interior sources (such as building services noise) and exterior sources (such as transport noise). This paper describes how the combination of the laboratory's acoustics, audio systems, and soundscapes can be used for scientific studies of indoor environmental quality.

MONDAY MORNING, 3 JUNE 2013 — 510D, 8:55 A.M. TO 12:00 NOON

Session 1aAO

Acoustical Oceanography: Estuarine Acoustics

Andone C. Lavery, Cochair
Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536

David R. Barclay, Cochair
Memorial Univ. of Newfoundland, P.O. Box 4200, St John's, NF A1C 5S7, Canada

Chair's Introduction—8:55

Invited Paper

9:00

1aAO1. The impact of acoustic oceanographic methods on estuarine dynamics research. W. Rockwell Geyer, Peter Traykovski, and Andone Lavery (Woods Hole Oceanogr. Inst., 98 Water St., MS-12, Woods Hole, MA 02543, rgeyer@whoi.edu)

Estuaries present unique challenges for observational oceanographers, due to their intense spatial gradients and unrelenting temporal variability. The influence of spatial and temporal variation of estuarine structure and flow on the time-averaged regime is the most important research problem in estuarine physical oceanography. Acoustic methods have played an essential role in revealing this spatial and temporal variability, and new advances in acoustic methods are continuing to provide the most important advances in observations of estuarine processes. The measurement of acoustic backscatter has been a mainstay of estuarine physical oceanography, first for providing qualitative images of the density structure, then for quantifying suspended sediment distributions, and most recently for quantifying the intensity of stratified turbulence. Improved resolution of new systems is revealing the internal structure of shear instability and the mechanics of the transition to turbulence. Acoustic Doppler techniques are so routine now as to be taken for granted, but their impact on the field cannot be overstated, and the new advances in pulse-coherent velocity profiling are continuing this revolution in acoustical oceanography. Acoustic propagation in estuaries has not yet received much attention, but its importance to the operation of autonomous vehicles and long-term monitoring should bring this challenging acoustics problem to the forefront.

Contributed Papers

9:20

1aAO2. The spatial properties of breaking wave generated and bedload transport generated noise in the sediment layer of a shallow water wave guide. David R. Barclay, Len Zedel (Phys. and Physical Oceanogr., Memorial Univ. of Newfoundland, P.O. Box 4200, St John's, NF A1C 5S7, Canada, dbarcl@gmail.com), Alex E. Hay, and Matthew G. Hatcher (Oceanogr., Dalhousie Univ., Halifax, NB, Canada)

In May of 2012, three weeks of ambient noise measurements from a hydrophone buried 30 cm deep in the sediment were recorded at Advocate Beach, a 1:10 sloped beach at the head of the Bay of Fundy, Nova Scotia. While tides varied the mean water depth between 0 and 4 m, 0.8 m surface waves passed overhead, driving sediment bedload transport and creating an ambient noise field in the sediment consisting of two primary components: noise generated by bubbles formed in breaking waves at the surface and noise generated by the collisions of sand, gravel, and cobble in the bedload transport along the seabed. Both of these noise sources are stochastic and can be described by their second order statistics: power spectral density, spatial coherence, and directional density. In an effort to distinguish these two noise sources, the spatial properties of three full wave models of the noise field in the sediment are compared, using an infinite sheet of sources placed near the surface of a Pekeris waveguide to model breaking wave noise, near the fluid–fluid interface of a Pekeris waveguide to model bedload transport noise, or near the fluid–fluid interface of two infinite half-spaces to model bedload transport noise. Using integral transforms to solve the wave equation, each noise model is shown to be spatially inhomogeneous with a unique depth dependent intensity and coherence.

9:40

1aAO3. Acoustic measurements of the spatial distribution of suspended sediment at three sites on the Lower Mekong River. Stephanie A. Moore (Civil Eng., Univ. of Ottawa, 161 Louis Pasteur St, Ottawa, ON K1N6N5, Canada, moore@uottawa.ca), Guillaume Dramais (UR HHLY Hydrology Hydraulics, Irstea Lyon, Lyon, France), Philippe Dussouillez (Ctr. Européen de Recherche et d'Enseignement des Géosciences de l'Environnement, Aix-en-Provence, France), Jerome Le Coz (UR HHLY Hydrology Hydraulics, Irstea Lyon, Lyon, France), Colin Rennie (Civil Eng., Univ. of Ottawa, Ottawa, ON, Canada), and Benoit Camenen (UR HHLY Hydrology Hydraulics, Irstea Lyon, Lyon, France)

The Mekong River spans thousands of kilometers, flows through six countries, and its basin is one of the world's richest in terms of biodiversity.

However, land-use changes, dredging of the river bed, and the construction of dams are changing its sediment dynamics and morphology. The resultant increases in bank erosion and reduction in sediment supply to floodplains may have adverse effects on the economical and biological productivity of the region. In order to monitor these changes, the current conditions must be well understood. Comprehensive measurements of the spatial distribution of sediment (both suspended and bed load) were made at three locations in different physiographic regions of the Lower Mekong at the end of the 2012 rainy season. Acoustic Doppler Current Profilers and a multifrequency acoustic backscatter system, the AQUAscat, were used in combination with water sampling to provide high resolution measurements of concentration and grain size. The AQUAscat consisted of four monostatic transducers operating at 0.5, 1, 2.5, and 4 MHz. At each site, it was deployed horizontally at five across-stream positions and 5–10 depths per vertical; a 10 m profile was recorded at each point. The spatial distribution of particle size and concentration were determined using multifrequency inversions of (1) backscattered intensity and (2) attenuation calculated from the intensity profiles. This data set provides a baseline to which to compare future measurements at these sites.

10:00–10:20 Break

10:20

1aAO4. Acoustic propagation characteristics of the estuarine salt wedge. D. Benjamin Reeder (Oceanography, Naval Postgrad. School, 73 Hanapepe Loop, Honolulu, Hawaii 96825, dbreeder@nps.edu)

The estuarine environment often hosts a salt wedge—a layer of denser seawater advected by the rising tide under fresh water discharged by the river. The nature of the stratification is a function of the tide's range and speed of advance, river discharge volumetric flow rate, and river mouth morphology. The competing effects of temperature and salinity on sound speed present the question: Is the salt wedge acoustically observable? Using temperature and salinity profiles collected *in situ*, numerical results show that the salt wedge can impact acoustic propagation. Acoustically, this environment consists of two isospeed layers separated by a thin gradient. While this three-layer very shallow water acoustic waveguide is typically dominated by high angle multipath propagation, refraction occurring in the gradient layer allows some low-angle energy from near-surface sources to be trapped in the upper layer. Acoustic fluctuations observed at an upstream or downstream receiver depend upon the interaction between the advancing and receding tide and the river discharge, which can include the presence of internal waves propagating along the top of the salt wedge.

10:40

1aAO5. Quantification of stratified turbulence using acoustic propagation and broadband scattering techniques. Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., MS 11 Bigelow 211, 98 Water St., Woods Hole, MA 02543, alavery@whoi.edu), Wayne R. Geyer (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), and Malcolm E. Scully (Physical Oceanogr., Old Dominion Univ., Norfolk, VA)

Narrowband acoustical backscattering techniques have been used for decades as a tool for remote imaging of small-scale physical processes in energetic coastal environments, such as oceanic internal waves and microstructure, on spatial and temporal scales difficult to probe with in situ measurements. However, to date, it has been challenging to infer quantitative information about turbulent intensity from the measured backscatter, in part due to uncertainty in the sources of scattering. In contrast to narrowband techniques, emerging broadband techniques result in increased spectral classification and quantification capabilities. Broadband backscattering collected in the Connecticut River Estuary in 2009 in concert with in situ measurements of turbulence have illustrated the potential of these techniques for quantitative remote-sensing of microstructure intensity over relevant spatial and temporal scales. These measurements have resulted in remote quantification of finescale variability of turbulent mixing as well as examination of the mechanisms and structure of shear instability across a broad range of stratification and shear conditions. Upcoming acoustic propagation measurements directed at measuring acoustic scintillation in the Connecticut River Estuary in the presence of strongly stratified turbulence and shear instabilities, which allow remote-sensing of the path-averaged statistical structure and motion of the intervening flow, will also be discussed.

Invited Paper

11:00

1aAO6. Acoustics and estuarine ecology: Using active and passive methods to survey the physical environment, vegetation, and animals in North Carolina's coastal estuaries. Joseph J. Luczkovich (Inst. for Coastal Sci. and Policy, East Carolina Univ., M. S. 169, Greenville, NC 27858, luczkovichj@ecu.edu), Mark W. Sprague (Physics, East Carolina Univ., Greenville, NC), Cecilia S. Krahforst (Coastal Resources Management, East Carolina Univ., Greenville, NC), John P. Walsh (Geological Sci., East Carolina Univ., kitty Hawk, NC), Audrey J. Pleva (Biology, East Carolina Univ., Greenville, NC), and Dean E. Carpenter (Albemarle-Pamlico National Estuary Partnership, Raleigh, NC)

Estuarine systems have complex interactions of physical and biological processes. Regular observations are needed in order to understand their dynamics. Acoustic observation systems (echosounders, acoustic Doppler current profilers (ADCPs), and passive acoustic dataloggers) can provide observations on a wide spectrum of processes in estuaries. We have used echosounders to monitor changes in bathymetry, submerged aquatic vegetation, fishes, and invertebrates over time. In addition, sediment changes, resuspension events, turbidity, and waves are monitored using ADCPs. The higher trophic level species of fishes and marine mammals that are soniferous have been monitored by passive acoustic methods. We provide examples of each acoustic method used to study the dynamics of seagrasses, fishes, and the physical environment of the Albemarle, Pamlico, Currituck, and Core Sounds in North Carolina. While it is possible to combine methods to use acoustics to measure the dynamics of estuarine systems (estuarine observing systems), the challenge we face is to ground-truth these acoustic metrics using traditional sampling methods (e.g., quadrats for plants, trawls for fishes, and water samples for sediments) and integrate each of these measures. We could then examine the effect of storms, waves, and resuspension events on estuarine plant and animal distributions and abundances using acoustics metrics.

Contributed Paper

11:20

1aAO7. Investigation of low-frequency acoustic tissue properties of seagrass. Gregory R. Enenstein, Craig N. Dolder, Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 4700 W Guadalupe St #A-437, Austin, TX 78751, gregenenstein@gmail.com), and Jean-Pierre Hermand (Acoust. & Env. Hydroacoustics Lab, Université Libre de Bruxelles (ULB), Brussels, Belgium)

Understanding the acoustic properties of seagrass is important for applications in mine hunting, shallow water sonar performance, and acoustic remote sensing for ecological surveys. Previous laboratory and field investigations have shown that the plant biomass and tissue structure of seagrass, rather than just the overall gas content, play a determinant role in its acoustic behavior. Hence, effective medium models of propagation through seagrass meadows have been ruled out, and a complete description of both tissue structure and tissue elastic properties is required to describe the acoustic response of seagrass meadows. To begin to address these deficiencies, a resonance tube experiment was set up to determine the low-frequency acoustic response of multiple species of seagrass in relation to leaf biomass and tissue acoustic compliance independent of tissue structure. Responses to frequency-modulated signals in the range from 0.5 to 10 kHz were obtained for *Thalassia testudinum* (turtle grass) and *Halodule wrightii* (shoal grass), two species with well-differentiated morphological features. An elastic waveguide model was used to account for the minor effect of the tube walls on the resonance characteristics. Initial measurements of tissue compliance will be presented. [Work supported by ONR and ARL:UT.]

11:40–12:00 Panel Discussion

MONDAY MORNING, 3 JUNE 2013 — 519A, 9:00 A.M. TO 11:40 A.M.

Session 1aBA

Biomedical Acoustics: Ultrasound Tomography

Yun Jing, Chair

Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695

Invited Papers

9:00

1aBA1. Three-dimensional nonlinear inverse scattering: Quantitative transmission algorithms, refraction corrected reflection, scanner design, and clinical results. James Wiskin (Bioengineering, Univ. of Utah, 3216 Highland Dr., Ste. 100, Salt Lake City, UT 84106, jwiskin.cvus@gmail.com), David Borup (CVUS, LLC., Salt Lake City, UT), Michael Andre (Radiology, Univ. of California VA Medical Ctr., San Diego, CA), Steven Johnson (CVUS, LLC., Salt Lake City, UT), James Greenleaf (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN), Yuri Parisky (Radiology, Mammoth Hospital, Mammoth Lakes, CA), and John Klock (CVUS, LLC., Salt Lake City, UT)

Research in quantitative whole breast ultrasound imaging has been developing rapidly. Recently, we published results from 2D transmission inverse scattering algorithms, based on optimization, incorporating diffraction, refraction, and limited multiple scattering effects, using data collected from an early prototype, which showed the feasibility of high resolution quantitative imaging of the breast tissue speed and attenuation, and concomitant refraction corrected reflection imaging. However, artifact problems in speed and attenuation result from the 2D algorithms and the data characteristics. The reflection algorithm uses the speed map to model refractive effects of rays, so these artifacts are unacceptable. The 3D inverse scattering algorithm presented here, using data from a new prototype, overcomes most of these artifacts. We then use a 3D refraction corrected 360° compounded reflection algorithm for high resolution speckle free reflection images. We discuss the transmission and reflection algorithms and the advanced scanner used to collect the data, as well as initial clinical results from the Mayo Clinic, Breast Cancer Imaging Center, Orange County, and the University California, San Diego. We show examples of cysts, fibroadenomas, calcifications, cancers, and DCIS, in dense, fatty, and average breast tissue, and compare these with hand-held ultrasound, MRI, and mammography, where available.

9:20

1aBA2. Quantitative ultrasound tomography. Koen W.A. v. Dongen and Neslihan Ozmen-Eryilmaz (Lab. of Acoust. Wavefield Imaging, Delft Univ. of Technol., P.O. Box 5046, Delft 2600 GA, Netherlands, k.w.a.vandongen@tudelft.nl)

Whole breast ultrasound is gaining interest as a possible alternative to mammography, as it is cost effective, patient friendly, and avoids the usage of ionizing radiation. Due to the similarity in both measurement setups, scientists have investigated if tomographic reconstruction algorithms originally developed for x-ray tomography, such as inverse radon transforms, are also applicable to ultrasound tomography. However, the multiple scattering of an acoustic wave field inside the breast as well as diffraction and refraction effects results in a severe blurring of the obtained images. In order to overcome these limitations, people are developing imaging algorithms, which are based on the acoustic wave equations. To show the limitations and potentials of the various imaging algorithms, we computed synthetic data for a cancerous breast model using a full wave solution method. Next, we tested and compared different imaging algorithms varying from a ray based inverse radon transform up to a full-wave nonlinear inversion technique. The latter one has the advantage that, as we will show, it allows for accurate speed of sound profile reconstructions at the cost of a severe computational load.

9:40

1aBA3. A contrast source inversion method for breast cancer detection.
N. Ozmen-Eryilmaz and K. van Dongen (Lab. of Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628 CJ, Netherlands, n.ozmen-eryilmaz@tudelft.nl)

Tomographic ultrasound imaging is gaining popularity in breast cancer detection. Reconstructing the acoustic properties of a breast from the ultrasound measurements is stated as a nonlinear inverse problem, which is usually solved by linearized methods because of computational efficiency. However, linearization of the problem reduces the quality of the reconstruction. To improve the accuracy, we developed and tested a three-dimensional nonlinear inversion method that allows for three-dimensional reconstruction of the breast in terms of speed of sound. The method, referred to as contrast source inversion (CSI), uses an integral equation formulation to describe the inverse acoustic scattering problem. The resulting integral equation is solved to reconstruct the unknown contrast (speed-of-sound profile of the breast). The contrast and contrast sources (the product of the contrast with the total field) are iteratively updated by minimizing a cost functional using conjugate gradient directions. In this study, we tested the CSI method on synthetic data retrieved form full-wave simulations for a realistic three-dimensional cancerous breast model. Results show that the CSI method outer performs other conventional methods as it yields speed-of-sound reconstructions that are akin to the model. This shows that the approach offers a contribution to the detection of breast cancer.

10:00

1aBA4. Approximation error method for full-wave tomography.
Janne Koponen, Tomi Huttunen, Tanja Tarvainen (Appl. Phy., Univ. of Eastern Finland, Yliopistonranta 1, PL 1627, Kuopio 70211, Finland, janne.koponen@uef.fi), and Jari Kaipio (Mathematics, Univ. of Auckland, Auckland, Finland)

In ultrasound tomography (UT), the speed of sound (SOS) is reconstructed based on ultrasound measurements made on the surface of the object. As a part of the reconstruction process, which includes the solution of the inverse problem, propagation of acoustic signals in the medium is simulated using a forward model. Consequently, modeling errors can generate artifacts into reconstructed SOS. Accurate full-wave models can be computationally heavy and thus impractical in many real applications. On the other hand, approximate models typically lead to less accurate reconstructions. In this study, measurement noise and modeling errors of UT are modeled in Bayesian framework, and a numerical method that takes both errors into account is developed. The performance of the method is investigated by numerical simulations in which artifacts generated by a fast but less accurate forward model and approximate boundary conditions are compensated.

10:20

1aBA5. Flawed transducer detection using random sample consensus for ultrasound tomography.
Tianren Wang and Yun Jing (Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., Eng. Bldg. 3, 3141, Raleigh, NC 27695, twang10@ncsu.edu)

In this paper, we present a random sample consensus (RANSAC) based ultrasound travel-time tomography method. Conventionally, all the time-of-flight (TOF) data between each two transducers are used to estimate the sound speed distribution. However, failing to identify the inaccurate TOF data (outliers) due to flawed transducers would reduce the accuracy of the estimated sound speed distribution. In our proposed approach, a small subset of TOF data were first randomly selected from the original TOF data, and then applied to the tomography algorithm to estimate a rough sound speed distribution. The rest of the TOFs data was applied to the rough distribution, and the goodness of fit was calculated. If most of the data fitted well in the estimated distribution, then all the well-fitted data (including the subset) was used to estimate a final sound speed distribution. Otherwise, there were outliers expected in the subset, and a new subset of the TOFs data would be randomly selected again. This repeated until most of the data fitted well in the estimated distribution. Simulation results showed that our method could effectively detect and eliminate outliers and increase the accuracy of estimating sound speed distribution.

10:40

1aBA6. Expressiveness of temperature-induced changes in backscattered energy in conventional B-mode images.
Cesar A. Teixeira (CISUC, Univ. of Coimbra, Polo II, Pinhal de Marrocos, Coimbra 3030-290, Portugal, cteixei@dei.uc.pt), Marco von Kruguer (Biomedical Eng. Program, COPPE—Federal Univ. of Rio de Janeiro, Rio de Janeiro, Brazil), André V. Alvarenga (Lab. of Ultrasound-Directory of Scientific and Industrial Metrol. (Dimci), INMETRO, Rio de Janeiro, Brazil), and Wagner C. Pereira (Biomedical Eng. Program, COPPE—Federal Univ. of Rio de Janeiro, Rio de Janeiro, Brazil)

Changes on conventional B-mode images have been correlated with temperature, aiming to develop a reliable method for noninvasive temperature estimation. The assumption is that temperature variations induce wave propagation changes that modify the backscattered ultrasound signal and these changes have an expression in ultrasonographic images. One of the main effects is the change on the image intensity that is mainly caused by temperature-related changes in backscattered energy (CBE) from tissue inhomogeneities. It is reported that CBE is dependent on medium speed-of-sound and density, behaving in different ways for lipid or aqueous scatterers. In this paper, we demonstrate that CBE has an expression on B-mode images recorded from conventional ultrasound scanners. We observed that different regions have positive, negative, or undefined correlations with temperature, and that this behavior is due to the dependence of CBE with scatterers type. This differentiated behavior enables the segmentation of different structures inside the same tissue. Our experimental setup consisted in the temperature elevation from 36 to 44 °C (hyperthermia range) of *ex-vivo* tissue samples. We considered bovine muscle and porcine muscle and fat. For both samples, we observed coherent segmentations of the different structures, pointing for a potential clinical application of the proposed analysis.

11:00

1aBA7. Electromagnetic hydrophone for high-intensity focused ultrasound measurement.
Pol Grasland-Mongrain (Université de Lyon, 151 Cours Albert Thomas, Lyon 69424, France, pol.grasland-mongrain@inserm.fr), Jean-Martial Mari, Bruno Gilles, and Cyril Lafon (LabTAU, INSERM U1032, Lyon, France)

An ultrasonic hydrophone based on the Lorentz force is introduced. When a metallic wire is moved by ultrasound while submitted to a magnetic field, the Lorentz force induces an electrical current proportional to the integral of pressure along the wire. 2D pressure field mapping is achieved by performing a tomography through wire translations and rotations in the imaging plane. Performances of this hydrophone are assessed in this study. Signal is linear over pressure from 10 kPa to at least 10 MPa with a determination coefficient R^2 above 0.997. Excellent resistance to cavitation has been observed. Frequency bandwidth was measured against three different wire diameters: 70 μm, 100 μm, and 210 μm. Results showed that upper cut-off frequency decreases with increasing wire diameter. Additional measurements showed that wire tension has no visible effect on the signal. Such characteristics are potentially of great interest for high-intensity focused ultrasound and shockwave transducers calibration.

11:20

1aBA8. Ultrasonic projection imaging of biological media.
Krzysztof J. Opielinski and Tadeusz Gudra (Electron., Wroclaw Univ. of Technol., Wybrzeze Wyspianskiego 27, Wroclaw, Low Silesia 50-370, Poland, krzysztof.opielinski@pwr.wroc.pl)

The study presents the method of ultrasonic projection imaging of biological media, using single ultrasonic probes and 2-D piezoelectric transducer arrays. Dedicated research stands were set up and used to perform ultrasonic projection scanning of various biological media (and phantoms of the media) that were submerged in water. Based on such measurements, images of the heterogeneous internal structure of the studied objects were

obtained, which show two-dimensional distributions of the projection values of acoustic parameters. Those parameters were derived from recorded pulses of ultrasonic wave transmitted sequentially through a fixed projection surface. The obtained projection images were analyzed with respect to the method and quality of representation of the studied structures. Additionally, contrast resolution of ultrasonic projection images of the heterogeneous structure of a biological medium was estimated in relation to the size of the heterogeneity and with respect to scanning resolution and longitudinal resolution. Ultrasonic projection imaging can be applied in medicine for diagnostic examination of women's breast.

MONDAY MORNING, 3 JUNE 2013 512AE, 9:00 A.M. TO 12:00 NOON

Session 1aEA

Engineering Acoustics: Thermoacoustics I

Roger T. Richards, Chair
NUWC, Newport, RI 02841

Contributed Papers

9:00

1aEA1. Heat transfer enhancement through thermoacoustically driven streaming. Randall A. Ali and Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Grad. Prog. in Acoust., P.O. Box 30, State College, PA 16804-0030, randallali@gmail.com)

We recently reported on a simple standing-wave thermoacoustic engine that was intended for use as a self-powered monitor of temperature within a resonator that was similar to a nuclear fuel rod [J. Acoust. Soc. Am. **132**(3), Pt. 2, 1993–1994 (2012)]. An additional potential benefit of such a device is the enhanced heat transfer between the heat source and the surrounding coolant produced by the acoustic streaming generated by the high-amplitude acoustic standing wave within the resonator. By adding a remotely operated linear actuator that can depress a valve at the ambient-temperature end of the resonator, we are able to squelch the acoustic resonance by modification of that boundary condition without changing any other operating parameters (e.g., heater power). We will report heat transfer measurements made in a calorimeter at several input thermal power levels, with and without the presence of the thermoacoustic oscillations. [Work supported by the U.S. Department of Energy.]

9:20

1aEA2. Finite element simulation of a two-dimensional standing wave thermoacoustic engine. Jan A. de Jong, Ysbrand H. Wijnant, and André de Boer (Eng. Technol., Univ. of Twente, P.O. Box 217, Enschede, Overijssel 7500AE, Netherlands, j.a.dejong@utwente.nl)

A finite element analysis has been performed on a theoretical two-dimensional standing wave thermoacoustic engine using the linearized thermoacoustic equations in the frequency domain. This analysis is used to obtain the stability curve of the thermoacoustic engine, which in turn is used to calculate the oscillation onset temperature difference across the stack. The results are compared with existing theory including the long-pore approximation, as originally derived by Rott *et al*. In addition, the time-averaged effects of the acoustic wave are obtained using weakly nonlinear thermoacoustic theory. This includes the second order time-averaged equations for energy, momentum, and continuity. The saturation amplitude of the acoustic pressure oscillation and the required heat input to sustain the oscillation is obtained. The theory allows for calculation of acoustic streaming patterns. The particle path calculations provide insight to the minor loss mechanism occurring at the interface between the stack and the free tube for low acoustic velocity amplitudes (laminar flow).

9:40

1aEA3. Calculation of thermoacoustic functions with computational fluid dynamics. Simon Bühler (Thermal Eng., Univ. of Twente, P.O. Box 217, Enschede 7500AE, Netherlands, s.buhler@utwente.nl), Douglas Wilcox (Chart Inc., Troy, NY), Joris P. Oosterhuis, and Theo H. van der Meer (Thermal Eng., Univ. of Twente, Enschede, Netherlands)

Thermoacoustic functions are important parameters of one-dimensional codes used for the design of thermoacoustic engines. The thermal and viscous thermoacoustic functions allow the inclusion of three dimensional effects in one-dimensional codes. These functions are especially important in the regenerator of a thermoacoustic engine, where the thermoacoustic heat pumping occurs. Even though analytical solutions were derived for uniform pores, the thermoacoustic functions for complex geometries such as stacked screen or random fiber regenerators cannot be calculated analytically. In order to gain more insight into the geometry induced complex flow fields, the procedure of Udea *et al*. (2009) to estimate the thermoacoustic functions was applied in computational fluid-dynamic simulations. By using two measurement locations outside of the regenerator and modeling the regenerator as an array of uniform pores, it is possible to estimate the thermoacoustic functions for complex geometries. Furthermore, a correction method is proposed to quantify the entrance effects at the beginning and end of a regular pore. The simulations are first validated for a uniform cylindrical pore with the help of the analytical solution. Then the correction method is successfully applied to a cylindrical pore with the results closely matching the analytical solution. Finally, the method is applied to the model of a staggered, stacked screen regenerator.

10:00

1aEA4. Acoustic characteristics of a flexible sound generator based on thermoacoustic effect. Takehiro Sugimoto and Yoshiki Nakajima (NHK Sci. & Technol. Res. Lab., 1-10-11 Kinuta, Setagaya-ku, Tokyo 1578510, Japan, sugimoto.t-fg@nhk.or.jp)

A flexible sound generator based on the thermoacoustic effect was proposed for use with thin and flexible devices. The sound generator was composed of three thin films made of aluminum, polyimide, and graphite. The aluminum functions as an electrode for heat radiation, the polyimide as a heat insulator, and the graphite as a heat sink. Thickness of each layer is 50 nm, 75 μm, and 40 μm, respectively. The area of the electrode is 100 mm $\times$ 4 mm. The proposed sound generator was modeled considering several boundary conditions and using the heat conduction equation. Then, radiated

sound was analytically described as a function of the input signal's frequency. Experimental measurement was carried out and the frequency response calculated by the model agreed with the measurement result. An experimental study was conducted on the relationship between the fundamental response and the harmonic distortion. Surface vibration was observed with the laser Doppler velocimeter. The observation revealed that the proposed device is a vibration-free sound generator. Detailed comparison between the calculation and the measurement will be discussed in the presentation.

10:20–10:40 Break

10:40

1aEA5. Study on thermoacoustic system to drive by low temperature—Effects of loop-tube thermoacoustic system connected with parallel double stacks on the onset temperature ratio. Yosuke Nakano (Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe, Kyoto 610-0321, Japan, dum0331@mail4.doshisha.ac.jp), Shin-ichi Sakamoto (Univ. of Shiga Professor, Hikone, Shiga, Japan), and Yoshiaki Watanabe (Doshisha Univ., Kyotanabe, Kyoto, Japan)

As the temperature ratio of the both ends of the stack increases gradually and reaches a critical value, sound waves begin to oscillate in the thermoacoustic system. This temperature ratio is called the onset temperature ratio. It is necessary to decrease the onset temperature ratio for practical use of the thermoacoustic system: the use of the factory exhaust heat and the solar heat. In a previous study, thermoacoustic system with series connected a number of prime movers was designed. This system can drive by lower onset temperature ratio than thermoacoustic system with a prime mover. However, it is considered that the heat loss increase when the heat is carried to a number of high heat exchangers. Therefore, loop-tube thermoacoustic system connected with parallel double stacks (parallel loop system) was proposed. This system can drive two prime movers by a heat input part because it is connected prime movers in parallel. In this report, the onset temperature ratio of this system was compared with that of normal loop-tube system with a prime mover. As a result, we confirmed that parallel loop system can drive by lower onset temperature ratio than normal loop-tube system.

11:00

1aEA6. The effect of resonance mode control by expanding of cross-section area on cooling capacity in a loop-tube type thermoacoustic cooling system. Manabu Inoue (Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe 610-0321, Japan, dmm1011@mail4.doshisha.ac.jp), Shin-ichi Sakamoto (Univ. of Shiga Professor, Hikone, Japan), Yosuke Nakano, and Yoshiaki Watanabe (Doshisha Univ., Kyotanabe, Japan)

One of assignments for practical realization of a loop-tube type thermoacoustic system is improvement of cooling capacity. It is known that cooling capacity of the system is improved by setting phase adjuster (PA), which reduces cross-section of the system locally. This is considered that because setting PA enables to control resonance mode and setting PA controls to one wavelength. However, there is much dissipation of energy in PA because of reducing cross-section. More cooling capacity of the system is expected by control resonance mode at less dissipation of energy. Therefore, expanding phase adjuster (EPA) as a devise, which expands cross-section locally, is proposed to reduce dissipation of energy. If setting EPA enables to control resonance mode, cooling capacity is expected to improve more. At first, we make sure that setting EPA enables to control resonance mode. Next, we make a comparison between the system set EPA and PA. As a result, cooling capacity of the system with PA is higher than with EPA. However, more cooling capacity is expected by shift the condition of EPA because energy used on cooling in the system with EPA is more than with PA.

11:20

1aEA7. On discontinuity waves and vibrations in thermo-piezoelectric bodies. Adriano Montanaro (Dept. of Mathematics, Univ. of Padua, via Trieste, 63, Padova 35121, Italy, montanar@math.unipd.it)

With regard to a body composed of a linear thermo-piezoelectric medium, referred to a natural configuration, we consider processes for it constituted by small displacements, thermal deviations, and small electric fields superposed to the natural state. We show that any discontinuity surface of order r greater than 1 for the above processes is characteristic for the linear thermo-piezoelectric partial differential equations. We show that discontinuity surfaces of order 0 generally are not characteristic; hence, the conditions are written, which characterize the discontinuity surfaces of order 0 that are characteristic. We find the ordinary differential equations of propagation for plane progressive waves and standing waves. Then we characterize the ones whose wavefronts are characteristic.

11:40

1aEA8. Computational fluid dynamics simulation of Rayleigh streaming in a vibrating resonator. Joris P. Oosterhuis, Simon Bühler (Thermal Eng., Univ. of Twente, P.O. Box 217, Enschede 7500AE, Netherlands, j.p.oosterhuis@utwente.nl), Douglas Wilcox (Chart Inc., Troy, NY), and Theo H. van der Meer (Thermal Eng., Univ. of Twente, Enschede, Netherlands)

Rayleigh streaming is a time-averaged flow that can exist in the thermal buffer tubes of thermoacoustic prime movers and refrigerators and is driven by the viscous stresses close to the solid boundaries. This mean flow leads to mean convective heat transport, which can have large impact on the performance of thermoacoustic devices. Rayleigh streaming in a standing wave resonator is simulated using a commercially available computational fluid dynamics (CFD) code and is compared to existing analytical models of Hamilton *et al*. (2003). A test case is developed, and a standing wave is generated by applying a harmonic volume force to the domain. Both the inner and outer streaming vortices are well described for a range of radii from $R/\delta\nu = 3\ldots20$ and the magnitude of the streaming velocity matches analytical values. This paper shows the possibility of using available as-is CFD software for the simulation of streaming in a standing wave resonator. The presented results pave the way for the simulation of more complex geometries and studies to reduce the negative effects Rayleigh streaming can have on thermo-acoustic prime mover and refrigerator efficiency.

Session 1aMU

Musical Acoustics: String Instrument Measurements

Agnieszka Roginska, Cochair
New York Univ., 35 West 4th St., Rm. 1077, New York, NY 10012

Chris Waltham, Cochair
Phys. & Astronomy, UBC, 6224 Agricultural Rd., Vancouver, BC V6T 1Z1, Canada

Contributed Papers

9:00

1aMU1. Eigenvalue shapes compared to forced oscillation patterns of guitars. Malte Muenster, Jan Richter, and Rolf Bader (Systematic Musicologie, Univ. of Hamburg, Pilatuspool, 19, Hamburg, Hamburg 20355, Germany, m.muenster@arcor.de)

Thirty-two guitars are measured geometrically and acoustically. The geometries of the top plate with its bracing as well as its thickness, the back plate with all bracing, the ribs, and rims are transferred to a CAD model. The top plate and the back plate of these guitars are measured using a 121-microphone array, back-propagating the sound field onto the top and back plates. Therefore, the guitars are once driven by impulses at the guitar bridge, once by plucking all notes on all strings up to the 12th fret to reconstruct the forced-oscillation patterns. Large differences are found with respect to the basic modes between the different guitars in terms of frequency and shape of their eigenmodes. Comparing the measured and calculated eigenvalues with the forced-oscillation modes driven by the strings, it appears that the eigenmode shapes often differ from the forced-oscillation patterns considerably.

9:20

1aMU2. Calculating guitar sound radiation by forward-propagating measured forced-oscillation patterns. Jan Richter, Malte Münster, and Rolf Bader (Univ. of Hamburg, Gefionstrasse 11, Hamburg 22769, Germany, janrichter81@gmx.de)

The radiation patterns of 32 guitars are investigated. Therefore, the top and back plates are measured using a 121-microphone array, back-propagating the recorded sound field onto the guitar top and back plates. Both, the eigenvalues and the forced oscillation patterns are measured, the latter by plucking the guitar strings for all possible notes. For each note, the forced-oscillation radiation pattern is calculated for 20 partials up to 4 kHz. These radiation patterns are then forward-propagated into the surrounding space around the guitar. Considerable differences appear between the different guitars within the same frequency region in terms of shape and intensity. Also, for similar frequencies, different patterns may appear, depending on the string and note played.

9:40

1aMU3. Measuring the haptic behavior of an acoustic guitar as perceived by the player by means of a vibrating actuator. Marcello Giordano and Marcelo M. Wanderley (Input Devices and Music Interaction Lab., CIRMMT, McGill Univ., 555, Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, marcello.giordano@mail.mcgill.ca)

Two sets of recordings of the vibration produced by plucking the fifth and the second string of an acoustic guitar were acquired using an accelerometer secured to the neck of the instrument. Vibrations from both sets could be reproduced using a recoil-type vibrating actuator attached at the neck of the guitar. In one of the sets, salient spectral features of the original recordings were altered. We performed a preliminary study involving nine volunteer participants, blindfolded and artificially deafened using earplugs and loud white noise played through headphones. They were asked to discriminate, by holding the neck of the instrument with their left hand, between "fake" (i.e., actuator-produced) or real vibrations, produced by the experimenter plucking either string two or five. Our aim was to assess if any of the spectral features altered in the second set of recordings increased the recognition rate of actuator-produced vibrations as being "fake." These features, if present, would then be likely to carry crucial information and should be therefore modeled with extreme care in the simulation of the haptic behavior of the instrument. Results show that, at least for string five, we were capable of identifying one feature (a peak in the vibration spectrum located at 548 Hz), which, if altered, made the recognition rate as "fake" rise, statistically significantly, from 55% to 89%.

10:00

1aMU4. Acoustic imaging of string instrument soundboxes. Chris Waltham, Evert Koster, Nils Smit-Anseeuw, and Aaron Zimmer (Phys. & Astronomy, UBC, 6224 Agricultural Rd., Vancouver, BC V6T 1Z1, Canada, cew@phas.ubc.ca)

A circular 30-microphone array of 90 cm radius has been used to produce acoustic images of several string instruments. The soundboxes were excited by an automated impact hammer, and the instruments were suspended inside the array in such a manner that both the soundbox and hammer mechanism could be rotated in the horizontal plane. By normalizing all microphone signals to the hammer signal, data could be assembled as if from an array of many times 30 microphones. Images were formed using the inverse frequency response function method. The array and data analysis code were tested with a rectangular plate set in a large plywood baffle, a system that was straightforward to simulate numerically. Due to the limited spatial resolution set by the array geometry and frequency range—typically 200–1000 Hz—instruments with long soundboxes were chosen for initial testing: a gothic harp, a guzheng, and a guqin.

10:20–10:40 Break

10:40

1aMU5. Characterization of bridge motions on the violin using polymer sensor technology. Gunnar Gidion and Reimund Gerhard (Phys. and Astronomy, Univ. of Potsdam, Geschwister-Scholl-Str. 75, Potsdam, Brandenburg 14471, Germany, gunnargidion@web.de)

Recent developments in minimally invasive polymer-film sensors permit the *in-situ* detection of mechanical vibrations in musical instruments without significantly disturbing the acoustics of the instrument. As an example, we present measurements of vibrations of a violin between the feet of the bridge and the top plate. To this end, calibrated fluoropolymer-film sensors were matched to the geometry of the bridge feet. The forces exerted on the top plate by either bridge foot can be measured separately during excitation of a

string with the bow. The differences in amplitude and phase between bass and treble foot vibrations exhibit the distinctly asymmetrical nature of bridge motions, which of course also depend on the string and the note that are being played, respectively. In comparison with the simultaneously detected string vibrations and the radiated sound, the filter characteristics of the bridge are clearly identified in the spectral representation. As the bridge is also the main agent for the coupling from the body to the string, it is suggested that the observed variations in bridge motion are closely connected to the fact that the playability of a violin changes sometimes quite drastically from note to note.

11:00

1aMU6. High resolution radiation pattern measurements of a grand piano—The effect of attack velocity. Agnieszka Roginska, Justin Mathew, Jim Anderson (New York Univ., 35 West 4th St., Rm. 1077, New York, NY 10012, roginska@nyu.edu), and Alex U. Case (Univ. of Massachusetts Lowell, Lowell, MA)

The sound radiation pattern of a grand piano is highly complex and depends on the shape of the soundboard, construction of the frame, reflections from the lid, and other parts of the instrument's structure. The spectral energy generated by and emitted from the instrument is further complicated by the sound production mechanism (hammers, strings), the attack velocity, and results in independently complex behaviors depending on the register of the piano. This paper presents the acoustic measurements of the radiation pattern of a grand piano using a high spatial resolution measurement technique. Measurements of a Yamaha Diskclavier were taken using a 32-channel microphone array with a 2-in. spacing between capsules. The complex radiation patterns and overtone structure is analyzed for middle-C at three attack velocities—pianissimo, mezzo forte, and forte. Comparisons of the effect of attack strength on frequency response and radiation pattern are presented.

11:20

1aMU7. Properties of violin glides in the performance of cadential and noncadential sequences in solo works by Bach. Jiaxi Liu (Faculty of Music, Univ. of Cambridge, Darwin College, Silver St., Cambridge CB3 9EU, United Kingdom, jiaxi.liu1@gmail.com)

This study examines the articulatory changes ("glides") between the leading tone and tonic of cadential vs noncadential semitone sequences in solo violin performance. It was predicted that though these glides would have similar slopes, they would differ in duration and in semitone intonation, and that these latter properties could characterize the expression of cadential finality and the structural insignificance of noncadential sequences. Cadential (46) and noncadential (58) targets from 17 recordings by 13 professional violinists were analyzed using narrow-band spectrograms. Glide durations comprised 16% of the overall duration of semitone sequence irrespective of structure function. However, cadential glides comprised 28% of the duration of the leading tone compared with 11% for noncadential glides. As predicted, the leading tone tended to be sharp in both contexts, but the mean cadential interval was nonsignificantly larger by 18 cents, mainly because the tonic tended to be tuned more accurately in cadential sequences. Finally, the glide direction was linear and followed the natural vibrato trajectory in both contexts as expected. These data confirm that articulatory modifications play a prominent role in the performance of intended musical structure and suggest that such distinctions will influence structural expectancies.

MONDAY MORNING, 3 JUNE 2013 — 511BE, 8:55 A.M. TO 12:00 NOON

Session 1aNS

Noise: Advanced Hearing Protection and Methods of Measurement I

Christian Giguere, Cochair
Audiol./SLP Program, Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H8M5, Canada

Elliott H. Berger, Cochair
Occupational Health & Env. Safety Div., 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650

Chair's Introduction—8:55

Invited Papers

9:00

1aNS1. Attenuation as a function of the canal length of custom-molded earplugs. Jennifer B. Tufts, Siyuan Chen (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06269, jennifer.tufts@uconn.edu), and Lynne Marshall (Naval Submarine Medical Res. Lab., Groton, CT)

The fit of a custom-molded earplug (CMEP) and the amount of attenuation it provides can be affected by variables related to the original earmold impression and the subsequent manufacturing process. One variable thought to affect the amount of attenuation is the length of the canal portion of the CMEP. In this pilot study, we systematically examined the relationship between CMEP canal length and attenuation in four human subjects. Two men and two women were fitted with CMEPs extending past the second bend of the ear canal. The attenuation provided by the CMEPs was measured over four visits to the laboratory. Prior to each visit, the canal portion of the subject's test CMEP was shortened by 2 mm. As expected, attenuation decreased as canal length decreased for all subjects. However, the rate and pattern of decrease varied markedly. Anecdotal reports of comfort as a function of canal length also varied markedly. Results suggest that the critical region/s in the ear canal for maintaining a good acoustic seal may vary from person to person. Implications for future study will be discussed.

9:20

1aNS2. "Calibrating" the insertion depth of roll-down foam earplugs. Elliott H. Berger (Occupational Health & Env. Safety Div., 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650, elliott.berger@mmm.com)

Since introduction in 1972, roll-down slow-recovery foam earplugs have become nearly ubiquitous. They are used widely in industries and by consumers around the world. Their performance has been reported in numerous journal articles and they are often part of laboratory experiments, either as the object of the study or as a reference device that is used as a control or to assure exclusion of noise from the ear to facilitate data acquisition. As such it is important to be able to describe their performance since although they generally provide high levels of protection, the amount of protection and its spectral dependence is a function of insertion depth. Real-ear attenuation results will be presented for a range of insertions from that which caps the earcanal to full earcanal insertion past the second bend. The results will be compared to published data to demonstrate how they can be used to estimate the quality of fit that was likely achieved vs that which was reported. These data will be useful to researchers who wish to "calibrate" the quality of insertion they are achieving in their own studies going forward. [The author is an employee of 3M and the research was funded by 3M.]

9:40

1aNS3. Attenuation characteristics of fit-compromised earmuffs and various non-standard hearing protectors. Laurie Wells, Elliott H. Berger, and Ron W. Keiper (Occupational Health & Env. Safety Div., 3M, 3M Ctr., 235-2W-75, St. Paul, MN 55144-1000, Laurie.Wells@mmm.com)

Excessive noise exposure can be successfully mitigated by proper use of legitimate hearing protection devices. However, real-life circumstances sometimes drive people to use compromised or alternative means of protection. This paper reports attenuation data measured in the 3M E•A•RCAL facility over several years, in conformance with ANSI real-ear attenuation at threshold test standards (S3.19-1974, S12.6-1984, and S12.6-2008 Method A) and also provides, for comparison, one dataset from the open literature (fingers/palms). The loss of attenuation was measured for various earmuffs worn in less than ideal conditions, including earmuffs worn in conjunction with various safety glasses, hairnets, head covers, hoods, earmuff cushion covers, and ball caps. Data were also obtained for non-standard means of blocking sound, including long hair, cotton balls, and even use of palms and/or fingers to block the ears. Results demonstrated that the effects on earmuff attenuation varied from none at all (suitable cushion cover) to as much as 12 dB (hooded sweatshirt). Realizing that people adapt hearing protectors to meet their needs is one step toward optimizing hearing protection selection and use; knowing the significance of these adaptations is the next step. [The authors are employees of 3M and the research was funded by 3M.]

10:00

1aNS4. Comparison of subjective and objective methods for the measurements of hearing protector devices attenuation and occlusion effect. Hugues Nélisse (Service de la recherche, IRSST, 505 Blvd De Maisonneuve Ouest, Montreal, QC H3A 3C2, Canada, hugues.nelisse@irsst.qc.ca), Cécile Le Cocq (Département de génie mécanique, École de Technologie Supérieure, Montréal, QC, Canada), Jérôme Boutin (Service de la recherche, IRSST, Montréal, QC, Canada), Jérémie Voix, and Frédéric Laville (Département de génie mécanique, École de Technologie Supérieure, Montréal, QC, Canada)

With the increase popularity of individual fit testing and miniaturization of electronic components, the field-microphone-in-real-ear approach (F-MIRE) is becoming more appealing and well suited for estimating hearing protection devices (HPD) attenuation both in laboratory and in "real world" occupational conditions. The approach utilizes two miniature microphones to simultaneously measure the sound pressure levels in the ear canal under the hearing protector, as well as outside of the protector. In this study, experiments on several human subjects were carried out in order to examine the various factors relating the subjective and objective attenuation values. The subjects were first instrumented on both ears with miniature microphones outside and underneath the protector. They were then asked to go through a series of subjective hearing threshold measurements followed by objective microphone recordings using high level diffuse field broadband noises. Earmuffs, earplugs, and double-protection were tested for each subject, and attenuation values were compared. Additionally, an objective scheme to measure the occlusion effect was developed and tested using subjects' voice as the excitation and the same microphone setup. Results obtained for the attenuation values as well as the occlusion effect levels are presented and discussed.

10:20–10:40 Break

Contributed Papers

10:40

1aNS5. Implementation of a simplified, artificial external ear test fixture for measurement of the earplug induced auditory occlusion effect. Martin Brummund (Dept. of Mech. Eng., École de technologie supérieure, 1100, rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, martin.brummund.1@ens.etsmtl.ca), Franck Sgard (Service de la recherche, IRSST, Montreal, QC, Canada), Yvan Petit, Frédéric Laville (Dept. of Mech. Eng., École de technologie supérieure, Montreal, QC, Canada), and Jérôme Boutin (Service de la recherche, IRSST, Montreal, QC, Canada)

Earplugs remain a frequently used short-term solution for occupational hearing conservation. Due to comfort limitations, as induced by, e.g., the occlusion effect, workers often only wear earplugs for limited amounts of time and are likely to develop professional hearing loss. The occlusion effect expresses itself in the low frequencies through an altered perception of the wearer's own voice and the amplification of physiological noises that occur upon earplug insertion. While many studies examined the occlusion effect experimentally, no study was found that attempted to implement an artificial external ear model dedicated to the measurement of the objective occlusion effect. A simplified external ear test fixture can help to better assess and design earplugs, because it allows standardized experimental testing. This work describes the implementation of a cylindrical artificial test fixture of the human outer ear that comprises the auditory canal as well as the bony, cartilaginous, and skin tissues that are made up of rigid polyurethane foam and two different types of silicone, respectively. Obtained measurement results are compared to literature findings.

11:00

1aNS6. Impulse noise attenuation by earplugs measured with the use of an acoustical test fixture and with the participation of subjects. Rafal Mlynski and Pawel Gorski (Wibroacoustic Hazards, Central Inst. for Labour Protection – National Res. Inst., Warszawa, Poland, pawel@ciop.pl)

The effectiveness of impulse noise attenuation by hearing protector devices is most often determined collecting the data from measurements. In impulse noise conditions with high peak sound pressure level, it is necessary to replace a subject in a measurement with an acoustical test fixture. The use of the acoustical test fixture is important because of the potential risk of hearing damage occurring during impulse noise tests, performed with the participation of subjects. The impulse noise attenuation by earplugs determined from measurements carried out using acoustical test fixture was compared with attenuation determined with the participation of subjects (MIRE technique). The acoustical test fixture complied with the acoustic and mechanical requirements described in Standard No. ISO 4869-3 and was equipped with a chamber representing the external ear canal and a 2 cm^3 chamber reflecting the acoustic properties of the middle ear. The results of measurements carried out with two different methods were comparable.

11:20

1aNS7. Influence of the external ear tissue domains on the sound attenuation of an earplug predicted by a finite element model. Guilhem Viallet (Mech. Eng., École de technologie supérieure, 1100, rue Notre-Dame ouest, Montreal, QC H3C 1K3, Canada, guilhem.viallet.1@ens.etsmtl.ca), Franck Sgard (Noise and Vib., Institut de Recherche de Robert Sauvé en santé et en sécurité du travail, Montreal, QC, Canada), and Frédéric Laville (Mech. Eng., École de technologie supérieure, Montreal, QC, Canada)

Earplugs are a widespread solution to prevent the problem of hearing loss in the workplace environment, but they do not always perform as desired. Using a model of the ear canal occluded by an earplug could be helpful to perform sensitivity analyses (geometry and materials of the earplug) and to better understand the role of the earplug. The human external ear is a complex system made up of different tissues with a 3D geometry. In practice, it is reduced to a 2D cylindrical geometry for the acoustical tests fixtures. The purpose of this study is to compare the insertion loss predicted by a 3D complex finite element model of the ear canal surrounded by different tissue domains (skin, soft tissue, and bone) and occluded by a silicon earplug versus a 2D axisymmetric model of the same system. In both models, some investigations are made in order to verify if the models could be simplified by replacing the tissue domains by mechanical impedances. These investigations are made to reduce the complexity of the models and to discuss the relevance of whether or not including external ear tissue domains in a sound attenuation model of an earplug.

11:40

1aNS8. Sound transfer path analysis to model the vibroacoustic behavior of a commercial earmuff. Sylvain W. Boyer (Département de Génie Mécanique, École de Technologie Supérieure, 1100 rue Notre-Dame, Ouest, Montréal, QC H3C 1K3, Canada, sylvain.boyer.1@ens.etsmtl.ca), Olivier Doutres (Groupe d'Acoustique de l'Universite de Sherbrooke, Université de Sherbrooke, Sherbrooke, QC, Canada), Franck Sgard (Institut de Recherche Robert-Sauvé en Santé et en Sécurité du Travail, Montréal, QC, Canada), Frédéric Laville (Département de Génie Mécanique, École de Technologie Supérieure, Montréal, QC, Canada), and Jérôme Boutin (Groupe d'Acoustique de l'Universite de Sherbrooke, Université de Sherbrooke, Montréal, QC, Canada)

Hearing protection devices (HPD), such as earmuffs, are widely used to protect workers from noisy environments. Numerical predictive tools can be used to simulate the vibroacoustic behavior of earmuffs and thus assess their sound attenuation and improve their acoustical design. The present work describes the implementation of a vibroacoustic finite element numerical model of an earmuff coupled to a rigid baffle in the frequency range from 20 to 5000 Hz. An experimental assessment of the sound transfer paths through each element of the earmuff (cup, cushion, and foam lining) using a specific acoustical test bench is first proposed. This analysis is then used to target the right level of model complexity for each component. An experimental validation of the FEM model is then carried out.

MONDAY MORNING, 3 JUNE 2013 | 519B, 8:55 A.M. TO 11:20 A.M.

Session 1aPAa

Physical Acoustics and Biomedical Acoustics: Acoustics in Microfluidics and for Particle Separation I: Standing Waves, Streaming, and Radiation Forces

Lawrence A. Crum, Cochair
Appl. Phys. Lab., Univ. of Washington, Ctr. for Industrial and Medical Ultrasound, Seattle, WA 98105

Michel Versluis, Cochair
Phys. of Fluids Group, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands

Chair's Introduction—8:55

Invited Papers

9:00

1aPAa1. First-principle simulation of the acoustic radiation force on microparticles in ultrasonic standing waves. Mads Jakob Herring Jensen (COMSOL A/S, Diplomvej 373, Lyngby 2800, Denmark, mads@comsol.dk) and Henrik Bruus (Dept. of Phys., Techn. Univ. of Denmark, Lyngby, Denmark)

The recent development in the field of microparticle acoustophoresis in microsystems has led to an increased need for more accurate theoretical predictions for the acoustic radiation force on a single microparticle in an ultrasonic standing wave. Increasingly detailed analytical solutions of this specific problem can be found in the literature [Settnes and Bruus, Phys. Rev. E **85**, 016327 (2012), and references therein], but none have included the complete contribution from thermoviscous effects. Here, we solve this problem numerically by applying a finite-element method to solve directly the mass (continuity), momentum (Navier-Stokes), and energy conservation equations using perturbation theory to second order in the imposed time-harmonic ultrasound field. In a two-stage calculation, we first solve the first-order equations resolving the thermoviscous boundary layer surrounding the microparticle and with a perfectly matched layer as a non-reflecting boundary condition for the scattered waves. These first-order solutions are then used as source-terms for solving the time-averaged second-order equations [Muller *et al.*, Lab Chip **12**, 4617 (2012)] and in particular to determine the second-order time-averaged hydrodynamic stress on the particle surface. From this, we deduce the radiation force and compare it as a function of the physical parameters to existing analytical results.

9:20

1aPAa2. Acoustic standing wave based microsystem for low-concentration oil detection and separation. Han Wang (Dept. of Elec. and Comput. Eng., Texas A&M Univ., College Station, TX), Zhongzheng Liu (Dept. of Mech. Eng., Texas A&M Univ., College Station, TX), Chiwan Koo (Dept. of Biomed. Eng., Texas A&M Univ., College Station, TX), Sungman Kim (Dept. of Elec.and Comput. Eng., Texas A&M Univ., College Station, TX), Younghak Cho (Dept. of Mech. Syst. Design Eng., Seoul National Univ. of Sci. and Technol., Seoul, Republic of Korea), Yong-Joe Kim (Dept. of Mech. Eng., Texas A&M Univ., College Station, TX), and Arum Han (Dept. of Elec. and Comput. Eng., Texas A&M Univ., 309C WERC, TAMU 3128, College Station, TX 77843-3128, arum.han@ece.tamu.edu)

Detection and quantification of extremely small amount of oil on site and at low cost has broad applications in environmental monitoring, both in oil spills as well as in routine marine/coastal ecosystem monitoring. For example, dispersed oil, generated through the use of chemical dispersants in oil spills to break up oil slick into small droplets so that they can be rapidly diluted in 3D space, are the greatest concern and poses the most challenges in detection. Fluorometry is the current standard method, however is bulky and expensive, limiting its wide deployment in the field. Here we demonstrate for the first time the development of an acoustic standing wave based microfluidic platform capable of processing large amount of liquid samples from which dispersed oil can be concentrated and separated to a detectable level by acoustophoretic force. The microfluidic platform consists of a recirculation channel structure into which dispersed oil droplets can be continuously separated from the main sample flow stream. A piezoelectric transducer attached at the bottom of the silicon-glass microfluidic channel creates the acoustic standing wave that exerts acoustophoretic force to oil droplets. An optical detector measures the presence of concentrated oil droplets by their distinct fluorescent signatures.

Contributed Papers

9:40

1aPAa3. Large volume flow rate acoustophoretic phase separator for oil water emulsion splitting. Jason P. Dionne (FloDesign Sonics Inc., 380 Main St., Wilbraham, MA 01095, j.dionne@fdsonics.com), Brian McCarthy, Ben Ross-Johnsrud (Mech. Eng., Western New England Univ., Springfield, MA), Louis Masi (FloDesign Sonics Inc., Wilbraham, MA), and Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA)

Efficient separation technologies for multi-component liquid streams that eliminate waste and reduce energy consumption are needed. Current technologies suffer from high cost of energy, use of consumables, fouling, and limited separation efficiency of micron-sized particles. We propose a novel platform technology consisting of a large volume flow rate acoustophoretic phase separator based on ultrasonic standing waves. The acoustic resonator is designed to create a high intensity three dimensional ultrasonic standing wave resulting in an acoustic radiation force that is larger than the combined effects of fluid drag and buoyancy, and is therefore able to trap, i.e., hold stationary, the suspended phase. The action of the acoustic forces on the trapped particles results in concentration, agglomeration, and/or coalescence of particles and droplets. Heavier than water particles are separated through enhanced gravitational settling, and lighter particles through enhanced buoyancy. A first prototype consists of a 2 in. by 1 in. flow chamber driven by a single 1 in. by 1 in. transducer at 2 MHz, with flow rates of 30 L/h, and measured oil separation efficiencies in excess of 95%. A second prototype is designed to further scale the system to flow rates of 150 L/h. [Work supported by NSF SBIR 1215021 and NSF PFI:BIC 1237723.]

10:00

1aPAa4. Acoustic radiation force on a sphere without restriction to axisymmetric fields. Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Lab., The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, hamilton@mail.utexas.edu)

The analysis presented at the previous ASA meeting related to investigation of the acoustic radiation force on a sphere embedded in a soft elastic medium with shear modulus that is several orders of magnitude smaller than its bulk modulus. The acoustic field was assumed to be axisymmetric and the spherical scatterer to be located on the axis of the acoustic beam. When one of these conditions is violated, the problem loses its symmetry. In this talk, the acoustic radiation force is considered in the more general case of nonaxisymmetric fields. The calculation is performed in Lagrangian coordinates. All acoustic fields, incident as well as scattered, depend on all three spherical coordinates. The incident and scattered waves, which include both potential and solenoidal parts, are expanded with respect to spherical harmonics. An analytical expression for the acoustic radiation force derived in this investigation may contain as many spherical harmonics as needed. In limiting cases when the scatterer is in liquid and only two modes, monopole and dipole, remain in the scattered fields, the solution for the acoustic radiation force recovers the results reported by Gor'kov [Sov. Phys. Doklady **6**, 773 (1962)]. [Work supported by NIH DK070618 and EB011603.]

10:20

1aPAa5. Viscous contributions to low-frequency scattering, power absorption, radiation force, and radiation torque for spheres in acoustic beams. Philip L. Marston (Phys. and Astron. Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

An analysis of the dipole response of solid spheres illuminated by plane acoustic traveling waves [Settnes and Bruus, Phys. Rev. E **85**, 016327 (2012)] has implications for estimating the magnitude of viscous corrections to quantities of broader interest. Their results may be recast to give the viscous correction to the dipole scattering s-function for solid spheres. For the present discussion, it may be assumed that the Stokes layer is thin relative to the sphere's radius, giving a simple reduction in magnitude of the dipole s-function (which is unimodular in the lossless case). The power absorption for plane waves and Bessel beams follow immediately from a prior formulation [Zhang and Marston, Phys. Rev. E **84**, 035601 (2011)] as does the axial radiation force. The plane-wave force agrees after correcting a minor error in Settnes and Bruus. A condition $G = 0$ previously noted for low-ka negative radiation forces on spheres in Bessel beams [Marston, J. Acoust. Soc. Am. **120**, 3518–3524 (2006)] still gives negative forces for sufficiently large spheres. The torque caused by first-order vortex beams may also be estimated [Zhang and Marston, Phys. Rev. E **84**, 065601 (2011)]. [Work supported by ONR.]

10:40

1aPAa6. Direct numerical simulations of acoustic streaming in standing wave tubes using the Lattice Boltzmann Method. Yasser Rafat, Kaveh Habibi, and Luc Mongeau (Mech. Eng., Mcgill Univ., 817 Sherbrooke St. West, Montreal, QC, Canada, yasser.rafat@mail.mcgill.ca)

One important factor in the efficiency of thermoacoustic engines is acoustic streaming, which causes convective heat transfer between high and low temperature reservoirs. Most experimental and numerical studies performed so far have focused on Rayleigh streaming. Less work has been done on acoustic streaming due to the stack. Most numerical studies of Rayleigh streaming were performed using Navier-Stokes based numerical methods. In this study, large eddy numerical simulations were performed using schemes based on the lattice Boltzmann method (LBM). The model considered a simplified thermoacoustic refrigerator made of a rectangular standing wave resonator with a flat plate spoiler. Low-amplitude results obtained for Rayleigh streaming velocity magnitudes were compared with linear acoustic theory for verification. High amplitude recirculated streaming flow structures around the edges of the flat plate spoiler were identified. These are likely to contribute to heat transfer much more than Rayleigh streaming. Parametric studies were performed to investigate the effects of Strouhal number and spoiler edge shape. The results confirm that vertical edge streaming flows play a significant role in thermoacoustic heat transport.

11:00

1aPAa7. Three-dimensional analysis of the acoustic radiation pressure: Application to single-beam acoustical tweezers. Diego Baresch, Régis Marchiano (Institut Jean le Rond d'Alembert, UMR CNRS 7190, UPMC-CNRS, 4, Place Jussieu, Paris 75005, France, diego.baresch@upmc.fr), and Jean-Louis Thomas (Institut des NanoSciences de Paris, UMR CNRS 7588, UPMC-CNRS, Paris, France)

Recent studies on the acoustic radiation forces exerted by sound impinging spherical objects suggest the use of structured wavefronts for particle entrapment and controlled manipulation. In the scope of understanding why it is made possible to trap and manipulate small particles with sound, we present a general model for the acoustic radiation forces in three dimensions. A first generalization comes from the extension of well known results for the radiation pressure of plane waves to incident wavefields having arbitrary wavefronts. Second, the elastic spherical target of any dimension is allowed to be arbitrarily located within the wavefield. Introducing a new class of "single-beam" acoustical tweezers, we discuss the capabilities of different acoustical beams to achieve particle trapping and manipulation tasks. In addition, using an efficient experimental setup, we report the propagation of a peculiar beam carrying orbital angular momentum, namely an acoustical vortex, which is our selected candidate to achieve the first three-dimensional acoustic trap for elastic particles.

MONDAY MORNING, 3 JUNE 2013 — 519B, 11:20 A.M. TO 12:00 NOON

Session 1aPAb

Physical Acoustics and Biomedical Acoustics: Acoustics in Microfluidics and for Particle Separation IIa: Bubbles and Drops

Yong-Jae Kim, Cochair
Mech Eng., Texas A&M Univ., College Station, Texas 77843

Martin Wiklund, Cochair
Appl. Phys., Royal Inst. of Technol., KTH-Albanova, Stockholm 10691, Sweden

Invited Papers

11:20

1aPAb1. Perturbation analysis of flow about spherically pulsating bubble at the velocity node of a standing wave. Mohammad AlHamli (Aerosp. & Mech. Eng., Univ. of Southern California, Olin Hall OHE 430 (MC-1453), Los Angeles, CA 90089-1453, moalhamli@gmail.com), Alexey Y. Rednikov (TIPs-Fluid Phys., Université Libre de Bruxelles, Brussels, Belgium), and Satwindar S. Sadhal (Aerosp. & Mech. Eng., Univ. of Southern California, Los Angeles, CA)

An analysis using the singular perturbation method for a radially pulsating gas bubble at the velocity node of a standing wave was conducted with $\varepsilon = U_0/(a\omega) \ll 1$ as a small parameter and $\omega a^2/\nu \gg 1$ as a large parameter. Here,a, U_0, ω, and ν are length scale, velocity scale, frequency, and kinematic viscosity, respectively. While the mean oscillatory flow around the gas bubble has no net time-averaged flow component, viscous steady streaming arises due to the nonlinearity of the flow dynamics. However, with bubble surface being considered shear-free, the vorticity generation in the system is quite weak as compared with what would result from a solid boundary. Not surprisingly, the steady streaming is also weak. As already known, the steady streaming would not arise with purely radial pulsations of a bubble in an otherwise quiescent liquid. For the case of a non-pulsating bubble at the velocity node, streaming is seen at $O(\varepsilon^2)$. However, as seen with the case of a radially pulsating bubble at the velocity antinode, interaction of two oscillatory fields creates streaming at lower order. The phase difference between radial and lateral oscillations was found to play a significant role in both the streaming direction and intensity.

11:40

1aPAb2. Acoustic bubble sorting of ultrasound contrast agents. Michel Versluis (Phys. of Fluids Group, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, m.versluis@utwente.nl)

Ultrasound contrast agents are coated microbubbles with radii ranging from 1 to 10 μm. Medical transducers typically operate at a single frequency; therefore, only a small selection of bubbles will contribute to the nonlinear contrast through resonance. Thus, the sensitivity of contrast-enhanced ultrasound can be improved by narrowing down the size distribution. Monodisperse bubble can be formed in a flow-focusing geometry. However, it requires extensive skills in microfluidics technology and in surface chemistry. Here, we present a simple lab-on-a-chip technique to sort microbubbles on-line in a traveling ultrasound wave. A broad range of the parameter space of bubble size and frequency has been characterized to provide physical input parameters for a simple force balance model. We find good agreement for the modeled displacement as a function of the bubble radius for a range of sizes in the unbounded fluid. Within the confinement of the sorting chip, we find good agreement for the resonance behavior and overall with a smaller displacement than predicted as a result of bubble–wall interactions.This novel sorting strategy may lead to an overall improvement of the sensitivity of contrast echo of at least one order of magnitude.

MONDAY MORNING, 3 JUNE 2013 514ABC, 8:55 A.M. TO 12:00 NOON

Session 1aPP

Psychological and Physiological Acoustics: In Memory of Bertram Scharf: Five Decades of Contributions to Auditory Perception

Mary Florentine, Cochair
SLPA & ECE, Northeastern Univ., 106-A FR, 360 Hutntington Ave., Boston, MA 02115

Huanping Dai, Cochair
Speech Lang. and Hearing Sci., Univ. of Arizona, P.O. Box 21071, 1131 E. 22nd St., Tucson, AZ 85721-0071

Chair's Introduction—8:55

Invited Papers

9:00

1aPP1. Bertram Scharf and his critical contributions to the field. Harry Levitt (Adv. Hearing Concepts, P.O. Box 610, Bodega Bay, CA 94923, harrylevitt@earthlink.net)

Jerry Tobias introduced me to Bert Scharf via his book Foundations of Modern Auditory Theory. Our research in the 1970s was on the prediction of speech intelligibility from acoustic measurements; the similarity between Fletcher's 20 frequency bands of equal contributions to intelligibility and psychoacoustic measurements of critical bands was of great interest. Bert's chapter on critical bands provided an insightful, comprehensive, concise, and critical review of the state of the art in critical-band research. Our mutual research interests brought us together when he asked me to review a draft of a paper he was preparing. My review was highly critical and after submitting it to him I felt I had been too unforgiving in my review (reviewers can be wrong). I called him to explain that I had been overly critical of a fine paper. His response was the opposite of what I expected. He said it was the best review he had received and that he was extremely grateful for my input. I realized then that he—not only set high standards for others—but for himself as well. By adhering consistently to his high standards, his many contributions to the field have been long-lasting and critical.

9:20

1aPP2. An overview of Bertram Scharf's research in France on loudness adaptation. Sabine Meunier (LMA-CNRS-UPR 7051, Aix-Marseille Univ, Centrale Marseille, 31 chemin Joseph-Aiguier, Marseille 13402, France, meunier@lma.cnrs-mrs.fr)

Since 1978, Professor Bertram Scharf divided his time between the United States and France. He was a Visiting Scientist at the Laboratoire de Mécanique et d'Acoustique in Marseille until the mid-1990s and collaborated with the University of Marseille (Faculté de Médecine) until his death. One of Bertram Scharf's major contributions to the field of psychoacoustics is in the area of loudness. He first studied spectral loudness summation, when he started working at Harvard University. In France, his work on loudness focused mainly on loudness adaptation. He wrote, "Loudness resembles pain in that it decreases as a function of time only under special stimulus conditions." Bertram Scharf's work with his French colleagues defined aspects of loudness adaptation in its direct (simple loudness adaptation) and indirect (induced loudness adaptation) forms. They studied how the auditory system recovers from loudness adaptation and examined a possible physiological basis for loudness adaptation.

9:40

1aPP3. Spectral loudness summation: From the 1960s to the present. Jesko L. Verhey and Jan Hots (Dept. of Exp. Audiol., Otto von Guericke Univ. Madgeburg, Leipziger Str. 44, Magdeburg 39120, Germany, jesko.verhey@med.ovgu.de)

In general, the loudness level of a broader sound is higher than that of a narrower sound centered at the same frequency, an effect commonly referred to as spectral loudness summation. From the late 1950s onwards, Scharf published several articles on this topic investigating how stimulus parameters such as the level, number, and spectral separation of components of a complex tone and the spectrum shape affects the magnitude of spectral loudness summation, and how spectral loudness summation is altered in hearing-impaired listeners and under masking. In a contribution to the proceedings of the first international symposium on hearing, Scharf also provided important information of the effect of duration on spectral loudness summation, stimulating our own research in this field. This talk will provide an overview of Scharf's work on spectral loudness summation and how the view on this topic has changed over time. It will be shown that even today there are aspects of this effect that are not completely understood in the light of current loudness models.

10:00–10:20 Break

10:20

1aPP4. Context effects in loudness. Michael Epstein (Auditory Modeling and Processing Lab., Speech-Lang. Pathol. and Audiol., Northeastern Univ., 360 Huntington Ave., 106A FR, Boston, MA 02115, m.epstein@neu.edu) and Mary Florentine (Commun. and Digital Signal Processing Ctr., Speech-Lang. Pathol. and Audiol., Northeastern Univ., Boston, MA)

Bertram Scharf made contributions to numerous topics in the loudness literature. In particular, he brought a great deal of insight into the current understanding of contextual effects in loudness. Some of the contextual effects that he studied include: (1) loudness adaptation, the decline in loudness of the latter portion of a continuous sound, (2) induced loudness reduction, the phenomenon by which a preceding stronger tone reduces the loudness of a weaker tone, (3) temporary loudness shift, a decline in the loudness of weaker sounds due to a physical fatigue of the cochlear amplifier, and (4) loudness enhancement, in which a brief sound is made louder when it follows a stronger sound within a short duration. Context effects serve as complex reminders of the necessity of careful design of any psychoacoustical experiment in which level varies. These effects also result in the breakdown of all loudness models, as virtually all calculations of loudness are performed for sounds without regard for previous stimuli. [Work supported by NIH-NIDCD.]

10:40

1aPP5. Connecting cues to signals in auditory attention. Ervin Hafter (Psychology, Univ. of California, 1854 San Lorenzo Ave., Berkeley, CA 94707, hafter@berkeley.edu)

Among his many fields of study, Bert Scharf played a major role in our understanding the role of auditory attention, especially at the level of basic psychophysics. Of special importance to this talk is the profound influence that he had on research in our lab (myself, Bert Schlauch, Joyce Tang, Kourosh Saberi, and Poppy Crum) through his work on signal uncertainty in masking and its alleviation by specific informational cues. Scharf's resurrection of the probe-signal method led us to examine the effects of uncertainty on both the means and variances of effective bandwidths used by listeners in a detection task. Also shown will be cases where we used different kinds of cues to study detection at various levels of processing including judgments based on: specific spectral components, complex pitches derived from sets of harmonics and locations in frequency reliant on mentally tracking an FM stimulus through a period of occlusion.

11:00

1aPP6. Neural correlates of auditory attention. Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu)

Bert Scharf's seminal studies on selective auditory attention were, in many ways, ahead of the times. Twenty years ago, many psychoacousticians viewed any consideration of "cognitive" factors or any effects driven by the intent of the listener, rather than the acoustics of the input, as outside of their realm of interest. However, today, a plethora of laboratories are exploring questions about what acoustic features enable listeners to focus attention, how bottom-up stimulus attributes interact with top-down control signals to determine what source a listener attends in a mixture of sources, and what neural mechanisms realize such selective auditory attention. This talk reviews recent work exploring selective auditory attention using a combination of behavioral studies and neuro-imaging techniques, all of which suggest that (1) listeners can focus attention on one, and only one, auditory object or stream at a time, and (2) executive control regions of the brain are engaged during attention to reduce across-object interference in the representation of whatever object is in the attentional foreground. These studies underscore the importance of auditory attention in allowing us to communicate in everyday settings containing multiple sound sources, and thus the foresight of Bert in tackling this problem when most others did not.

11:20

1aPP7. Tuning in the time domain revealed through detection of auditory signals of unexpected duration or presentation time. Beverly Wright (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60202, b-wright@northwestern.edu) and Huanping Dai (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

Bert Scharf was interested in how expectation affects auditory performance. He explored this question using the probe-signal method of Greenberg and Larkin, in which the listener is led to expect a particular stimulus but is occasionally presented with an unexpected but equally detectable one. The detectability of unexpected stimuli provides insight into the listener's template for the expected stimulus. Bert's expectation research, which focused on the frequency domain, inspired us to extend the inquiry to the time domain. We have seen that signal detection can be quite poor for signals of unexpected duration as well as for signals presented at unexpected times, indicating that listeners attend selectively to these two temporal aspects of sound. However, this temporal tuning is much broader for starting time (hundreds of milliseconds) than for signal duration (can be <25 ms). Thus, it appears that listeners can select the template for signal detection with considerable accuracy, but do not apply the selected template strictly to the expected starting time of the signal. We are grateful to Bert for his mentorship and keen interest in this topic. [Work supported by NIH/NIDCD.]

11:40

1aPP8. Inspiration from Bertram Scharf's work. Fan-Gang Zeng (Ctr. for Hearing Res., Depts. of Anatomy and Neurobiology, Biomedical Eng., Cognit. Sci. and Otolaryngol.– Head and Neck Surgery, Univ. of California, 110 Med Sci E, Irvine, CA 92697, fzeng@uci.edu)

In 1990, Bertram Scharf and I discussed about me doing a post doc in his laboratory. The opportunity to work with Bertram did not materialize, but his work in loudness, efferents, and attention has been a continuing inspiration not only for my own research but also for auditory perception, physiology, and audio engineering in general. For example, a Google Scholar search of "Bertram Scharf" on November 8, 2012, produced 1517 citations for his top 10 papers, with 6 on loudness and critical bands, 2 on efferents, and 2 on attention. Here I highlight two recent projects that have been inspired by Scharf's work. The first project showed significant loudness adaptation in patients with auditory neuropathy, particularly those with otoferlin deficits. This result directly supports Scharf's proposition that simple loudness adaptation is due to a sensory process, which in this case can be pinned down to transmitter release and replenishment in the hair cell and nerve synapse. The second project extended Scharf's theoretical work in efferents and attention to improving feedback control in cochlear implant users and tinnitus sufferers. This line of work could improve cochlear implant speech perception in noise and reduce internal gain to alleviate tinnitus.

Session 1aSA

Structural Acoustics and Vibration: Measurement and Modeling of Structures with Attached Noise Control Materials I

Franck C. Sgard, Cochair
IRSST, 505 Boulvd de Maisonneuve O, Montreal, QC H3A3C2, Canada

Noureddine Atalla, Cochair
GAUS Mech. Eng., Univ. of Sherbrooke, Sherbrooke, QC J1K 2R1, Canada

Chair's Introduction—8:55

Invited Papers

9:00

1aSA1. Tunneling effect on the sound transmission loss of a flat structure coupled with a porous material. Franck C. Sgard (Direction Scientifique, IRSST, 505 Boulvd de Maisonneuve O, Montreal, QC H3A3C2, Canada, frasga@irsst.qc.ca), Noureddine Atalla, Mohammad Gholami (GAUS, Univ. of Sherbrooke, Montreal, QC, Canada), and Hugues Nelisse (Direction Scientifique, IRSST, Montreal, QC, Canada)

It is well known that when measuring sound transmission loss (STL) in a laboratory, among all test conditions, the location of a specimen in an aperture affects the results, due to the tunneling effect. Previous studies have considered this effect for flat single panels and double walls but the case of a panel with attached sound package seems to have received very little attention. This paper deals with the application of a modal approach to study the STL of a rectangular plate coupled with a porous material located inside a tunnel. The sound absorbing material is supposed to be described by a transfer matrix calculated using a Transfer Matrix Method, which relates interstitial pressure and total normal stress on both sides of the material. The model is validated by comparison with finite element/boundary element computations. Numerical results are shown to illustrate the validity of the proposed hybrid modal-TMM methodology and its use to investigate the niche effect in the presence of a sound absorbing material.

9:20

1aSA2. A hybrid modeling approach for vibroacoustic systems with attached sound packages. Luca Alimonti, Noureddine Atalla, Alain Berry (Mech. Eng., Universitè de Sherbrooke, 1747 Rue Marcil, Sherbrooke, QC J1J 2H7, Canada, luca.alimonti@usherbrooke.ca), and Franck Sgard (IRSST, Montreal, QC, Canada)

Modeling complex vibroacoustic systems including poroelastic materials using finite element (FE) based methods can be computationally expensive. Several attempts have been made to alleviate this drawback, such as high order hierarchical basis and substructuring approaches. Still, these methods remain computationally expensive or limited to simple configurations. On the other hand, analytical approaches, such as the Transfer Matrix Method (TMM), are often used, thanks to the lower computational burden. However, since the geometrical flexibility of the FE method is always needed in the low/mid-frequency range, attempts have been made to couple the FE model of the master system with a TM model of the sound package. Although these hybrid approaches seem promising, the open literature is not comprehensive. The aim of this work is to present a hybrid FE-TMM approach based on a Green's function formulation. The idea is to account for the sound package by approximating the effects over the treated surface using fundamental solutions (i.e., Green's functions) obtained by the TMM. A benchmark representative of typical applications is used to illustrate the capabilities of the presented methodology in terms of efficiency and accuracy in comparison to other classical methods.

9:40

1aSA3. A finite element solution strategy based on Padé approximants for fast multiple frequency sweeps of coupled elastic, poroelastic, and internal acoustic problems. Romain Rumpler and Peter Göransson (Aeronautical and Vehicle Eng., KTH Royal Inst. of Technol., Teknikringen 8, Stockholm 10044, Sweden, rumpler@kth.se)

Analyses involving structural-acoustic finite element models including three-dimensional modeling of porous media are, in general, computationally costly. While being the most commonly used predictive tool in the context of noise and vibrations reduction, efficient solution strategies enabling the handling of large-size multiphysics industrial problems are still lacking, particularly in the context where multiple frequency response estimations are required, e.g., for topology optimization, multiple load cases analysis, etc. In this work, an original solution strategy is presented for the solution of multi-frequency structural-acoustic problems including poroelastic damping. Based on the use of Padé approximants, very accurate interpolations of multiple frequency sweeps are performed, allowing for substantial improvements in terms of computational resources, i.e., time and memory allocation. The method is validated and demonstrated for its potential on 3D applications involving coupled elastic, poroelastic, and internal acoustic domains.

10:00

1aSA4. Discontinuous Galerkin Methods for poroelastic materials. Olivier Dazel (LAUM UMR CNRS 6613, University Le Mans, Avenue Olivier Messiaen, Le Mans Cedex F-72085, France, olivier.dazel@univ-lemans.fr) and Gwenael Gabard (ISVR, Univ. of Southampton, Southampton, United Kingdom)

In this work, we are interested in the development of a Discontinuous Galerkin Method (DGM) for sound absorbing materials. These materials are commonly used for noise and vibration control. The objective of this method is to discretize the structure and to represent in each element the field as a superposition of local solutions such as plane waves. This type of methods have shown their efficiency by requiring much smaller numbers of degrees of freedom compared to standard polynomial interpolations (i.e., FEM) especially when the frequency is increased. For poroelastic materials, the solutions are expressed in terms of Biot waves (two of them associated to compression waves and one corresponding to shear waves). The poroelastic problem is expressed as a first order model and the formulation of numerical flux at interfaces between elements is derived and implemented. Compared to classical DG methods for standard acoustics, this method is applied here to a dissipative medium and to a two-displacement field involving shear and compression waves. Several two-dimensional cases will be presented in order to validate the method and to compare against analytical and finite-element solutions (in displacement and mixed formulation). Results will be discussed in terms of accuracy of the method, errors, and conditioning of the linear systems.

10:20

1aSA5. Numerical simulation of acoustic waves in air and poroelastic media using the partition of unity finite element method. Jean-Daniel Chazot (Roberval UMR7337, Universite de Technologie de Compiegne, Rue Personne de Roberval, Compiegne 60200, France, jean-daniel.chazot@utc.fr), Benoit Nennig (LISMMA, SUPMECA Paris, Saint-Ouen, France), and Emmanuel Perrey-Debain (Roberval UMR7337, Universite de Technologie de Compiegne, Compiegne, France)

Foams and fibrous materials are used in a large range of applications such as automotive or building acoustics. Their properties can be described with either a poroelastic model or an equivalent fluid model. These models, also used in geophysics, are now widely spread and are also available in some commercial finite element software applications. However, the discretization level required to achieve reasonable accuracy is not always acceptable in the mid-frequency range. In such case, the Partition of Unity Finite Element Method (PUFEM) using plane wave functions seems appropriate to avoid this limitation. The PUFEM has been recently applied to rigid frame materials with and without coupling with an acoustic cavity. It has also been applied efficiently to poroelastic materials. The present work focuses on the coupling between a poroelastic material, i.e., described with Biot's equations, and an air cavity. Some practical examples are tested to demonstrate the efficiency of the PUFEM for solving noise control problems at medium frequency, but also to underline the precautions that must be taken when dealing with an air-porous interface.

10:40

1aSA6. On internal mean flow in porous absorbers and its effect on attenuation properties. Susann Boij (Marcus Wallenberg Lab. for Sound and Vib. Res., Dept. of Aeronautical and Vehicle Eng., KTH Royal Inst. of Technol., 2004 Yolo Ave., Berkeley, California 94707, sboij@kth.se), Anna Färm (Scania CV AB, Stockholm, Sweden), Peter Göransson (Marcus Wallenberg Lab. for Sound and Vib. Res., Dept. of Aeronautical and Vehicle Eng., KTH Royal Inst. of Technol., Stockholm, Sweden), and Ragnar Glav (Scania CV AB, Stockholm, Sweden)

In vehicle applications, absorbing materials are often used to attenuate sound. In, for example, exhaust systems and on noise encapsulations, the absorber is exposed to flow. This creates a boundary layer above the absorber, which affects the impedance of the surface, and hence alters the absorption properties. In addition to this effect, the flow itself may enter the absorbent material due to high pressure and forced flow paths. An investigation of the effects that internal flow in the absorber imposes on the acoustic properties is presented. One way to describe the effect is by a change in flow resistivity. The effect is investigated for typical absorbers used in noise encapsulation for trucks. The Transfer Matrix Method is applied to calculate the resulting absorption and reflection coefficient for absorbers with changed flow resistivity in layers at the surface. The possibility to model the changed properties of the absorber with internal mean flow by means of Biot theory is also explored, together with a discussion on suitable experimental methods to verify and further investigate the effects.

11:00

1aSA7. Investigating the transmission loss effect via optimizing the insulator package on vehicle's firewall. Sajjad Beigmoradi (Automotive Eng. Dept., Iran Univ. of Sci. & Technol., No. 13, Emmami Alley, Golzarand Alley, Safdari St. Navab Safavi St, Tehran, Iran, s.beigmorady@gmail.com), Kambiz Jahani, and Hassan Hajabdollahi (Mech. Eng. Dept., Iran Univ.of Sci. & Technol., Tehran, Islamic Republic of Iran)

Nowadays, noise and vibration of attributes of motor vehicles have a dominant effect on customers' judgment about the cars, and hence, car manufacturers and OEM suppliers have dedicated remarkable time, budget, and concern to the investigations in this field. From NVH perspective, firewall is the foremost structural member in vehicle body design, since it is the main pass for transferring the engine induced noise to the passenger cabin. In this research, effect of the insulator package is studied through different configurations considering the radiated noise level and the optimized design is proposed using an optimization procedure. Indeed, it is concluded that adding the optimized insulator package can significantly refine the noise transmissibility while avoiding structural modifications in the firewall.

11:20

1aSA8. A method for measuring the acoustic properties of a porous sample mounted in a rigid ring in acoustic tubes. Thomas Dupont, Philippe Leclaire (Drive, ISAT, Université de Bourgogne, BP 31 - 49 rue Mlle Bourgeois, Nevers 58027, France, thomas.dupont@u-bourgogne.fr), Raymond Panneton, Kévin Verdière (Gaus, Université de Sherbrooke, Sherbrooke, QC, Canada), and Saïd Elkoun (Gaus, Université de Sherbrooke, Sherbrooke, Alberta, Canada)

This study presents a method to measure the acoustic properties of a homogeneous porous material with a support or a reduction element in an acoustic tube. Some materials tested have a lateral size much smaller than the tube's diameter, as they cannot be produced in the correct dimensions without corrupting the material; this also permits the testing of the same samples in a large frequency bandwidth

by using different section tubes. Moreover, the acoustic leaks on the material boundaries can significantly change the transmission loss measured in tubes. To rectify these problems, rings can be placed on each material surface. The presence of these rings can influence the acoustic indicator measurement; while this effect is negligible for tubes with a large cross section, it is not for tubes with a small cross section. To correct, or remove, the influence of the rings, we propose to use an application of the parallel assembly process of the transfer matrix method, which has recently been proposed by Panneton *et al*. [*Proceeding Internoise*, New York (2012)]. Measurements on classical porous materials with and without reductions are proposed and compared to simulated results. The ring's effects and the proposed corrections are discussed for different materials.

11:40

1aSA9. Acoustic characterization of graded porous materials under the rigid frame approximation. Jean-Philippe Groby, Olivier Dazel (Laboratoire d'Acoustique de l'Universit, Le Mans, France), Laurent De Ryck (LMS Int., Leuven, Belgium), Amir Khan, and Kirill Horoshenkov (School of Eng., Univ. of Bradford, Great Horton Rd., Bradford BD7 1DP, United Kingdom, a.khan117@bradford.ac.uk)

Graded porous materials are of growing interest because of their ability to improve the impedance matching between air and material itself. Theoretical models have been developed to predict the acoustical properties of these media. Traditionally, graded materials have been manufactured by stacking a discrete number of homogeneous porous layers with different pore microstructure. More recently, a novel foaming process for the manufacturing of porous materials with continuous pore stratification has been developed. This paper reports on the application of the numerical procedure proposed by De Ryck to invert the parameters of the pore size distribution from the impedance tube measurements for materials with continuously stratified pore microstructure. Specifically, this reconstruction procedure has been successfully applied to retrieve the flow resistivity and tortuosity profiles of graded porous materials manufactured with the method proposed by Mahasaranon *et al*. In this work, the porosity and standard deviation in pore size are assumed constant and measured using methods, which are applied routinely for homogeneous materials characterization. The numerical method is based on the wave splitting together with the transmission Green's functions approach, yielding an analytical expression of the objective function in the least-square sense. The objective function is constructed to minimize the discrepancy between the predicted and measured reflection coefficient spectra.

MONDAY MORNING, 3 JUNE 2013 — 515ABC, 8:55 A.M. TO 11:40 A.M.

Session 1aSCa

Speech Communication: Distinguishing Between Science and Pseudoscience in Forensic Acoustics I

Geoffrey Stewart Morrison, Cochair
Forensic Voice Comparison Lab., School of Elec. Eng. & Telecommunications, Univ. of New South Wales, NSW 2052, Australia

James Harnsberger, Cochair
Univ. of Florida, 402 NW 24th St., Gainesville, FL 32607

Chair's Introduction—8:55

Invited Papers

9:00

1aSCa1. Distinguishing between science and pseudoscience in forensic acoustics. Geoffrey Stewart Morrison (Forensic Voice Comparison Lab., School of Elec. Eng. & Telecommunications, Univ. of New South Wales,NSW 2052, Australia, geoff-morrison@forensic-voice-comparison.net)

In this presentation, I argue that one should not attempt to directly assess whether a forensic analysis technique is scientifically acceptable. Rather one should first specify what one considers to be appropriate scientific principles governing acceptable practice, then consider any particular approach in light of those principles. I focus on one principle: The validity and reliability of an approach should be empirically tested under conditions reflecting those of the case under investigation using test data taken from the relevant population. Versions of this principle have been key elements in several reports on forensic science, including forensic voice comparison, published over the last four-and-a-half decades. I consider the aural-spectrographic approach to forensic voice comparison (also known as "voiceprint" or "voicegram" examination) in light of this principle, and also the currently widely practiced auditory-acoustic-phonetic approach (these two approaches do not appear to be mutually exclusive). Finally, I challenge the audience members to consider what each of them thinks constitutes the relevant scientific principles regarding acceptable practice, and then consider their own approach to forensic-acoustic analysis in light of those principles.

9:20

1aSCa2. A Canadian perspective on forensic science versus pseudoscience. Brent Ostrum (Sci. and Eng. Directorate, Canada Border Services Agency, 14 Colonnade Rd., Ste. 280, Ottawa, ON K2E 7M6, Canada, brent.ostrum@cbsa-asfc.gc.ca)

This presentation will provide my personal observations regarding forensic science versus pseudoscience in the Canadian legal system. I am neither a lawyer nor a judge; rather, I am a forensic scientist with over 25 years of experience in the Canadian system. My presentation focuses on relevant criteria for expert evidence considered in Canadian courts. The key ruling in R. v. Mohan (1994) provides

the start of the discussion with subsequent court rulings adding various elements. In Canada, we have had several judicial inquiries, such as the Kaufmann Commission, that can serve to guide experts. Select aspects of the 2009 NAS report "Strengthening Forensic Science in the United States: A Path Forward" will also be referenced. There are some common "criteria" often used by courts in different jurisdictions to assess expert evidence, including forensic acoustics. In other words, some basic expectations for all forms of expert evidence can be identified. I will attempt to show how select "sciences" have tried to fulfill those expectations. This will involve some commentary on issues of individual examiner competency, oversight at a system level (e.g., accreditation), and the need for proper and adequate method validation.

9:40–10:00 Break

10:00

1aSCa3. Voice stress analyses: Science and pseudoscience. Francisco Lacerda (Dept. of Linguist., Stockholm Univ., Universitetsvagen 10 C, Stockholm SE-106 91, Sweden, frasse@ling.su.se)

Voice stress analyses could be relevant tools to detect deception in many forensic and security contexts. However, today's commercial voice-based lie-detectors are not supported by convincing scientific evidence. In addition to the scientific implausibility of their working principles, the experimental evidence invoked by the sellers is either anecdotal or drawn from methodologically flawed experiments. Nevertheless, criminal investigators, authorities, and even some academics appear to be persuaded by the ungrounded claims of the aggressive propaganda from sellers of voice stress analysis gadgets, perhaps further enhanced by the portrays of "cutting-edge voice-analysis technology" in the entertainment industry. Clearly, because there is a serious threat to public justice and security if authorities adopt a naïve "open-minded" attitude toward sham lie-detection devices, this presentation will attempt to draw attention to plausibility and validity issues in connection with the claimed working principles of two commercial voice stress analyzers. The working principles will be discussed from a phonetics and speech analysis perspective and the processes that may lead naïve observers into interpreting as meaningful the spurious results generated by such commercial devices will be examined. Finally, the scope and limitations of using scientific phonetic analyses of voice to detect deception for forensic purposes will be discussed.

10:20

1aSCa4. Assessing acoustic features in the speech of asylum seekers. Judith K. Rosenhouse (Linguist. Unit, SWANTECH Ltd., 89 Hagalil St., Haifa 32684, Israel, swantech@013.net)

One of the areas of forensic linguistics concerns asylum seekers who speak languages which are foreign to the official language of the country where they apply for asylum. Identifying and verifying their real national background may be difficult if their speech manner reveals non-typical properties of their (real or alleged) native languages. Governments submit such asylum seekers' speech samples for linguistic analysis on various levels, including phonetic acoustics. This aspect of forensic linguistics raises questions about the scientific merit of such an analysis. Our aim is to examine some of the questions which relate to segmental and supra-segmental features that are analyzed acoustically based on recorded samples of asylum seekers' (alleged) native language and compared with the same features as known from the literature. We demonstrate such issues by examples from the speech of Arabic-speaking asylum seekers whose native tongue is (supposed to be) some local dialect but the recording includes various foreign features reflecting different dialects or languages. These questions involve sociolinguistic factors that affect individual speakers' speech production due to a complex and unstable life-history. We suggest that the acoustic methods currently used in speech analysis in this context could be considered pseudo-science in many cases.

10:40

1aSCa5. Analysis criteria for forensic musicology. Durand R. Begault, Heather D. Heise, and Christopher A. Peltier (Audio Forensic Ctr., Charles M. Salter Associates, Inc., 130 Sutter St., Floor 5, San Francisco, CA 94104, durand.begault@cmsalter.com)

Expert testimony for forensic musicology addresses a broad spectrum of legal issues, including the authentication and differentiation of published compositions and musical recordings, performance rights, and legal determinations regarding copyright infringement. While legal cases involving music and performance infringement date back as far as the 19th century, the field of forensic musicology has no stated methodology by which an objective forensic determination can be made. Expert opinions based merely on subjective impression or resulting from the "golden ear" syndrome are pseudo-scientific and not objectively based. This paper proposes scientific methods and recommendations for analysis based on stated criteria, with the goal of controlling examiner bias. Considerations include analyses of composition, performance, and acoustical features, and factors such as melody, harmony, rhythm, and orchestration; pitch, tone, vibrato, and embellishment; metadata analysis; recording technologies; and digital signal processing, including "effects." By engaging in a series of structured categorizations, the forensic expert can establish a consistent, replicable, and objectively verifiable means of determining whether or not a recorded piece of music has been misappropriated.

11:00–11:40 Panel Discussion

Session 1aSCb

Speech Communication: Digital Speech Processing (Poster Session)

Mark VanDam, Chair
Washington State Univ., P.O. Box 1495, Spokane, WA 99202

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

1aSCb1. Precision and error of automatic speech recognition. Mark VanDam (Speech and Hearing Sci., Washington State Univ., P.O. BOX 1495, Spokane, WA 99202, mark.vandam@wsu.edu) and Noah H. Silbert (Ctr. for the Adv. Study of Lang., Univ. of Maryland, College Park, MD)

Automatic speech recognition (ASR) software developed by the LENA Research Foundation (Boulder, CO) is an increasingly important tool in psycholinguistics. Naturalistic day-long recordings are segmented and assigned talker labels including those for *KEY CHILD, ADULT MALE*, and *ADULT FEMALE*. Performance of the system is a serious concern for ASR in general, not just the LENA system. Additional evidence of the software's performance is necessary to better interpret and understand accumulating research using this tool. Here we analyze the correspondence between computer and human segment labels corresponding to children, mothers, and fathers. Segments machine-labeled as *ADULT MALE, ADULT FEMALE*, and *KEY CHILD* were played to judges who identified each segment as *Mother, Father, Child*, or *Other*. Judges' responses were analyzed in terms of agreement, precision, and error. Overall agreement between machine and human coding was in the 70% range with Cohen's $\kappa > .55$. Machine coding appears to be better at coding *KEY CHILD* segments than the *ADULT* segments, and agreements for *ADULT MALE* labels were better than for *ADULT FEMALE* labels. The ASR system performed similarly when assigning segment labels for children and fathers, but less well for mothers. Overall error rates were generally very low. [Work support from NIH-NIDCD R01DC009560.]

1aSCb2. Unsupervised machine learning for the accurate classification of the discourse maker *like* in code-switching utterances. Page E. Piccinini (Linguistics, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0108, ppiccinini@ucsd.edu) and Eric R. Kramer (Medical Scientist Training Program, Univ. of California, San Diego, La Jolla, CA)

Spanish-English bilinguals use the discourse marker *like* in English, Spanish, and code-switching utterances. An acoustic analysis found that the [l] and diphthong in *like* is produced differently depending on the type of utterance in which it occurs. To investigate the possible perceptual relevance of these differences, we built a logistic-polynomial regression model to classify *like* tokens based on acoustic data. The model first projects F1 and F2 values onto a space of time-dependent polynomials. We then apply multinomial logistic regression to classify these polynomials as English, Spanish, or code-switching. The area under the curve was 0.75, showing classification was significantly greater than random. This model outperforms models that rely on static values for F1 and F2, either at the midpoint of the token (DeLong's test, $p < 0.004$), or two data points: the midpoint of the [a] of the diphthong, and the midpoint of the [ɪ] of the diphthong (DeLong's test, $p < 0.04$). The superiority of the polynomial model suggests that the time-dependent progression of F1 and F2 values, rather than absolute formant values, is important for predicting an imminent code-switch. We hypothesize that listeners leverage these time-dependent changes in F1 and F2 to anticipate code-switches.

1aSCb3. Using a computational model for the auditory midbrain to explore the neural representation of vowels. Laurel H. Carney, Jiashu Li, Tianhao Li (Biomedical Eng. and Neurobiol. & Anatomy, Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu), and Joyce M. McDonough (Linguistics, Univ. of Rochester, Rochester, NY)

A formant-based approach to representing vowel quality is anchored in acoustic theory and is well documented in perception studies and in auditory modeling. This ongoing study investigates the representation of vowels in the responses of auditory models at the level of the midbrain (inferior colliculus). Previous modeling and physiological results have shown that formant structure is conveyed by changes in neural rates of midbrain cells that are tuned to amplitude modulations near voice pitch frequency. The current study examined model population responses to 20 speakers (10 males, 10 females) reciting 12 English vowel contrasts from the Hillenbrand *et al*. database [*J. Acoust. Sci. Am.* 97, 3099 (1995)]. Pairwise correlations across model population responses for each vowel were used to evaluate variability in the neural representations. Results show that the acoustical variability associated with the vowel contrasts is maintained in these neural representations. Thus, variability in the acoustic vowel space is maintained after the nonlinear responses of realistic auditory-nerve models and midbrain models for amplitude modulation tuning. Our goal is to extend our knowledge of the neural representation of the vowel space using a computational model for the responses of auditory neurons to ensembles of speech tokens.

1aSCb4. Generative approach for robust acoustic model training for blindly separated speech recognition. Norihide Kitaoka, Yuto Dekiura, and Kazuya Takeda (Dept. of Media Sci., Nagoya Univ., Furo-cho Chikusa-ku, Nagoya 464-8603, Japan, kitaoka@nagoya-u.jp)

We propose a generative acoustic model training method for robust speech recognition with blind sound source separation as a front-end. Multiple microphone systems are often used for the separation. In such situation, separated speech is severely distorted and thus the recognition rate significantly drops. If we can measure transmission characteristics from the sound sources with various directions to the microphones, we can simulate to receive various mixed speech made by multiple speakers speaking with overlaps to each other. Then we separate the simulated overlapped speech using a blind source separation method such as frequency domain independent component analysis (FDICA) and use the separated speech to train HMM acoustic models to recognize such separated speech. Our method can generate such distorted speech enormously without recording the real speech spoken to the microphone system. We evaluate the models in the continuous Japanese speech recognition and show the effectiveness.

1aSCb5. Speaker identification in reverberant environments. Noha Korany (Elec. Eng., E. E. Dept., Faculty of Eng., Alexandria Univ., Alexandria 21544, Egypt, nokorany@hotmail.com)

The performance of speaker identification process degrades in reverberant environments, as reverberation leads to clear physical effects on the perceived signals. This paper investigates the effect of room reverberation on

the identification rate. However, various reverberant environments are simulated, and the impulse response is convolved with dry speech signals. The reverberant speech database is used by the identification engine within the train and the test phases. Then, statistical identification technique using the Gaussian Mixture Model (GMM) is implemented. Three types of features, Mel-Frequency Cepstrum Coefficients (MFCC), Perceptual Linear Predictive Cepstrum Coefficients (PLPCC), and Relative Spectral Perceptual Linear Predictive Cepstrum Coefficients (RASTA-PLPCC) are extracted. Various types of features are integrated and used for the classification problem. Finally, the performance of the recognition process is evaluated while varying the duration of the train and the test signals, the features used for the classification problem, and the room reverberation. A series of physical measures that correlate with various attributes of the sound perceived in rooms, such as the reverberation time T60, the clarity index C80, the definition D, are calculated. Then, their effect on the identification rate is investigated.

1aSCb6. Experimental study of shout detection with the rahmonic structure. Naoto Kakino, Takahiro Fukumori, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is012085@ed.ritsumei.ac.jp)

The surveillance systems with microphones have been developed to achieve a secure society. These systems can detect hazardous situations with observed speech but are generally very expensive. This is because the conventional systems manually detect hazardous situations by security officers. Thus, we focus on the automatic shout detection method, which can estimate hazardous situations. The acoustic model based on the Gaussian mixture model has been proposed as the conventional method to identify shouted and natural speeches. However, these methods have a problem that it is necessary to prepare huge training samples to accurately detect shouted speech. In the present paper, we focus on the rahmonic structure, which shows a subharmonic of fundamental frequency in the cepstrum domain because the rahmonic structure tends to arise in the shouted speech. In the present paper, we therefore propose the detection method of the shouted speech based on rahmonic structure. More specifically, we investigate rahmonic structure in the shouted speeches, and detect the shouted speech by utilizing the rahmonic structure model. We conducted evaluation experiments to confirm the effectiveness of the proposed method. As a result of evaluation experiments, we confirmed the effectiveness of the proposed method.

1aSCb7. Suppression of clipping noise in observed speech based on spectral compensation with Gaussian mixture models and reference of clean speech. Makoto Hayakawa, Takahiro Fukumori, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is033080@ed.ritsumei.ac.jp)

In recent years, the development of communication system allows people to easily record and distribute their speech. However, in the speech recording, clipping noise degrades sound quality when the level of input signal is excessive for the maximum range of amplifier. In this case, it is necessary to suppress clipping noise in the observed speech for improving its sound quality. Although a linear prediction method has been conventionally proposed for suppressing clipping noise, it has a problem with degradation of the restoration performance by cumulating error when the speech includes a large amount of clipping noise. This paper describes a method for suppression of clipping noise in observed speech based on spectral compensation. In this method, the power spectral envelope of speech on each frame in the lower frequency band is noise suppressed to by using Gaussian Mixture Models (GMM), and the one in the higher frequency band is restored by referring to the clean speech. We carried out evaluation experiments with a speech quality, and confirmed the effectiveness of the proposed method toward the speech, which includes a large amount of clipping noise.

1aSCb8. Detection for Lombard speech with second-order mel-frequency cepstral coefficient and spectral envelope in beginning of talking-speech. Takayuki Furoh, Takahiro Fukumori, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is0038sv@ed.ritsumei.ac.jp)

In noisy environments, the recorded speech is distorted by the additional noise and the Lombard effect. Thus, the automatic speech recognition (ASR) performance is degraded in noisy environments. To solve this problem, noise reduction methods have been proposed as the conventional study. However, in the conventional study, the improvement of ASR performance for the Lombard effect was not discussed well enough. In the present paper, we focus on the robustly detection for Lombard effect speech (Lombard speech). This is because the ASR system can employ a suitable acoustic model by detecting the Lombard speech. We previously proposed the detection for Lombard speech based on second-order MFCC and fundamental frequency. The previously proposed method however requires longer utterances to detect Lombard speech. We therefore newly propose the detection method for Lombard speech with second-order MFCC and spectral envelope in beginning of talking-speech. To detect the Lombard speech at a short time, the proposed method employs variable weights corresponding to elapsed time for second-order MFCC and spectral envelope. As a result of evaluation experiments, we confirmed that the detection time was reduced from the conventional method.

1aSCb9. The detection of the sleepiness from the sounds obtained inside of the body. Masanoti Akita, Hiroyuki Kamitabira, Tomohiko Yoshida, Syogo Kanemitsu, and Yoichi Midorikawa (Faculty of Eng., Dept. of Elec. and Electron. Eng., Oita Univ., 700 Dannoharu, Oita 8701192, Japan, makita@oita-u.ac.jp)

This paper shows that the detecting method of the sleep-in sleep state or sleepiness from the sound signals in the human body. In former report, we showed that the detection of the sleepiness concerns with the piezoelectric sensors attached on car seat. Our preliminary examination showed that the spectrum of signals from the piezoelectric sensor have the tendency that the shapes of the spectral envelopes are flattened. And the sounds in human body are considered to have similar features. In this experiment, the signals of the piezoelectric sensor on the seat and the sounds in the human body are measured at the same time and the relation between the sounds and sleepiness are examined. The sounds inside the body are measured using NAM microphone system. The spectral envelopes of the signals from the left side and the right side of breath are calculated. The spectral envelopes from the seat are calculated at the same time. Twenty-two measurements by four examinees are done, and 8 sleeping data are measured. Flatness of the envelopes is defined using the lower order of cepstral coefficients, and the increase of the flattened spectrum is observed by the sounds from the sleeping data of three quarters.

1aSCb10. Modeling occurrence tendency of adventitious sounds and noises for detection of abnormal lung sounds. Takanori Okubo, Masaru Yamashita, Katsuya Yamauchi, and Shoichi Matsunaga (Engineering, Nagasaki Univ., 1-14, Bukyo-machi, Nagasaki 852-8521, Japan, mat@cis.nagasaki-u.ac.jp)

Diagnosis of pulmonary emphysema by using a stethoscope is based on the common knowledge that abnormal respiratory (adventitious) sounds usually appear in patients with pulmonary emphysema. However, the spectral similarity between adventitious sounds and noises at auscultation makes highly accurate automatic detection of adventitious sounds difficult. In this paper, we have proposed a novel method for distinguishing between normal lung sounds in healthy subjects and abnormal sounds, including adventitious sounds in patients, taking into account the occurrence tendency of adventitious sounds and noises. According to our investigation results, adventitious sounds occur repeatedly in successive inspiratory/expiratory phases of patients. On the other hand, noise sounds mix at random in lung sounds of both patients and healthy subjects. In our method, the occurrence tendency of these sounds is described using Gaussian distribution of a random variable obtained by subtracting the acoustic likelihood for abnormal respiration from the likelihood for normal respiration. The spectral likelihood calculated using hidden Markov models and the validity score of the occurrence tendency of the adventitious/noise sounds are combined to derive the classification result. Our method achieved a higher classification rate of 94.1% between normal and abnormal lung sounds than that achieved using the conventional method (87.4%).

1aSCb11. Audio quality evaluation by experienced and inexperienced listeners. Nadja Schinkel-Bielefeld (Audio Dept., Fraunhofer Inst. IIS, Am Wolfsmantel 33, Erlangen 91058, Germany, nadja.schinkel-bielefeld@iis.fhg.de), Netaya Lotze (Deutsches Seminar, Leibniz Universität Hannover, Hannover, Germany), and Frederik Nagel (Audio Dept., Fraunhofer Inst. IIS, Erlangen, Germany)

Basic audio quality of coded audio material is commonly evaluated using ITU-R BS-1534 Multi Stimulus with Hidden Reference and Anchors (MUSHRA) listening test. MUSHRA guidelines call for experienced

listeners. However, the majority of consumers using the final product are no expert-listeners. Also the degree of expertise in a listening test may vary among listeners in the same laboratory. It would be useful to know how the audio quality evaluation differs between trained and untrained listeners and how training and actual tests should be designed in order to be as reliable as possible. To investigate the rating differences between experts and non-experts, we performed MUSHRA listening tests with 13 experienced and 11 inexperienced listeners using 5 speech and audio codecs delivering a wide range of basic audio quality. Except for the hidden reference, absolute ratings of non-experts were consistently at least 10% higher than those of experts. However, they could be mapped to each other by a z-transform. For lower quality values, confidence intervals were significantly larger for non-experts than for experts. Experienced listeners set more than twice as many loops as non-experts, compared more often between codecs and listened to high quality codecs for a longer duration than non-experts.

1aSCb12. Influence of amplification scheme and number of channels on aided speech-intelligibility performance. Amyn M. Amlani (Dept. of Speech and Hearing Sci., Univ. of North Texas, 907 W Sycamore St., P.O. Box 305010, Denton, TX 76203, amlaniam@unt.edu), Sneha V. Bharadwaj (Dept. of Commun. Sci. and Disord., Texas Woman's Univ., Denton, TX), and Shirin J. Jivani (Dept. of Speech and Hearing Sci., Univ. of North Texas, Denton, TX)

Modern hearing aids offer a wide range of channels (i.e., filters) and amplification schemes. Our previous work revealed that increasing the number of channels, in conjunction with a fast-fast amplification scheme, results in (a) the spectral flattening of the vowels /i, u, ^/ (Amlani *et al*., 2011), and (b) reduced consonant- and vowel-identification accuracy in impaired listeners (Amlani *et al*., 2012). In the present study, we assess the performance of impaired listeners and their normal-hearing controls on the perception of everyday speech using the Connected Speech Test (Cox *et al*., 1987, 1988). The stimuli were processed through a simulated hearing aid with varying amplification schemes (linear, compression [fast-fast, slow-slow, fast-slow]) and number of channels (2, 8, 16). Findings revealed that while speech-intelligibility performance improved markedly with everyday speech compared to /CVC/ words for both groups, normal-hearing listeners identified the target words significantly better than impaired listeners did. Speech-intelligibility performance was similar across number of channels for normal-hearing listeners, but decreased significantly with a fast-fast amplification scheme. For impaired listeners, performance declined for channels greater than 2 and with the inclusion of the fast-fast amplification scheme. We discuss the implication of these findings relative to clinical application and hearing aid design.

1aSCb13. Relationship between subjective and objective evaluation of noise-reduced speech with various widths of temporal windows. Mitsunori Mizumachi (Dept. of Elec. Eng. and Electron., Kyushu Inst. of Technol., 1-1 Sensui-cho, Tobata-ku, Kitakyushu, 805-8440, Japan, mizumach@ecs.kyutech.ac.jp)

It is necessary to enhance adverse speech signals for building useful speech interfaces. Speech enhancement is essential under noisy and reverberant acoustic environments. Therefore, quality assessment of the enhanced speech signals should be also an important issue in noise reduction and dereverberation. Subjective evaluation is given by carrying out listening tests, and objective evaluation is provided by speech distortion measures. However, there is the discrepancy between subjective and objective evaluation of speech distortion. The author has investigated the relationship between subjective and objective evaluation of noise-reduced speech signals. The objective speech distortion was calculated in each short-term frame, of which length was fixed, and the statistical characteristics of the short-term speech distortion were investigated using higher-order statistics such as skewness and kurtosis. The preliminary result suggested that skewness of the short-term speech distortion could give an explanation for the discrepancy between subjective and objective evaluation. Further investigation of the relationship between subjective and objective evaluation of noise-reduced speech signals is carried out with a variety of temporal window widths. [Work supported by NEDO, Japan.]

1aSCb14. A Hidden Markov Model based speaker identification system using mobile phone database of North Atlantic Treaty Organization words. Shyam S. Agrawal, Shweta Bansal, and Dipti Pandey (KIIT College of Eng., Sohna Rd.,Near Bhondsi, Gurgaon, Haryana 122102, India, dr.shyamsagrawal@gmail.com)

This paper describes results of an experiment to conduct text independent speaker identification of large number of speakers (about 100) using a standard vocabulary of about 23 NATO words—such as Alfa, Bravo, etc. These words in isolation were spoken in a sound treated room by Hindi natives having very good education in English (both males and females) and recorded by a three channel data recording system—the cardioid microphone, electret condenser microphone, and a NOKIA mobile telephone. The pre-processed digitized database of isolated words was further processed to determine 39 MFCC's and their derivatives and used to build an HMM model for each speaker based on all the words. The HMM model was trained using an HTK tool kit to generate the model parameters and tested using Viterbi algorithm. The identification of speakers was done in a closed set manner, based on comparison of each NATO word in the model. In addition to correct identification, false acceptance and false rejection scores were also found. The results show varying performance due to variations in channels, male/female speakers. The overall identification scores vary between 60% and 70%. The paper gives detailed analysis of results.

1aSCb15. A time-synchronous histogram equalization for noise robust speech recognition. Fumiya Takahashi, Masaharu Kato, and Tetsuo Kosaka (Grad. School of Sci. and Eng., Yamagata Univ., 4-3-16 Jonan, Yonezawa, Yamagata 992-8510, Japan, tar11506@st.yamagata-u.ac.jp)

The histogram equation (HEQ) technique is commonly adopted for feature space normalization in speech recognition systems. In this technique, a transform function is calculated directly from the histograms of both training and test data, and the nonlinear effects of additive noise are compensated. In order to estimate the transform function accurately, a certain amount of data are required. However, this is not suitable for real-time application because at least several seconds of evaluation data need to be accumulated before the transform function can be calculated. This means that the system cannot start the recognition process until the end of utterance. In this research, we aim to develop a new speech recognition method based on the HEQ technique for real-time processing. This method is called "time-synchronous frame-weighted HEQ (ts-FHEQ)." In the time-synchronous decoding, lack of data for estimating the histogram becomes a major problem. To resolve this problem, we introduce a frame weighting approach, where the degree of transform is controlled according to the number of data frames. Our speech recognition experiments verified that the proposed technique shows good performance and achieves substantial reduction of calculation time.

1aSCb16. An investigation of vowel substitution rules in the automatic evaluation system of English pronunciation. Kei Sato, Masaharu Kato, and Tetsuo Kosaka (Grad. School of Sci. and Eng., Yamagata Univ., 4-3-16 Jonan, Yonezawa, Yamagata 992-8510, Japan, tna01014@st.yamagata-u.ac.jp)

We investigate the performance improvement of an automatic evaluation system of the English pronunciation of Japanese learners. In this system, Japanese and English acoustic models are used to detect mispronunciation at a phoneme level. Hidden Markov models (HMMs) are used as acoustic models. Mispronunciation is detected by comparing the output likelihoods of the two models. In order to improve the performance of this system, we investigate certain mispronunciation rules, which represent common mispronunciations among Japanese learners. We use four mispronunciation rules: vowel insertion (at the end of a word), vowel substitution, vowel insertion (between consonants), and consonant substitution. In this system, the accuracy of the mispronunciation rules is particularly important. The rules are determined on the basis of the knowledge of phonetics in our previous system. However, the effectiveness of the rules has not been analyzed quantitatively, and we do so in this work. A knockout procedure is used to select effective rules. By selecting effective rules, we found that the correlation coefficient between the subjective evaluation value and the system performance improved from 0.757 to 0.858.

1aSCb17. A method to estimate a temporally stable spectral envelope for periodic signals. Masanori Morise and Yoichi Yamashita (Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu, Shiga 525-8577, Japan, masanori.morise@gmail.com)

Vocoder-based speech synthesis system requires fundamental frequency (F0) and spectral envelope. Since the sound quality of synthesized speech depends on the estimation performance, methods that can accurately estimate two parameters are crucial to synthesize natural speech. In particular, spectral envelope estimation is more difficult than F0 estimation. One of the problems in spectral envelope estimation is that the result depends on the windowing time, type of window function, and its length. Conventional methods such as LPC or Cepstrum cannot remove the temporal variation. TANDEM-STRAIGHT can estimate a temporally stable spectral envelope, whereas the processing that consists of averaging two power spectra, smoothing and post-processing is complex. To simplify the processing, we propose a method based on pitch synchronous analysis and spectral smoothing. The proposed method can estimate a temporally stable spectral envelope from only one power spectrum processed by a specialized window function. The window function and its length are determined to remove the temporal variation. The objective evaluation was conducted to verify the temporal variance and the estimation performance. The result suggested that the proposed method can estimate the temporally stable power spectrum as well as TANDEM-STRAIGHT.

1aSCb18. Processing time improvement for speech enhancement based local projection using dynamic parameters. Phongphan Phienphanich, Charturong Tantibundhit (Elec. and Comput. Eng., Thammasat Univ., 99 Moo 18 Phaholyothin Rd., Ampher Khlongluang, Pathumthani 12120, Thailand, 5410030067@student.tu.ac.th), and Chutamanee Onsuwan (Linguistics, Thammasat Univ., Khlongluang, Pathumthani, Thailand)

Local projection (LP) has been widely used to enhance speech by transforming noisy speech into two orthogonal subspaces: noise (S1) and signal plus small amount of noise (S2) subspace. S1 is removed and S2 is transformed into time domain resulting in the enhanced speech. Satisfactory results with significantly improved speech quality have been reported by several works although the processing time was not taken into account. Four parameters to be considered are embedding dimension (d), time delay (τ), numbers of iteration, and minimal embedding dimension. Speech quality is increased by the increase of d-parameter, resulting in decrease of τ-parameter value ($d\times\tau$ kept constant) and the dramatic increase of the processing time. The goal is to come up with the best $d\times\tau$ parameter for each iteration, while speech quality remains almost unaffected. Rather than using a fixed $d\times\tau$ parameter, a dynamic approach is taken. Specifically, τ is initially set to 1 and incremented by 1 for next iteration. The experimental results tested on Thai initial rhyming words corrupted by three noise types (white, car, and street) each at SNR levels of 10, 5, 0, -5 dB showed that the proposed method significantly reduced the processing time for white noise by 35% ($p < 0.01$).

1aSCb19. Acoustic-to-articulatory inversion by analysis-by-synthesis using cepstral coefficients. Julie Busset and Yves Laprie (LORIA/CNRS, 615 rue du jardin botanique, Villers-lès-Nancy 54600, France, yves.laprie@loria.fr)

This paper deals with acoustic to articulatory inversion of speech by using an analysis by synthesis approach. We used old X-ray films of one speaker to (i) the develop a linear articulatory model presenting a small geometric mismatch with the subject's vocal tract mid sagittal images (ii) and design an adaptation procedure of cepstral vectors used as input data. The adaptation exploits the bilinear transform to warp the frequency scale in order to compensate for deviation between synthetic and natural speech. This enables the comparison of natural speech against synthetic speech without using cepstral liftering. A codebook is used to represent the forward articulatory to acoustic mapping, and we designed a loose matching algorithm using spectral peaks to access it. This algorithm, based on dynamic programming, allows some peaks in either synthetic spectra (stored in the codebook) or natural spectra (to be inverted) to be omitted. Quadratic programming is used to improve the acoustic proximity near each good candidate found during codebook exploration. The inversion has been tested on speech signals corresponding to the X-ray films. It achieves a very good geometric precision of 1.5 mm over the whole tongue shape unlike similar works evaluating the error at 3 or 4 points corresponding to sensors located at the front of the tongue.

1aSCb20. An overview of the development of resources, techniques, and, systems for Indian spoken languages. Shyam S. Agrawal (KIIT, College of Eng., Sohna Rd.,Near Bhondsi, Gurgaon, Haryana 122102, India, dr.shyamsagrawal@gmail.com)

India possesses a large variety of languages and dialects spoken in different parts of the country. These languages possess some unique linguistic, phonological, and phonetic properties different from European languages. Research is being done in several of Indian languages—such as Hindi, Bangla, etc. to study the articulatory, acoustic—phonetic and prosodic nature for the purpose of creating standards of phonetic representation of phonemes and Pronunciation Lexicon in Indian Languages. Comprehensive and task specific language corpora, speech databases in laboratory as well as in mobile communication situation and the tools/technics required for processing of speech signals are being developed. The emphasis is on developing multilingual human-machine interaction systems. Some of the recently developed systems include multi-lingual speech recognition system for voice enabled services, multilingual text to speech synthesis system, speaker and language identification system for general purpose and forensic applications. Recognition of emotions in spoken speech, spoken language translation system, etc. The paper presents an overview of such studies conducted in various laboratories, academic institutions, and industries in India pertaining to these areas. The technologies used for data collection, processing, and recognition/ synthesis, etc., utilized and status of the development have been mentioned.

1aSCb21. Performance estimation of speech recognition based on Perceptual Evaluation of Speech Quality and acoustic parameters under noisy and reverberant environments with Corpus and Environment for Noisy Speech RECognition 4. Takahiro Fukumori, Masato Nakayama, Takanobu Nishiura, and Yoichi Yamashita (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, cm013061@ed.ritsumei.ac.jp)

CENSREC-4 evaluation framework has been distributed for evaluating distant-talking speech under various noisy and reverberant environments. It however has not been evaluated how variable noisy and reverberant features in this contains. We thus try to evaluate CENSREC-4 with our designed noisy and reverberant criteria based on PESQ and acoustic parameters. We specifically focus on criteria to represent the difficulty of noisy and reverberant speech recognition, and also confirm why it is difficult to easily evaluate the recognition accuracy in a part of CENSREC-4 corpus with our proposed noisy and reverberant criteria. We first designed the noisy and reverberant criteria using the relationship among the D value, the PESQ, and the ASR performance. We then tried to estimate the recognition accuracy in various noisy and reverberant environments with CENSREC-4. We carried out evaluation experiments to confirm the difficulty to easily evaluate the recognition accuracy in a part of CENSREC-4 courpus0. As a result of evaluation experiments, we confirmed that it was difficult to estimate the accuracy of noisy and reverberant speech recognition in heavy noisy and reverberant environment with CENSREC-4. We therefore confirmed that CENSREC-4 contained very challenging and variable noisy and reverberant data.

1aSCb22. On instantaneous vocal tract length estimation from formant frequencies. Adam Lammert and Shrikanth Narayanan (Signal Anal. and Interpretation Lab., Univ. of Southern California, 3740 McClintock Ave., Rm. 400, Los Angeles, CA 90089, lammert@usc.edu)

The length of the vocal tract and its relationship with formant frequencies is examined at fine temporal scales with the goal of providing accurate estimates of vocal tract length from acoustics on a spectrum-by-spectrum basis despite unknown articulatory information. Accurate vocal tract length estimation is motivated by applications to speaker normalization and biometrics. Analyses presented are both theoretical and empirical. Various theoretical models are used to predict the behavior of vocal tract resonances in

the presence of different vocal tract lengths and constrictions. Real-time MRI with synchronized audio is also utilized for detailed measurements of vocal tract length and formant frequencies during running speech, facilitating the examination of short-time changes in vocal tract length and corresponding changes in formant frequencies, both within and across speakers. Previously proposed methods for estimating vocal tract length are placed within a coherent framework, and their effectiveness is evaluated and compared. A data-driven method for VTL estimation emerges as a natural extension of this framework, which is then developed and shown to empirically outperform previous methods on both synthetic and real speech data. A theoretical justification for the effectiveness of this new method is also explained. [Work supported by NIH.]

MONDAY MORNING, 3 JUNE 2013 — 510A, 9:00 A.M. TO 12:00 NOON

Session 1aSP

Signal Processing in Acoustics, Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Array Signal Processing for Three-Dimensional Audio Applications I

Yang Hann Kim, Cochair
Mech. Eng., KAIST, 373-1 Science Town, Daejon-shi 305-701, South Korea

Jung-Woo Choi, Cochair
Mech. Eng., KAIST, 373-1 Daehak-ro, Yusung-gu, Daejeon 305-701, South Korea

Invited Papers

9:00

1aSP1. Control of frame loudspeaker array by minimizing fluctuations of frequency response and synthesized wave front. Akio Ando (Sci. and Technol. Res. Lab., NHK, 1-10-11 Kinuta Setagaya, Tokyo 157-8510, Japan, andio@a.memail.jp) and Aya Tokioka (Dept. of Commun. and Integrated Syst., Grad. School of Sci. and Eng., Tokyo Inst. of Technol., Tokyo, Japan)

In sound reproduction with accompanying pictures, the localization of sound on the image display is problematic because a loudspeaker cannot be placed on the display. To gain a stable localization, the use of loudspeaker array set on the frame of the display may be a solution. In general, the loudspeaker array enables to control the perceptual depth of sound image by generating the appropriate curvature of the wave front corresponding to the source position. However, the frequency response and the shape of the wave front reproduced by such a frame loudspeaker array sometimes deteriorate, particularly when the virtual sound source has a certain distance back from the display. In this study, new parameters are introduced to scale the deterioration of the frequency response and the shape of the wave front. A new method is also introduced to generate the input signals to the loudspeakers on the basis of minimization of these parameters. The experimental result showed that the method generates the sound with small deterioration of the frequency response and the wave front regardless of the depth of the virtual source position, meaning that it can be used for the sound reproduction for 3D television.

9:20

1aSP2. Optimal beamformer designed for robustness against channel mismatch based on Monte Carlo Simulation. Mingsian R. Bai and Ching-Cheng Chen (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 30013, Taiwan, msbai63@gmail.com)

Design of beamformers that withstand mismatch in channel characteristics of gain, phase, and position has been a key issue in array signal processing. These mismatch factors are random in nature and generally intractable by deterministic approaches. This paper examines the effects of channel mismatch on beamformer performance from a statistical perspective. The aim of this work is twofold: analysis and synthesis. In the analysis phase, the mismatch factors of microphone characteristics are assumed to be random variables following either uniform or Gaussian distribution. In the light of the Monte-Carlo Simulation (MCS), statistics including the mean, maximum, minimum, and the probability density function (pdf) of Directivity Factor (DF) can be efficiently obtained with random sampling. This provides useful information for choosing performance measures in the next synthesis phase. Optimal parameters of superdirective array designed using least squares (LS) and convex optimization (CVX) are determined based on the preceding performance measures. Simulation results have shown that the proposed statistical approach with different performance measures provided various performance-robustness tradeoffs in terms of parameter range for optimal beamformers.

9:40

1aSP3. Hybrid immersive three-dimensional sound reproduction system with steerable parametric loudspeakers. Chuang Shi, Ee-Leng Tan, and Woon-Seng Gan (School of Elec. & Electron. Eng., Nanyang Technol. Univ., 50 Nanyang Ave., S2-B4a-03, DSP Lab, Singapore 639798, Singapore, shichuang@ntu.edu.sg)

A loudspeaker must be both dispersive and directive to accurately reproduce spatial audio from digital media. To address this problem, an audio system that has a unique combination of conventional and parametric loudspeakers has previously been proposed and proved to be effective to reproduce an immersive 3D soundscape. However, this system has two drawbacks: (1) There is only one fixed "sweet spot," and (2) only one listener within the "sweet spot" can enjoy the complete experience. Therefore, a hybrid 3D sound

reproduction system combining conventional loudspeakers with a pair of steerable parametric loudspeakers is proposed in this paper. By using this new combination of conventional and steerable parametric loudspeakers, the "sweet spot" can be steered toward the listener's head position. Thus, the listener no longer needs to keep his head stationary while watching movies or playing games, which resulting in a more relaxing and pleasant experience. Furthermore, a dual-beamsteering method is proposed for the parametric loudspeaker, which provides a flexible software-control solution to allow the 3D sound experience to be enjoyed by two listeners simultaneously. This paper provides the system overview and highlights the key processing techniques in rendering a "steerable" immersive 3D soundscape.

10:00

1aSP4. Virtual sound source generation: Its various methods and performances. Dong-Soo Kang (Ctr. for Noise and Vib. Control, Dept. of Mech. Eng., KAIST, Daejeon, South Korea), Jung-Min Lee (Grad. School of Culture Technol., KAIST, Daejeon, South Korea), Jung-Woo Choi, Min-Ho Song, and Yang-Hann Kim (Ctr. for Noise and Vib. Control, Dept. of Mech. Eng., KAIST, YuseongGu GuseongDong 373-1, Daejeon, South Korea, yanghannkim@kaist.ac.kr)

There are many means to generate a virtual sound source, or sources in the region of interest. For example, Wave Field Synthesis (WFS) or Higher Order Ambisonics (HOA) are good examples. These methods normally assume that loudspeakers are spatially distributed in the space. The region of interest where the desired sound is generated can have arbitrary shape; enclosed by surrounding loudspeakers or partially enclosed. Therefore, the performance of the method would be affected by the boundary conditions as well as the wave length of desired wave field. In other words, how the waves are distributed in the selected space. In recent work [Choi and Kim, IEEE Trans. Speech Audio Process. **20**(7), 1976–1989 (2012)], a new approach was proposed to generate virtual sources in the space that is enclosed by an array of loudspeakers, which have been believed to be problematic with well-known methods. It is proved to be mathematically exact solution. However, "exact solution" does not necessarily mean that it is better than the others. In this paper, performances of these three methods are compared. Theoretical and experimental comparisons have been attempted and observed in this paper.

10:20

1aSP5. Evaluation of system configuration to check the suitability for the sound field rendering using the inverse approach. Jeong-Guon Ih (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., 373-1 Guseong-Dong, Yuseong-Gu, Daejeon 305-701, South Korea, J.G.Ih@kaist.ac.kr), Wan-Ho Cho (Div. of Physical Metrol., Korea Res. Inst. of Standards and Sci., Daejeon, South Korea), and Seung-Wan Hong (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea)

Sound field control by the inverse approach based on the acoustical holography is useful to render an arbitrary target sound field within a selected control zone if the target condition is given in a detailed format. This method needs information on the various factors constituting the total system: source array configuration, relative position of source and control region, assigning method of target field condition, conditioning method, etc. Because these factors heavily affect the accuracy of the generated sound field, a proper definition of the problem including all factors related to the system configuration is important. In this work, we have studied on the condition of major factors of various configurations to generate the target sound field efficiently with high accuracy. Because the difference between target and generated sound field strongly depends on the noise and the information error existing in the actual situation, the expected accuracy should be calculated in relation to the characteristics of system transfer matrix. To this end, variances of uncorrelated noise, condition number, and linear independency of the transfer matrix are evaluated to check the suitability of transfer matrix for accurately rendering the sound field in both free-field and enclosed space.

10:40

1aSP6. Extension of perceived source width using loudspeaker array. Jung-Woo Choi and Yang-Hann Kim (Ctr. for Noise and Vib. Control, Dept. of Mech. Eng., KAIST, YuseongGu GuseongDong 373-1, Daejeon 373-1, South Korea, khepera@kaist.ac.kr)

In this paper, a method to extend the perceived spatial width of a virtual sound source using multiple loudspeakers is proposed. Control of perceived source width or apparent source width (ASW) has been attempted by decreasing the inter-aural correlation. For this purpose, numerous decorrelators were proposed for stereo loudspeakers or headphones. However, these techniques are inadequate for sound field reproduction system incorporating multiple loudspeakers. For sound field reproduction, extension of source width has to be realized with three requirements. First, extension should not deteriorate the localization cue, provided by the reproduction system. Second, the coloration artifact, which induces by extra wavefronts other than the direct wave, should be minimized. Most importantly, the effect of source width extension has to be maintained over a large listening area. To design a spatial decorrelator that can meet these requirements, we design a proper target sound field with reduced inter-aural correlation over a zone of interest. The target sound field is reproduced by loudspeakers driven from a multipole expansion technique. The performance of the proposed method is verified by examining the inter-aural correlation coefficient (IACC) of the reproduced sound field over a wide area, as well as the ITD and ILD distributions.

Contributed Papers

11:00

1aSP7. Sound-field reconstruction performance of a mixed-order Ambisonics microphone array. Márton Marschall and Jiho Chang (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, mm@elektro.dtu.dk)

Recently, there has been increasing interest in using spherical microphone arrays for spatial audio recordings. Accurate recordings are important for a range of applications, from virtual sound environments for hearing research through to the evaluation of communication devices, such as hearing instruments and mobile phones. Previously, a mixed-order Ambisonics (MOA) approach was proposed to improve the horizontal spatial resolution of spherical arrays. This was achieved by increasing the number of microphones near the horizontal plane while keeping the total number of transducers fixed. The approach is motivated by the fact that human spatial hearing is most acute in the horizontal plane. This study presents simulations of the performance of an MOA rigid-sphere microphone array, and its robustness to variations in microphone characteristics. Specifications of a commercially available microphone were used to simulate self-noise, sensitivity, and phase response variations between the microphones. To quantify

the reconstruction error and the “sweet area” as a function of source elevation, the reconstructed sound field based on a simulated array measurement was compared to the reference sound field for both horizontal and elevated sources. It is expected that the MOA approach results in a larger sweet area for mid to high frequencies for horizontal sources.

11:20

1aSP8. Aircraft sound environment reproduction: Sound field reproduction inside a cabin mock-up using microphone and actuator arrays. Philippe-Aubert Gauthier, Cédric Camier, Olivier Gauthier, Yann Pasco, and Alain Berry (Mech. Eng., Université de Sherbrooke, 51, 8e Ave. Sud, Sherbrooke, QC J1G 2P6, Canada, philippe_aubert_gauthier@hotmail.com)

Sound environment reproduction of various flight conditions in aircraft cabin mock-ups is useful for the design, demonstration, and jury testing of interior aircraft sound quality. To provide a faithfully perceived sound environment, time, frequency, and spatial characteristics should be preserved. Physical sound field reproduction approaches for spatial sound reproduction are mandatory to immerse the listener in the proper sound field so that localization cues are recreated. A 80-channel microphone array was built and used to capture a 2-h recording of in-flight sound environments within an actual Bombardier CRJ aircraft. An instrumented cabin mock-up was used to reproduce, in the least-mean-square sense, the recorded sound field using a 41-channel trim-panel actuator array. In this paper, experiments with multichannel equalization are reported. One of the practical difficulties was related to the use of the trim panels as sound sources. Windows and trim panels introduce audible squeaks and rattles if driven at low frequencies. Bass management was therefore implemented. Floor shakers and a subwoofer were used to recreate the low frequency content while the trim panels were only used for the high frequency range. The paper presents objective evaluations of reproduced sound fields. Results and practical compromises are reported.

11:40

1aSP9. Design and implementation of a personal audio system in a car cabin. Jordan Cheer and Stephen J. Elliott (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Highfield, Southampton, Hampshire SO17 2LG, United Kingdom, j.cheer@soton.ac.uk)

The generation of personal listening zones in a car cabin would allow the different occupants to listen to different audio programs without the use of headphones. This would allow, for example, the driver to listen to a navigation system while the rear passengers watched a film. Personal audio systems have previously been implemented in mobile devices and monitors, for example, however, the investigation of the effects of an enclosure on the generation of personal listening zones has been limited. This paper presents an investigation of the effects of a car cabin sized enclosure on the generation of independent listening zones in the front and rear seats. The standard car audio loudspeaker array is used to produce independent listening zones at low frequencies, while a second array of small loudspeakers positioned at the four headrest positions is used to provide control over the rest of the audio bandwidth. The performance of the arrays is first simulated in a car cabin sized rectangular enclosure to understand the physical limits on the achievable control. The proposed arrays are then implemented in a real car to validate the simulation results and the results of a real-time implementation are presented.

MONDAY MORNING, 3 JUNE 2013 — 511AD, 8:55 A.M. TO 12:00 NOON

Session 1aUW

Underwater Acoustics: Seabed Scattering: Measurements and Mechanisms I

Charles W. Holland, Cochair
Appl. Res. Lab., The Penn. State Univ., P.O. Box 30, State College, PA 16801

Dale D. Ellis, Cochair
DRDC Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada

Gavin Steininger, Cochair
School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. (Ring Rd.), Victoria, BC V8P 5C2, Canada

Chair’s Introduction—8:55

Invited Papers

9:00

1aUW1. The small-slope approximation for layered seabeds. Darrell Jackson (Appl. Phys., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, drj@apl.washington.edu)

The small-slope approximation has found application to unlayered seabeds and is generally regarded as an improvement over methods that employ either small-roughness perturbation theory or the Kirchhoff approximation. Unfortunately, the usual small-slope ansatz fails when applied to layered seabeds, as it is inconsistent with perturbation theory. This ansatz is replaced by an alternative, which is found to satisfy the criteria of reciprocity and consistency with the perturbation and Kirchhoff approximations. This approach will be illustrated by computation of the coherent reflection coefficient and scattering strength for a seabed consisting of a single rough fluid layer over a semi-infinite, elastic basement with flat upper boundary. Computation time is significantly longer than for the unlayered case, increasing as the desired accuracy increases. The results will be contrasted with those obtained using a variety of existing approximations.

9:20

1aUW2. Seafloor measurements using synthetic aperture sonar. Anthony Lyons, Daniel Brown (Appl. Res. Lab., Penn State Univ., University Park, State College, PA 16803, apl2@psu.edu), Derek Olson (Grad. Program in Acoust., Penn State Univ., State College, PA), and Shawn Johnson (Appl. Phys. Lab., Johns Hopkins Univ., Laurel, MD)

The past decade has seen considerable growth in the use of synthetic aperture sonar (SAS) imaging systems in both the civilian and military domains. Although SAS systems are almost always uncalibrated, they can still yield information about the seafloor given an understanding of the mechanisms affecting the statistical properties of the images produced by these systems. This talk will describe our recent efforts to link SAS image statistics to seafloor properties through the use of seafloor scattering models. Sample results from several SAS systems encompassing frequencies ranging from 6 to 300 kHz will be shown.

Contributed Papers

9:40

1aUW3. Seabed roughness parameters for the Malta Plateau from joint backscatter and reflection inversion. Gavin Steininger (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. (Ring Rd.), Victoria, BC V8P 5C2, Canada, gavin.amw.steininger@gmail.com), Charles W. Holland (Appl. Res. Lab., The PennState Univ., State College, PA), Stan E. Dosso, and Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper presents seabed interface-scattering and geoacoustic parameters estimated on the Malta Plateau, Mediterranean Sea, by joint Bayesian inversion of monostatic backscatter and spherical-wave reflection-coefficient data. The data are modeled assuming a stack of homogeneous fluid sediment layers overlying an elastic basement. The scattering model also assumes a randomly rough water-sediment interface with a von Karman roughness power spectrum. Scattering and reflection data are inverted simultaneously using a population of interacting Markov-chains to sample roughness and geoacoustic parameters as well as residual error parameters. Trans-dimensional sampling is applied to treat the unknown number of sediment layers and unknown autoregressive order of the errors (to represent residual correlation). Results are considered in terms of marginal posterior probability profiles and distributions, which quantify the effective data information content to resolve scattering/geoacoustic parameters and structure. Results indicate well-defined scattering (roughness) parameters in good agreement with existing measurements, and a multi-layer sediment profile over a high-speed (elastic) basement, consistent with independent knowledge of sand layers over limestone. [Work supported by ONR.]

10:00

1aUW4. Energy exchange and scattering loss within a two-way coupled-mode formulation. Steven A. Stotts and Robert A. Koch (Env. Sci. Lab., Appl. Res. Labs/The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78759, stotts@arlut.utexas.edu)

The loss of energy due to propagation in an ocean environment arises from several possible processes. In a two-way coupled-mode description scattering effects from a rough bottom Pekeris waveguide can be isolated by excluding bulk attenuation, and intermodal energy exchange can be examined. The range- and frequency-dependent interaction of a single trapped mode exchanging energy with multiple continuum modes is identified within this framework. The goal of the analysis is two-fold. First, it provides insight into additional loss mechanisms that can be incorporated into current local mode models (c.f. Koch and Stotts, A Derivation of Energy Loss via Coupled Modes, *165th Meeting of the Acoustical Society of America*, June 2–7, 2013). Second, a comparison can be made to previous descriptions of additional attenuation, such as Kirchhoff scattering loss. Varying the bottom roughness permits tests of the applicability of the Born approximation.

10:20

1aUW5. Derivation of energy loss via coupled modes. Robert A. Koch and Steven A. Stotts (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, koch@arlut.utexas.edu)

A coupled mode formalism can describe energy loss within an ocean waveguide due to all possible mechanisms, including bulk attenuation, scattering from rough surfaces, and volume inhomogeneities. First order corrections produced from mode-couplings can be incorporated into the modal loss. Accounting for these losses within an adiabatic mode approach would provide improvements over current standard modal propagation models. Highlights of the formalism for scattering from a rough surface will be provided.

10:40

1aUW6. Influence of rough seabed surface on statistics of modal energy flux. David P. Knobles and Jason D. Sagers (ARL, UT at Austin, 10000 Burnet Rd., Austin, TX 78758, knobles@arlut.utexas.edu)

When sound propagates through a random media, the wave properties of the acoustic field may be viewed as stochastic variables. It is thus natural to investigate the relationship between the statistical properties of the acoustic field and the random fluctuations of the waveguide. The interest here is a 2-D ocean waveguide with a rough seabed. Numerical solutions to the 2-way integral coupled mode equations (ICME), for random realizations of the roughness from a wavenumber power spectrum, provide the statistics of the modal intensity and cross-mode coherence with range. The roughness induces mode coupling within the trapped spectrum, between the trapped and the continuum spectrum, and to the back propagating modal spectrum. Instead of a master equation for the modal intensities to study the connection between the statistics of the acoustics and the fluctuations in the waveguide as has been advocated in previous studies, the conservation law for acoustic energy flux is used to develop an expression for the individual modal Poynting vectors. In addition to exact numerical computations of the range derivatives of the modal energy flux vectors for both forward and backward propagation, a Poynting vector master equation is derived for the case where the Born approximation is valid.

11:00

1aUW7. An initial model-data comparison of reverberation and clutter from a near-shore site in the Gulf of Mexico. Dale D. Ellis (DRDC Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, daledellis@gmail.com) and John R. Preston (Appl. Res. Lab., The Penn State Univ., State College, PA)

Reverberation measurements were made in the Gulf of Mexico off Panama City, Florida, USA, in April 2012 in preparation for the main Transmission Reverberation Experiment (TREX) in May 2013. The data were gathered using the triplet section of the ONR Five Octave Research Array (FORA), deployed as a fixed receiver. By steering cardioid beams to the right or left, the array can reduce ambiguity. Beamformed data from the 2012 trial show background noise with high directionality and variability due to nearby shipping. Model predictions of reverberation and target are compared with data using a range-dependent Clutter Model, which uses adiabatic normal modes as the computational engine. The initial predictions use isovelocity water, over a sandy bottom halfspace with Lambert scattering, and bathymetry from the $GEBCO_{08}$ database. These initial results will be presented, hopefully supplemented by improved predictions with better environmental inputs and additional clutter data obtained during the May 2013 experiment. [Work supported by ONR Code 322 OA.]

11:20

1aUW8. Evidence for a common scale O(0.1) m that controls seabed scattering and reverberation in shallow water regions. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16801, holland-cw@psu.edu)

Analysis of the spectral content of long-range reverberation yields two observations. First, there is a remarkably similar scale, O(0.1)m, between three diverse continental shelf regions. This is surprising given general understanding of the complexity and diversity of geologic processes. Second, there is strong evidence that the scale is associated with heterogeneities within the sediment. Thus, sediment volume scattering, not interface scattering, controls long-range reverberation from a few hundred Hertz to several kilohertz. This is also unexpected given that at long-ranges the vertical grazing angles are less than the critical angle, and hence, the penetration of the acoustic field into the sub-bottom is expected to be modest. The consistency of the scale, O(0.1)m, suggests an underlying feature or mechanism that is consistent across many ostensibly diverse geological settings. Neither the feature nor mechanism is known at this time. Several hypotheses will be presented. [Work supported by ONR Ocean Acoustics.]

11:40

1aUW9. Rayleigh scattering of sound by spherically symmetric bodies. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

An obstacle's shape is often approximated by a sphere in analyses of sound scattering by air bubbles, objects on or near the seafloor, marine organisms, clouds of suspended particles, etc. Here, an asymptotic technique is developed to study low-frequency sound scattering from spherically symmetric inhomogeneous obstacles. The obstacle can be fluid, solid, or a fluid-filled solid shell. Physical properties of the obstacle are arbitrary piece-wise continuous functions of the distance to its center. The radius of the obstacle is assumed to be small compared to the wavelengths of sound in the surrounding fluid as well as of compressional and shear waves inside the obstacle. General properties of the sound scattering by spherically symmetric bodies are established. Resonant Rayleigh scattering is studied in detail. For plane and spherical incident waves, it is discussed which physical and geometrical parameters of the obstacle can be retrieved from the scattered acoustic field.

MONDAY AFTERNOON, 3 JUNE 2013 513ABC, 12:55 P.M. TO 4:20 P.M.

Session 1pAAa

Architectural Acoustics and Signal Processing in Acoustics: Advanced Analysis of Room Acoustics: Looking Beyond ISO 3382 II

Boaz Rafaely, Cochair
Dept. of Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva 84105, Israel

Samuel Clapp, Cochair
Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180

Michael Vorländer, Cochair
ITA, RWTH Aachen Univ., Neustr. 50, Aachen 52066, Germany

Chair's Introduction—12:55

Invited Papers

1:00

1pAAa1. Theoretic considerations on how the directivity of a sound source influences the measured impulse response. Ingo B. Witew, Tobias Knüttel, and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany, ingo.witew@akustik.rwth-aachen.de)

In previous investigations, it has been shown that the directivity of a measurement sound source has a significant influence on the measured room impulse response (RIR). Using a specialized method of analysis, the sources influence can be identified even in the very late part of the RIR even in very reverberant environments. These results seem to be surprising at first and contradict intuitive expectations. In this contribution, the findings are briefly discussed, and the congruence with general room acoustic theory is revised and discussed.

1:20

1pAAa2. Enhanced spatial analysis of room acoustics using acoustic multiple-input multiple-output systems. Hai Morgenstern and Boaz Rafaely (Ben-Gurion Univ. of the Negev, Ben-Gurion Univ. of the Negev, Beer-Sheva 84105, Israel, hai.morgenstern@gmail.com)

Standard acoustic measurements in enclosures typically employ single-input single-output (SISO) acoustic systems. The parameters obtained from these measurements describe features of energy decay and do not characterize spatial attributes of the enclosure. Directional analysis of enclosures became popular with the introduction of microphone and loudspeaker arrays. In particular, spherical arrays have been shown to be highly beneficial for spatial analysis. Spherical microphone arrays facilitate the estimation of the arrival direction of the direct and reflected sound, while the use of both loudspeaker and microphones arrays can support the estimation of both radiation and arrival directions, with the application of conventional beamforming methods. However, when several reflections are attributed to the same time bin in a discrete impulse response, reflection paths may not be uniquely determined by existing beamforming techniques. We present a new method to uniquely determine source and receiver directions for multiple reflections when time separation is unfeasible. The paper presents the formulation of the proposed method, also showing a simulation study to demonstrate the performance of the proposed method.