Session 4aID

Interdisciplinary: Plenary Lecture: Sensory Evaluation of Concert Hall Acoustics

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

Chair's Introduction—7:55

Invited Paper

8:00

4aID1. Sensory evaluation of concert hall acoustics. Tapio Lokki (Media Technol., Aalto Univ., P.O. Box 15500, Aalto FI-00076, Finland, Tapio.Lokki@aalto.fi)

Consumer products can be perceived in many ways, and individual taste influences quality judgments. Sensory evaluation techniques have been developed to reveal detailed information about perception of products; recently, sensory evaluation has also been shown to be very useful in subjective evaluation of concert hall acoustics. In particular, individual vocabulary based methods have helped to disentangle the detailed perceptual differences between the different seats within a hall and between different halls. For simultaneous and accurate comparison of acoustics, a symphony orchestra needs to play identically in each hall. Therefore, a symphony orchestra simulator has been developed. It consists of 34 loudspeakers reproducing synchronized recordings of individual musicians playing parts of symphonies in an anechoic chamber. In addition, an advanced spatial sound recording technique via impulse responses from a 3D microphone array is applied to reproduce the acoustics of a concert hall in laboratory conditions. Analysis of spatial impulse responses also enables spatio-temporal visualization of sound energy distributions at measured seats, thus helping us to link the physical properties of the sound to the perception and architecture of concert halls. Finally, this paper highlights our recent results to explain which perceptual characteristics of acoustics drive preference ratings.

THURSDAY MORNING, 6 JUNE 2013 513ABC, 9:00 A.M. TO 12:00 NOON

Session 4aAAa

Architectural Acoustics: Room Acoustics Computer Simulation I

Diemer de Vries, Cochair
RWTH Aachen Univ., Inst. fuer Technische Akustik, Aachen D-52056, Germany

Lauri Savioja, Cochair
Dept. of Media Technol., Aalto Univ., P.O. Box 15500, Aalto FI-00076, Finland

Invited Papers

9:00

4aAAa1. Toward a full-bandwidth numerical acoustic model. Jonathan Hargreaves and Yiu W. Lam (Acoust. Res. Ctr., School of Computing, Sci. & Eng., Univ. of Salford, Salford M5 4WT, United Kingdom, y.w.lam@salford.ac.uk)

Prediction models are at the heart of modern acoustic engineering. Current commercial room acoustic simulation software almost exclusively approximates the propagation of sound geometrically as rays or beams. These assumptions yield efficient algorithms, but the maximum accuracy they can achieve is limited by how well the geometric assumption represents sound propagation in a given space. This comprises their accuracy at low frequencies in particular. Methods that directly model wave effects are more accurate but they have a computational cost that scales with problem size and frequency, effectively limiting them to small or low frequency scenarios. This paper will report the results of initial research into a new full-bandwidth model which aims to be accurate and efficient for all frequencies; the name proposed for this is the "Wave Matching Method." This builds on the Boundary Element Method with the premise that if an appropriate interpolation scheme is designed then the model will become "geometrically dominated" at high frequencies. Other propagation modes may then be removed without significant error, yielding an algorithm which is accurate and efficient. This paper will present the general concepts of wave matching and the results from some numerical test cases.

9:20

4aAAa2. Fast multipole accelerated indirect boundary elements for the Helmholtz equation. Nail A. Gumerov, Ross Adelman, and Ramani Duraiswami (Inst. for Adv. Comput. Studies, Univ. of Maryland, 115 A.V. Williams Bldg., College Park, MD 20742, gumerov@umiacs.umd.edu)

The indirect boundary element method for the Helmholtz equation in three dimensions is of great interest and practical value for many problems in acoustics as it is capable of treating infinitely thin plates and allows coupling of interior and exterior scattering problems. In the present paper, we provide a new approach for treatment of boundary integrals, including hypersingular, singular, and nearly singular integrals via analytical expressions for generic boundary conditions on the both sides of the surface. The fast multipole accelerated boundary element solver in Gumerov and Duraiswami (2009) is extended to incorporate the indirect formulation. The new formulation is compared with the analytical solution of scattering off a disk. Previous authors have not provided such comparisons for an extended range of frequencies. The performance of the method and its scalability are investigated. It is demonstrated that problems with millions of boundary elements can be solved efficiently on a personal computer using the present method.

9:40

4aAAa3. Modeling binaural receivers in finite difference simulation of room acoustics. Jonathan Sheaffer (Acoust. Res. Ctr., School of Computing, Sci. and Eng., Univ. of Salford, Salford, Salford M5 4WT, United Kingdom, j.sheaffer@edu.salford.ac.uk), Craig Webb (Acoust. Group/Edinburgh Parallel Computing Ctr., Univ. of Edinburgh, Edinburgh, United Kingdom), and Bruno Fazenda (Acoust. Res. Ctr., School of Computing, Sci. and Eng., Univ. of Salford, Salford, United Kingdom)

Binaural room impulse responses are important for auralization as well as for objective research in room acoustics. In geometrical room simulation methods, obtaining such responses is easily achieved by convolving each computed reflection tap with a corresponding pre-measured angle-dependent head-related impulse response. Unfortunately, employing such an approach in wave based methods is challenging due to temporal overlap of room reflections in the calculated response. One alternative is to physically embed a listener geometry in the grid. Whilst this method is straightforward, it requires voxelization of a geometrically complex object. Furthermore, with non-conformal boundary conditions, the voxelized geometry is sample-rate dependent, meaning that numerical consistency is compromised. In this paper, we discuss the merits and drawbacks of embedding different listener geometries in the grid, ranging from a simple rigid sphere to a fully featured laser-scan of a Kemar mannikin. We then introduce a parametric model of a human listener whose head related effects are structurally approximated by digital filters. The model is applied to simulated results in order to extrapolate a binaural response from a single pressure-velocity receiver, without the need to embed any objects in the grid. A comparative analysis of the two methods is presented, and results are discussed in light of room acoustics modeling.

10:00

4aAAa4. Validation of adaptive rectangular decomposition for three-dimensional wave-based acoustic simulation in architectural models. Lakulish Antani (Comput. Sci., Univ. of North Carolina at Chapel Hill, 1100 W NC Highway 54 Bypass Apt 25G, Chapel Hill, NC 27516, lakulish@cs.unc.edu), Anish Chandak (Impulsonic, Inc., Chapel Hill, NC), Matthew Wilkinson (Arup Acoust., Los Angeles, CA), Alban Bassuet (Arup Acoust., New York, NY), and Dinesh Manocha (Comput. Sci., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Computer-based simulation is an increasingly popular way to predict the acoustics of real-world architectural designs. Most commercial acoustic simulation tools are based on geometric techniques and cannot accurately model low-frequency diffraction and other wave phenomena. Numerical wave simulation techniques can model these effects, but are less commonly used, since they are compute- and memory-intensive, and cannot scale to large spaces. Moreover, it is challenging to ensure that numerical methods do not suffer from high dispersion errors. Recent techniques have begun to overcome these limitations. One such method is adaptive rectangular decomposition (ARD), which combines analytical solutions to the wave equation in rectangular subdomains with a finite difference stencil for interface handling between subdomains, resulting in high-performance wave simulation with low dispersion error. ARD, along with high-performance ray tracing, are available as part of Impulsonic's IPL SDK, a software development kit that allows custom acoustic simulation tools to be easily built with state-of-the-art simulation technology. In this paper, we evaluate the performance and accuracy of the IPL SDK and ARD, by analyzing simulation results and comparing them against measurements obtained for real-world architectural designs.

10:20

4aAAa5. Simulation of non-locally reacting boundaries with a single domain boundary element method. Robertus Opdam (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany, rob.opdam@akustik.rwth-aachen.de), Diemer de Vries (Inst. of Tech. Acoust., RWTH Aachen Univ., Amsterdam, Netherlands), and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany)

The significance of taking into account non-locally reacting behavior of boundaries compared to the often-used locally reacting assumption in room acoustics has not been extensively investigated. To make this possible a boundary element method is developed, inspired on a seismic simulation method known as the WRW method. The novelty of this method compared to other boundary element methods (BEM) is that the calculation is performed in only one domain. There is no need for a fluid-structure coupling, which in general allows faster simulation times. The theory of the method is presented and some example structures are simulated with both locally and non-locally reacting behavior. The results are shown and discussed.

10:40

4aAAa6. Estimation of absorption coefficients values of surface materials using a diffusion equation model. Juan M. Navarro (Adv. Telecommunications Res. Group, San Antonio's Catholic Univ., Campus de los Jeronimos, s/n, Guadalupe, Murcia 30107, Spain, jmnavarro@ucam.edu), Jose J. Lopez (ITEAM, Universitat Politécnica de Valencia, Valencia, Spain), and Jose Escolano (Multimedia and Multimodal Processing Res. Group, Univ. of Jaen, Linares, Spain)

In the auralization process of an enclosure, the right definition of the acoustic properties of the materials is very important. Sometimes, the absorption coefficients of materials in a real room are not found in the literature and their measurement in the laboratory or *in-situ* are complex. Been the reverberation time of a room and its geometry known, but the absorption coefficient values of the materials that cover the room unknown, it is possible to estimate its values by means of an inverse problem using a room acoustics simulation model. Since the acoustic diffusion equation model is a fast simulation method, it can be used to perform an iterative process to estimate these values. In this paper, we propose a statistical procedure that compares actual measurement values of reverberation time with predictions obtained by the diffusion equation model. This process does an automatically adjustment whose ultimate goal is that the reverberation time predicted values do not differ from those measured in situ by more than 5%. As a preliminary work, this algorithm is tested in a cubic room obtaining satisfactory results, but can be extended to be employed in more complex geometry rooms and even with non homogeneous distribution.

11:00

4aAAa7. A diffusion equation model for investigations on acoustics in coupled-volume systems. Yun Jing (Multimedia and Multimedia Processing Res. Group, Univ. of Jaén, 911 Oval Dr., EB III, Campus box 7910, Raleigh, North Carolina 27695, yjing2@ncsu.edu), Ning Xiang (Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY), Juan M. Navarro (Polytechnic Sci. Dept., San Antonio's Catholic Univ., Murcia, Murcia, Spain), and Yun Jing (Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

Coupled-volume rooms remain as one of the most exciting and challenging research lines in room acoustics. Their benefits lie on their multiple-slope energy decay profiles, being of interest in many current concert halls. However, so far there is no consistent predictive model being able to help architects and acousticians in selecting appropriate design parameters. This work is devoted to studying effects of aperture-size and source/receiver positions on the energy decay characteristics. For this purpose, a diffusion equation model is used to model a coupled-volume system, providing an effective tool for analysis. The diffusion equation model is first validated by experimental investigations using scale models. Bayesian energy decay analysis is applied to the results of both the acoustical scale model and the diffusion-equation model to provide deeper insight in the energy decay characteristics and their dependence on the aperture sizes and the sound source/receiver positions.

11:20

4aAAa8. Time-domain formulation of an edge source integral equation. U. Peter Svensson (Dept. of Electron. and Telecommunication, Norwegian Univ. of Sci. and Technol., O.S. Bragstads pl. 2B, Trondheim NO-7491, Norway, svensson@iet.ntnu.no) and Andreas Asheim (Dept. of Comput. Sci., Katholieke Universiteit, Leuven, Belgium)

In computer simulations of sound in enclosures, diffraction components can be added to geometrical acoustics ones for increased accuracy. A computational problem with diffraction is the large number of higher-order terms that is generated. A recent frequency-domain edge source integral equation (ESIE) efficiently handles the sum of all higher-order diffraction for rigid, external scattering objects, while computing first-order diffraction separately. Here, a time-domain formulation of the same ESIE is presented. An initial version handles higher-order diffraction for separate scattering objects, such as stage ceiling reflectors, and the extension to general geometries is outlined. With this approach, in a first step, an incident transient sound field is computed at discretized edge points, including the outgoing directivity. In a second step, the effects of diffraction of arbitrarily high order is handled by solving the IE iteratively, yielding the complete edge source time signals. In a third step, the edge source signals are propagated to receiver points. Numerical issues will be discussed, including discretization strategies and how to handle shadow zone boundary singularities.

11:40

4aAAa9. The contributions of pairs of parallel surfaces in a simple analytical model of room reverberation. Jean-Jacques Embrechts (Intelsig Res. Group, Univ. of Liege, Campus du Sart-Tilman B28, Institut Montefiore, Liege 4000, Belgium, jjembrechts@ulg.ac.be)

In a recent paper [Embrechts, "Searching for a theoretical relation between reverberation and the scattering coefficients of surfaces in a room," in *Proceedings of the Acoustics 2012 Nantes Conference* (2012), 2397–2402], we derived an analytical model of the sound energy decay in a room from the acoustic radiative transfer equation. This model includes the surfaces' scattering properties and it is presently valid for rooms in which the cloud of image sources is approximately isotropic and constant for all receptor's positions. Its validity is extended in this paper by the inclusion of a pair of parallel surfaces. Indeed, it is known that parallel surfaces can introduce significant anisotropy in the cloud of image sources. We show that the room reverberation can be represented by a sum of exponential decays (except in its early part), each decay having its specific slope and amplitude depending on the surfaces' absorption and scattering properties. It is also shown how this simple model can be applied to speed up geometrical acoustics computer simulations.

Session 4aAAb

Architectural Acoustics and Psychological and Physiological Acoustics: Methods and Materials That Improve Speech Intelligibility for the Elderly and Hearing Impaired

Bonnie Schnitta, Chair
SoundSense, LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937

Chair's Introduction—8:55

Invited Papers

9:00

4aAAb1. Achieving optimal reverberation time in a room, using newly patented tuning tubes. Bonnie Schnitta (SoundSense, LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937, bonnie@soundsense.com)

The aging population has several acoustic requirements in order to optimize their ability to hear better, as well as feel better, in a room. First, there is the requirement that the reverberation time of the room must be reduced in order to assist in hearing. Secondly, there needs to be a lowering of the NC in order to increase the SNR. Increasing the SNR not only helps to assist in hearing, but also reduces some hearing aid problems. In addition to these two standard room requirements, there is also a need for the reduction of lower frequency sounds within a room, such as sounds typical of mechanical equipment. Recent data support the fact that there is a correlation between certain diseases and low frequency intolerance. Since standard products used to absorb sound have a greater absorption in speech frequencies there is a need for products that have greater absorption in lower frequencies. Ideally, these products should also be washable. This paper presents detail on each of this requirements, as well as recommendations on methods to achieve each requirement.

9:20

4aAAb2. Optimizing the signal to noise ratio in speech rooms using passive acoustics. Peter D'Antonio (RPG Diffusor Systems, Inc., 651C Commerce Dr., Upper Malboro, MD, pdantonio@rpginc.com)

Adults with normal hearing require roughly a 0 dB signal-to-noise ratio for good speech intelligibility. However, significantly higher values may be needed to compensate for neurological immaturity, sensorineural and conductive hearing losses, language proficiency and excessive reverberation. ANSI 12.60 addresses ways to lower the noise interference due to background levels and reverberation time. However, it is also possible to increase the signal, by reflecting or diffusing early reflections. Speech intelligibility is delivered in the consonants, which occur in the 2–6 kHz frequency range. Therefore, intelligibility can be enhanced by incorporating scattering surfaces, rather than solely surfaces that absorb sound in the 2–6 kHz region, on the front wall, lower side walls, and central ceiling areas, to increase the speech signal by temporal fusion. The decay time can be controlled with broadband absorption on the perimeter of the ceiling and upper wall surfaces. Since ceiling diffusion is an important design ingredient and the ceiling plane is coveted by many trades, including lighting, HVAC, speakers, sprinklers, etc., we will describe a 24 VDC combined LED lighting and sound diffusor, with a 24 VDC emergency lighting central battery system, dynamic lighting capability, and the ability to incorporate sonic actuators for announcements.

9:40

4aAAb3. A holistic approach to room design for hearing impaired populations. Jennifer Levins (Independent, 2669 E Thompson St., Philadelphia, PA 19125, jenlevins@gmail.com)

Acoustic design considerations for hearing impaired populations are widely misunderstood outside of the acoustics community. Some clients have even expressed the sentiment that room acoustics are not important because their patrons are hard of hearing. Contrary to this widely held belief, acoustical design is more critical for these populations. To properly design spaces for these communities, it is imperative to take a holistic approach, which considers not just architectural acoustics, but incorporates an understanding of the biological and psychological components of hearing impairment. It is also important to consider how room systems can be integrated with modern hearing technology. Addressing room acoustics, background sound levels, and audio technology should all be considered in the strategy of designing for hearing impaired persons. This is important not only for their comfort, but also for their health. Strategies and implications for a holistic approach will be discussed.

10:00

4aAAb4. Reduction in reverberation time, resulting from acoustic treatment behind the final surface layer of plywood or drywall. Steve Mittendorf (Mittendorf Quality Construction, 2552 5th Ave. West, Seattle, WA 98119, steve@mittqc.com)

When sound energy generated in room strikes a surface, it is partially reflected, partially transmitted, and partially absorbed. This is true for each layer of material in a wall, ceiling, or floor. The wave interaction with the surface depends on many factors, but the main factors that are typically involved in calculations of reverberation time are the frequencies of concern, the rigidity, and density of the surfaces, and the absorption of various objects in the room. For the case of an empty room, the estimation of the reverberation time is

simplified down to the absorptive properties of the surfaces. This paper presents results that show the importance of considering the composition of the surface (wall, floor, or ceiling) including materials located behind the exposed surfaces. Specifically, it will be demonstrated that a properly installed layer of a loaded vinyl sheeting under the final wall surface layer of drywall will produce a significant reduction in the room reverberation time. With this technique, the preferred reverberation time can be achieved more naturally, while accommodating design constraints such as washable surfaces and minimizing the amount of additional surface treatments required. An additional benefit in the use of the loaded vinyl product behind the surface is a significant improvement in the STC of the wall or ceiling in which it was installed.

Contributed Papers

10:20

4aAAb5. The sensitivity of hearing-impaired adults to acoustic attributes in simulated rooms. William M. Whitmer, David McShefferty, and Michael A. Akeroyd (Inst. of Hearing Res. (Scottish Section), Med. Res. Council, Glasgow Royal Infirmary, Glasgow G42 9UA, United Kingdom, bill@ihr.gla.ac.uk)

In previous studies, we have shown that older hearing-impaired individuals are relatively insensitive to changes in the apparent width of broadband noises when those width changes were based on differences in interaural coherence [Whitmer *et al*., J. Acoust. Soc. Am. **132**, 369–379 (2012)]. This insensitivity has been linked to senescent difficulties in resolving binaural fine-structure differences. It is therefore possible that interaural coherence, despite its widespread use, may not be the best acoustic surrogate of spatial perception for the aged and impaired. To test this, we simulated the room impulse responses for various acoustic scenarios with differing coherence and lateral (energy) fraction attributes using room modeling software (ODEON). Bilaterally impaired adult participants were asked to sketch the perceived size of speech tokens and musical excerpts that were convolved with these impulse responses and presented to them in a sound-dampened enclosure through a 24-loudspeaker array. Participants' binaural acuity was also measured using an interaural phase discrimination task. Corroborating our previous findings, the results showed less sensitivity to interaural coherence in the auditory source width judgments of older hearing-impaired individuals, indicating that alternate acoustic measurements in the design of spaces for the elderly may be necessary.

10:40

4aAAb6. Still able. Trent Still and Daniel Butko (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, William.T.Still-1@ou.edu)

In a world where most students are habitually connected to headphones, one student is harnessing power outside the sense of hearing to unite acoustics and craft into particular listening environments. Trent Still is a student, a craftsman, an avid fan of acoustics, and to my surprise legally deaf in one ear. What initially could be viewed as a hindrance within the study of acoustics, has developed into an avenue of expressive talent and determination. As a student of architectural design, Still focuses on materials, connections, and overall aesthetics of the listening environment. For example: in a recent gallery exhibit of handcrafted furniture, one of Trent's entries was a pair of handmade loudspeaker enclosures that were French cleated to the wall. They were not merely wall mounted; they were wall dependent. The wall cavity between framing members and the wall finish was part of the installation; thereby actively integrating acoustics into architecture. This paper does not focus solely on one student; it is about unequivocal enthusiasm for acoustical craft within inhabitable space. No matter what seems like a disadvantage or disability, students and educators can work together to ascertain visual and auditory beauty. Sight and sound are uniquely codependent.

THURSDAY MORNING, 6 JUNE 2013 517B, 9:00 A.M. TO 12:00 NOON

Session 4aAAc

Architectural Acoustics and National Council of Acoustical Consultants: Student Design Competition

Norman H. Philipp, Cochair
Pittsburg State Univ., 1701 S. Broadway, Pittsburg, KS 66762

Andy Miller, Cochair
BAi, LLC, 4006 Speedway, Austin, TX 78751

David Woolworth, Cochair
Oxford Acoustics, Inc., 356 CR102, Oxford, MS 38655

The Technical Committee on Architectural Acoustics of the Acoustical Society of America with support from the Robert Newman Student Award Fund and the National Council of Acoustical Consultants are sponsoring the 2013 Student Design Competition that will be professionally judged at this meeting. The 2013 design competition involves the design of a college performance hall and related facilities primarily for a school's strong opera program. The submitted designs will be judged by a panel of professional architects and acoustical consultants. An award of USD$1,250 will be made to the submitter(s) of the design judged "first honors." Four awards of USD$700 each will be made to the submitters of four entries judged "commendation."

Session 4aAB

Animal Bioacoustics and Noise: Modeling and Measurement of Anthropogenic Noise in Marine Environments

Bruce Martin, Chair
JASCO Appl. Sci., 32 Troop Ave., Ste. 202', Dartmouth, NS B3B 1Z1, Canada

Invited Papers

10:00

4aAB1. Computing cumulative sound exposure levels from anthropogenic sources in large data sets. Bruce Martin (Halifax, JASCO Appl. Sci., 32 Troop Ave., Ste.202', Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com)

The goal of many underwater acoustic environmental assessments is to characterize the soundscape in an area before, during, or after an anthropogenic activity. The assessment determines the range of baseline noise levels from natural and anthropogenic sources and the contribution of the new anthropogenic activity. The noise levels are considered in aggregate for possible effects on the environment. It is accepted that the effects of anthropogenic noise on marine life depend on the intensity and duration of exposure, the frequency content of the sound relative to the hearing abilities of the species, and the behavior context of the species exposed to the sounds. A growing body of scientific evidence is being analyzed to establish threshold sound levels and dose-response curves for injury or behavioral disturbance effects to marine life. Recent research is also raising new questions about the most appropriate ways to compute ambient sound levels and exposure metrics. In this paper, we present our methods for quantifying ambient sound levels and anthropogenic sound levels from shipping and seismic survey activities in large data sets. We also make recommendations on how to estimate background sound levels in the presence of these sound sources.

10:20

4aAB2. Spectral probability density as a tool for marine ambient noise analysis. Nathan D. Merchant (Dept. of Phys., Univ. of Bath, Claverton Down, Bath BA2 7AY, United Kingdom, n.d.merchant@bath.ac.uk), Tim R. Barton, Paul M. Thompson, Enrico Pirotta (Univ. of Aberdeen, Lighthouse Field Station, Cromarty, United Kingdom), D. Tom Dakin, and John Dorocicz (Ocean Networks Canada, Univ. of Victoria, Victoria, BC, Canada)

The empirical probability density of the power spectral density has been successfully applied as tool to assess signal variability and sensor system performance in the seismic literature. This paper presents the application of this analysis method to underwater ambient noise measurements, and demonstrates its utility in assessing the field performance of passive acoustic monitoring systems and the statistical distribution of noise levels across the frequency spectrum. Using example datasets from an autonomous passive acoustic recorder in the Moray Firth, Scotland, UK, and a cabled subsea observatory in the Strait of Georgia, British Columbia, we show how this method can reveal data limitations such as persistent tonal components and insufficient dynamic range, and phenomena such as bimodality and outliers, which may be undetected by standard analysis techniques. We then combine this approach with conventional percentiles and spectral averages, illustrating how the underlying noise level distributions influence these metrics, and propose this technique as a standard, integrative presentation of ambient noise spectra. Finally, the paper presents cumulative probability density as a method for frequency-domain characterization of chronic noise exposure in marine acoustic habitats.

Contributed Papers

10:40

4aAB3. Global ocean soundscapes. Michael B. Porter and Laurel J. Henderson (HLS Res., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA 92037, mikeporter@hlsresearch.com)

There has been increasing interest in understanding the effects of human-induced noise on the marine environment. Under a variety of programs around the world, researchers are modeling "soundscapes" that depict the undersea sound fields in localized areas such as national exclusive economic zones (EEZs). In this work, we develop techniques for modeling soundscapes on a truly global scale and present as an example world maps of ship noise. The resulting soundscapes compose a database for global shipping noise. The noise due to such shipping can travel very long distances producing sort of a background haze for localized modeling in the EEZs.

11:00

4aAB4. The effects of sound in the marine environment workbench: A simulation tool to predict the impact of anthropogenic sound on marine mammals. David C. Mountain (Biomedical Eng., Boston Univ., 44 Cummington St., Boston, MA 02215, dcm@bu.edu), David Anderson, GraHam Voysey, and Andrew Brughera (Hearing Res. Ctr., Boston Univ., Boston, MA)

The Effects of Sound in the Marine Environment (ESME) Workbench is a software tool designed to predict the impact of anthropogenic sounds on marine mammals. The ESME Workbench (http://esme.bu.edu) allows the user to use site-specific environmental data such as bathymetry and sound-speed profiles to predict sound propagation in a wide range of scenarios and to record the sound exposures received by virtual animals. The acoustic propagation models use range-dependent depth profiles and depth dependent

sound speed profiles to compute the received sound level for simulated animal from each simulated source. The propagation models use bottom and sea surface characteristics to account for losses that occur during reflection at these boundaries. Sound sources are specified through parameters such as source location, frequency, intensity, and beam pattern. The animal behavior is simulated using the 3 MB animal movement model. We will provide hands-on demonstrations at the meeting for those interested in learning more about the ESME Workbench. [Funded by ONR.]

11:20

4aAB5. Behavioral responses of humpback whales to seismic air guns.
Douglas H. Cato (Defence Sci. & Technol. Organisation & Univ. of Sydney, P.O. Box 44, Pyrmont, NSW 2009, Australia, doug.cato@sydney.edu.au), Michael J. Noad, Rebecca A. Dunlop (School of Veterinary Sci., Univ. of Queensland, Gatton, QLD, Australia), Robert D. McCauley (Ctr. for Marine Sci. & Technol., Curtin Univ. of Technol., Bentley, New South Wales, Australia), Hendrik Kniest (Univ. of Newcastle, Newcastle, NSW, Australia), David Paton (Blue Planet Marine, Canberra, ACT, Australia), Chandra P. Salgado Kent (Ctr. for Marine Sci. & Technol., Curtin Univ. of Technol., Bentley, WA, Australia), and K. Curt S. Jenner (Ctr. for Whale Res., Fremantle, New South Wales, Australia)

A study of the responses of humpback whales to seismic air guns is being conducted in Australian waters and two of four major experiments have been completed. It aims to assess the impact of seismic surveys on the whales and the effectiveness of ramp-up in mitigation. In separate trials, whales were exposed to a 20 cu in air gun, ramp-up in level from 20 to 440 cu in with an air gun array, and a "hard start" of 140 cu in. Trials exposing whales to air gun treatments were balanced by controls without air guns firing. Whales were tracked from land using theodolites. Behavioral observations were made from these land stations, from three small vessels, and from the source vessel. Vocalizing whales were tracked with an array of hydrophones. Dtags were attached to some of the whales. Observations were made before, during, and after exposure. Characterization of the sound field throughout the area and the exposure at each whale were determined from propagation measurements and recordings on the hydrophone array and several moored acoustic recording systems. Some preliminary results will be discussed. [Work supported by E&P Sound & Marine Life Joint Industry Program and the U.S. Bureau of Ocean Energy Management.]

11:40

4aAB6. Prediction of noise of moored ships.
Antonino Di Bella and Francesca Remigi (Dept. of Industrial Eng., Univ. of Padova, Via Venezia 1, Padova, PD 35131, Italy, antonino.dibella@unipd.it)

The European Directive 2002/49/CE suggests to map noise in harbor areas with methods mainly used for industrial noise, in accordance with ISO standards. Nevertheless, in many cases, these methods do not seem suitable to describe the effects of noise due to moored cruise ships. For noise measurements of ships, it is possible to refer to ISO 2922 standard, but it results ineffective for the estimation of noise levels at distances bigger than 25 m from the sound source and to obtain reliable information about sound power level of big vessels. The aim of this study, performed by the University of Padova on behalf of Venice Port Authority, is to improve a procedure for predicting noise of moored ships in the harbor area by the means of measures and reverse analysis with numerical models.

THURSDAY MORNING, 6 JUNE 2013 — 519A, 9:15 A.M. TO 11:40 A.M.

Session 4aBA

Biomedical Acoustics and Physical Acoustics: Biophysical Mechanisms of Sonoporation

Richard Manasseh, Cochair
Mech. Eng., Swinburne Univ. of Technol., P.O. Box 218, Hawthorn, VIC, Melbourne, VIC 3122, Australia

John S. Allen, Cochair
Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822

Chair's Introduction—9:15

Invited Papers

9:20

4aBA1. Size effect of complexed plasmid DNA to gene transfection efficiency of microbubble-mediated sonoporation.
Yoichiro Matsumoto, Yiwei Zhang, Takashi Azuma (Mech. Eng., The Univ. of Tokyo, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, ymats@fel.t.u-tokyo.ac.jp), Kiyoshi Yoshinaka (Mech. Eng., The Univ. of Tokyo, Tsukuba, Japan), Kensuke Osada, Kazunori Kataoka, and Shu Takagi (Mech. Eng., The Univ. of Tokyo, Tokyo, Japan)

Ultrasound-mediated gene transfection in the presence of microbubbles is a recently developed promising nonviral gene delivery method. Detailed dynamics of pore opening on the cell surface has not been clarified. Especially, the pore size is one of the most essential parameters. In this study, we investigated the size effect of the complexed plasmid DNA (pDNA) on the transfection efficiency by packaging within the polyplex micelles. Both naked pDNA and complexed pDNA were transfected into cultured NIH3T3 cells using ultrasound in the presence of microbubble contrast agent, Sonazoid. The both size of the hydrodynamic diameter of naked and complexed pDNA estimated by a dynamic light scattering measurement were 600 and 120 nm, respectively. The transfection rates of the complexed pDNA evaluated by counting the number of cells that exhibited green fluorescent was 1.67%, while that of the naked pDNA was 0.92%. This efficiency enhancement depending on the size reduction showed that the pore sizes were distributed in the range of pDNA diameters. Since complexation changes the structure of pDNA in size and stability, more detailed study will be discussed in the presentation.

9:40

4aBA2. Enhancement effect of ultrasound-induced microbubble cavitation on branched polyethylenimine-mediated vascular endothelial growth factor 165 (VEGF165) transfection. Juan Tu, Qian Li, Chunbing Zhang, and Dong Zhang (Physics, Inst. of Acoust., Nanjing Univ., #22 Hankou Rd., Nanjing 210093, China, juantu@nju.edu.cn)

Angiogenesis is a complex process that is mediated by growth factor. One isoform of the vascular endothelial growth factor, VEGF165, has been reported to be a dominant mediator and regulator of angiogenic process. Branched polyethylenimine (bPEI) has been widely used as a non-viral delivery vector for gene therapy. HEK 293T cells, mixed with bPEI:VEGF165 complexes with different N/P ratios, were exposed to 1-MHz ultrasound (US) pulses. The enhancement effect of microbubble inertial cavitation (IC) on bPEI-mediated VEGF165 transfection was systemically investigated, in an effort to optimize transfection efficiency using low nitrogen:DNA phosphate (N/P) ratios. The results show that: (1) Microbubble IC activity can be quantified as an IC "dose" (ICD) and will be affected by US parameters; (2) DNA transfection efficiency initially increases with the increasing ICD, then tends to saturate instead of achieving a maximum value while ICD keeps going up; (3) the measured ICD, sonopration pore size, and cell viability exhibit high correlation among each other; and (4) microbubble IC activity has less cytotoxicity than bPEI, although a combinatorial effect of IC activity and bPEI can be observed on cell viability. All the results indicated that ICD could be used as an effective tool to monitor and control US-mediated gene/drug delivery effect, and it is possible to optimize bPEI-mediated VEGF transfection efficiency with relatively low N/P ratios by employing appropriate US parameters.

10:00

4aBA3. Time-resolved high-speed fluorescence imaging of bubble-induced sonoporation. Michel Versluis (Phys. of Fluids Group, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, m.versluis@utwente.nl)

The uptake of drugs through a cell membrane is enhanced by the use of bubbles and ultrasound. Little is known about the physical mechanisms underlying the uptake at short timescales. Here we study the bubble-assisted uptake of propidium iodide (PI) by endothelial cells at a millisecond timescale using high-speed fluorescence imaging. Single microbubbles were insonified at a driving frequency of 1 MHz and at acoustic pressures varying from 200 to 1200 kPa for a duration of 10 and 100 cycles. At a pressure of 200 kPa and 10 cycles, 50% of the cells showed uptake of PI, and this percentage increased to 90% for a pressure of 400 kPa. At a pressure of 1200 kPa all cells showed uptake of PI. The high-speed fluorescence recordings revealed that a localized pore in the cell membrane is formed right at the position of the bubble. Uptake was observed within several milliseconds after insonation and the size of the induced pore was found to be dependent on the bubble radius. Furthermore, the inflow of PI is diffusion-driven. The pore is formed temporarily and closes within several seconds after the ultrasound exposure.

10:20

4aBA4. Ultrasound-mediated drug delivery with real-time cell permeability measurements. Pavlos Anastasiadis (Molecular Biosciences and Bioengineering, Univ. of Hawaii, Honolulu, HI), Michelle L. Matter (John A. Burns School of Med., Univ. of Hawaii, Honolulu, HI), and John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu)

Ultrasound-mediated drug and gene delivery offers a variety of novel possibilities for improved localized treatment of vascular- and cancer-related diseases. This therapeutic application benefits from the use of acoustic radiation force, which facilitates the exposure for enhanced binding from ligand-receptor interactions. The unique merits of ultrasound are the transient increase of cell permeability without any detrimental and irreversible side-effects. The related underlying molecular and cellular pathways of ultrasound-induced permeability and the subsequent recovery of cells are topics of on-going research. Real-time studies of cell behavior during and past-ultrasound exposure have been limited by the lack of appropriate techniques. The electric-cell impedance sensing (ECIS) technique is a suitable way of studying cell permeability changes in real-time. Its nanoscale sensitivity and speedy acquisition of data allows for the accurate and timely monitoring of cell behavior. Our preliminary results suggest that cells recover within 24–36 h post-exposure. During this time window the cells undergo drastic changes exhibiting an increased permeability of $2.4 \pm 0.6\ \Omega \cdot cm^2$ compared to $3.8 \pm 0.5\ \Omega \cdot cm^2$ that normal untreated cells exhibit.

Contributed Papers

10:40

4aBA5. Investigation on the inertial cavitation threshold of micro-bubbles. Xiasheng Guo, Dong Zhang, and Juan Tu (Dept. of Phys., Inst. of Acoust., No. 22, Hankou Rd., Nanjing 210093, China, guoxs@nju.edu.cn)

Experimental measurements and numerical analyses were performed to investigate the IC thresholds of two commercialized UCAs, albumin-shelled KangRun® and lipid-shelled SonoVue®. The IC thresholds of these two UCAs were measured at varied acoustic pulse lengths and bubble concentrations, according to the IC dose quantifications based on passive cavitation detection (PCD). Then, the shell properties of UCAs were estimated by fitting the measured acoustic attenuation data. Finally, the influences of acoustic pulse length and UCA shell properties on the microbubble nonlinear behaviors were discussed based on numerical simulations, which would give us better understanding of the dependence of microbubble IC threshold on the sonication condition and physical structure properties of the coating shells. The experimental results show that: (1) the IC threshold of UCAs is dependent on the acoustic driving conditions, the shell properties of UCAs and the bubble concentration; (2) for both the lipid- and albumin-shelled UCAs, the IC threshold generally decreases with the increasing UCA volume concentration; (3) IC threshold is observed higher for short-pulse excitation, then its value decreases as the acoustic pulse length increases from 5 to 20 cycles and finally tends to reach a steady state for even longer pulsed exposures.

11:00

4aBA6. Elucidating the effects of low-intensity ultrasound on mesenchymal stem cell proliferation and viability. Nirali Shah, Yosry Morsi (Mech. Eng., Swinburne Univ. of Technol., Melbourne, VIC, Australia), Ursula Manuelpillai (Ctr. for Reproduction and Development, Monash Inst. for Med. Res., Melbourne, VIC, Australia), Tim Barry, and Richard Manasseh (Mech. Eng., Swinburne Univ. of Technol., P.O. Box 218, Hawthorn, VIC, Melbourne, VIC 3122, Australia, rmanasseh@swin.edu.au)

The effects of low-intensity ultrasound (LIUS) on the proliferation, viability and extracellular matrix (ECM) production of mesenchymal stem cells (MSCs) were investigated. Continuous-wave ultrasound was applied at 1 MHz and and 350 mW/cm^2 to microwells, using a LIUS system assembled in the laboratory. Needle hydrophone mapping showed that pressure amplitudes ranged from 0.015 MPa at the well edge to 0.080 MPa at the center. The LIUS group received US for 10, 20, and 30 min/day for one week. Assays were performed daily. Relative to control, 10 and 20 min LIUS very significantly stimulated MSC proliferation and ECM synthesis, while 30 min LIUS had a significant adverse effect. The phenomenon that LIUS accelerates MSC proliferation, but only for appropriate exposures, has been noted previously in the literature. However, the actual relation between the physical forces generated by the LIUS and this phenomenon remains unknown. The fluid flow

pattern created by LIUS was studied by injecting dye in the well and Eckart-streaming-like motions were observed, while thermal effects were negligible. By employing LIUS with appropriate focusing and parameters, it might be possible to exploit MSCs for tissue engineering, independently of biochemical stimuli, and in a highly spatially organized manner.

11:20

4aBA7. "SonoBandage" a transdermal ultrasound drug delivery system for peripheral neuropathy. Matthew Langer, Sabrina Lewis, Shane Fleshman, and George Lewis (ZetrOZ, 421 N. Aurrora St., Ithaca, NY 14850, mlanger@zetroz.com)

Peripheral neuropathy (PN) is a difficult disease to manage. Symptomatic treatment focuses primarily on pain relief, using NSAIDs, opioids, tricyclic antidepressants, and selective serotonin norepinephrine reuptake inhibitors. There is potential for ultrasound transdermal drug delivery to improve the quality of care provided to patients with PN, since it is well-suited to peripheral nerves which are close to the skin. In addition, targeted delivery avoids many of the systemic consequences of taking a drug. We developed a wearable ultrasound drug delivery system called "SonoBandage" that combines low-impedance miniaturization of ultrasound transducer, RF electronics, and battery power supply, with a novel hydrogel coupling bandage loaded with salicylic acid NSAID. The design of the SonoBandage allows the device to be used over a range of ultrasound frequencies (0.1–3 MHz), intensities (0.1–3 W/cm^2), and durations (0.25–4 h) increasing system flexibility for drug delivery protocols. The SonoBandage with NSAID was evaluated on a bench-top model with freshly harvested porcine skin and synthetic biomimetic human skin membrane (Millipore Inc). Across the $n = 40$ samples studied, salicylic acid drug flux was increased by 2–20x as compared to control samples ($p < 0.01$) after 1–4 h of ultrasound treatment. SonoBandage has potential to be used as a practical NSAID delivery platform for peripheral neuropathy.

THURSDAY MORNING, 6 JUNE 2013 — 512AE, 9:00 A.M. TO 12:00 NOON

Session 4aEAa

Engineering Acoustics: Non-Contact Ultrasonic Methods

Michael R. Haberman, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Nico Declercq, Cochair
Georgia Tech Lorraine, 2 rue Marconi, Metz 57070, France

Invited Papers

9:00

4aEAa1. All-optical nonlinear frequency—Mixing acoustics of cracks. Vitalyi Gusev (IMMM, UMR-CNRS 6283, LUNAM Université, Université du Maine, avenue O. Messiaen, Le Mans 72085, France, vitali.goussev@univ-lemans.fr), Nikolay Chigarev, Sylvain Mézil, and Vincent Tournat (LAUM, UMR-CNRS 6613, LUNAM Université, Université du Maine, Le Mans, France)

Recent advances in all-optical evaluation of nonlinear cracks are reviewed. In experiments, the nonlinear acoustic waves are initiated by the absorption of radiation from a pair of laser beams intensity-modulated at two different frequencies. The detection of the acoustic waves at mixed frequencies, absent in the frequency spectrum of the laser intensity, is achieved by optical interferometry or deflectometry. The high contrast in crack imaging achieved by remote optical monitoring of the nonlinear acoustic processes is due to the strong dependence of the optoacoustic conversion efficiency on the state of the crack. The highest acoustic nonlinearity is observed in the transitional state of the crack, which is intermediate between the open and the closed ones. Several crack parameters can be estimated from the measurements of the dependence of the acoustic spectrum on the pump laser intensity. One-dimensional theory of the nonlinear frequency-mixing photo-acoustic crack imaging is presented. The theory relates experimental observation of the large number of mixed frequencies to strong bi-modular nonlinearity of the crack. The theory provides guidelines for the understanding of the dependence of the spatial resolution of this technique on the laser power and on the choice of a particular mixed-frequency component for the crack imaging. [Work supported by ANR project ANL-MEMS ANR-10-BLAN-092302.]

9:20

4aEAa2. Improving the focal quality of the time reversal acoustic noncontact source using a deconvolution operation. Brian E. Anderson, Timothy J. Ulrich, and Pierre-Yves Le Bas (Geophys. Group (EES-17), Los Alamos National Lab., MS D446, Los Alamos, NM 87545, bea@lanl.gov)

The time reversal acoustic noncontact source (TRANS) utilizes time reversal (TR) from several transducers arranged in a cavity to focus energy onto a solid surface to allow inspection of that surface. The advantage of TRANS is that multiple transducers may be used to increase the amplitude of the focused energy onto the surface and potentially allow interrogation of nonlinear surficial features such as cracks and delaminations. TR is known to be a matched signal process and therefore is limited in the temporal fidelity and spatial compression of the focused energy. Fortunately, using a deconvolution operation in conjunction with TR [or inverse filter similar to Tanter *et al*. (J. Acoust. Soc. Am. **108**, (2000), 223–234)] can greatly improve the quality of the spatial focusing as well as increasing the temporal fidelity. Visualizations of the impact of the deconvolution operation will be presented to provide insight into the increased quality of the spatial focusing along with results from using the deconvolution operation in conjunction with TRANS. [Work supported by Institutional Support (LDRD) at the Los Alamos National Laboratory.]

9:40

4aEAa3. Probing of crack breathing by pulsed laser-generated acoustic waves. Vincent Tournat, Chenyin Ni, Nikolay Chigarev (LAUM, CNRS, Université du Maine, Av. O. Messiaen, Le Mans 72085, France, vincent.tournat@univ-lemans.fr), Nicolas Delorme (IMMM, CNRS, Université du Maine, Le Mans, France), Zhonghua Shen (School of Sci., Nanjing Univ. of Sci. and Technol., Nanjing, China), and Vitalyi Gusev (IMMM, CNRS, Université du Maine, Le Mans, France)

Experimental results on all-optical monitoring of the nonlinear motion of a surface-breaking crack are reported. Crack closing is induced by quasi-continuous laser heating, while Rayleigh surface acoustic pulses and bulk longitudinal surface skimming acoustic pulses are also generated and detected by lasers. By exploiting the strong dependence of the acoustic pulses reflection and transmission efficiency on the state—open or closed—of the contacts between the crack faces, the parametric modulation of ultrasonic pulses is achieved. It is observed that bulk acoustic waves, skimming along the surface can be more sensitive to crack motion than Rayleigh surface waves. It has been found that crack closure by thermo-elastic stresses modifies the propagation paths of the acoustic rays from the point source to the point receiver. Consequently, the arrival times of the acoustic waves contain information on the state of crack closure induced by a particular intensity of laser heating. An important dependence of the detected signals on the initial width/state of the crack and on the presence of necks in the crack opening profile is revealed. It is demonstrated that the mode conversion of the skimming longitudinal bulk waves incident on the crack into the transmitted Rayleigh waves is very sensitive to imperfectness of crack closure. The proposed interpretation of the experimental observations is supported by atomic force microscopy measurements.

10:00

4aEAa4. The effects of the transducer beam properties on the ultrasonic geometrical characterization of periodically corrugated surfaces. Jingfei Liu and Nico F. Declercq (George W Woodruff School of Mech. Eng., Georgia Inst. of Technol., 2, rue Marconi, Metz 57070, France, benjamin.jf.liu@gatech.edu)

Periodically corrugated structures are common in many technological applications, and in most cases, the geometry of these corrugated structures are crucial for the designed functionality. As an effective nondestructive characterization method, an ultrasonic imaging technique is investigated in this work for the purpose of accurately characterizing the geometry of periodic corrugations. Among many factors that affect the imaging quality the properties of the transducer beam dominate. The effects of the spatial and spectral properties of transducer beams on the accurate characterization of the characteristic dimensions of corrugations are investigated in details both theoretically and experimentally. The possibility to accurately characterize the corrugation characteristic dimensions, the condition for accurate characterization, and the quantitative relationship between the characterization accuracy and the beam parameters are given. The ways to avoid the diffraction effects and reduce possible errors are also discussed. Experimental results are compared with optical measurements and good agreement is obtained. Both the general principles developed theoretically and the practical techniques proposed can work as a useful guidance for similar work.

10:20–10:40 Break

10:40

4aEAa5. Excitation of Rayleigh and zero-group-velocity Lamb waves using air-borne N-waves focused by an ellipsoidal reflector. Xiaowei Dai (Dept. of Civil, Architectural, and Environ. Eng., The Univ. of Texas at Austin, Austin, TX), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu), Yi-Te Tsai, and Jinying Zhu (Dept. of Civil, Architectural, and Environ. Eng., The Univ. of Texas at Austin, Austin, TX)

Air-coupled ultrasonic non-destructive testing (NDT) of elastic solids is a challenge due to the large impedance contrast between air and most materials used in industrial and structural applications. However, because air-coupled sensing offers many advantages such as rapid scanning of large areas and the elimination of part immersion for inspection, there remains strong incentive to find unique methods for air-coupled excitation of wave motion in elastic solids. This work presents experimental results of an in-air acoustic source that has been shown to excite wave motion in high impedance elastic solids. The source consists of a spark generator and an ellipsoidal reflector. The spark generator radiates a short-duration, high-amplitude acoustic signal as the result of an electrostatic discharge between two electrodes with high potential difference. Analogous to lithotripter, the spark is located at the near focus and generates an outgoing wave that is then focused at the far focus of the reflector which is co-located at the air-solid interface. Measurements of the air-borne acoustic wave in the free-field, the focused acoustic wave, and Rayleigh and zero-group-velocity (ZGV) Lamb waves generated in a concrete slab will be presented and analyzed. [Work supported by NIST Technology Innovation Program (TIP).]

11:00

4aEAa6. Detection and characterization of defects in aerostructures using non-contact ultrasonic transducers. Ngeletshedzo Netshidavhini and Raymond B. Mabuza (NDT and Phys., Vaal Univ. of Technol., Private Bag X021, Andries Potgieter St., Vanderbijlpark, Gauteng 1900, South Africa, ngeletshedzon@vut.ac.za)

This paper describes an investigation into the possibility of using non-contact ultrasonic transducers for detecting and characterizing defects in aerostructures. Our study deals with an ultrasonic method that underscores the recent advances in non-contact and analytical methodologies. Ultrasonic waves are generated by a transducer connected to a Sonatest DryScan 410D using through-transmission mode. In this investigation, ultrasonic through-transmission signals are analyzed for their amplitudes. We treat this problem analytically and experimentally. Various aspects of our approach are presented. The observations reported in this paper deal with defects in aluminum structures. Confirmation of defect geometry is obtained by comparing the results of the non-contact ultrasonic sensors with a conventional ultrasonic testing method. The results obtained are in good agreement with those of the conventional ultrasonic method, indicating that both techniques can be considered as quantitative nondestructive tools for detecting and characterizing defects. Results are presented and discussed.

11:20

4aEAa7. Coupling of finite difference elastodynamic and semi-analytic Rayleigh integral codes for the modeling of ultrasound propagation at the hip. Didier Cassereau, Pierre Nauleau, Quentin Grimal, Jean-Gabriel Minonzio (Laboratoire d'Imagerie Paramétrique, 15 rue de l'Ecole de Médecine, Paris 75006, France, didier.cassereau@upmc.fr), Aniss Bendjoudi, Emmanuel Bossy (Institut Langevin Ondes et Images, Paris, France), and Pascal Laugier (Laboratoire d'Imagerie Paramétrique, Paris, France)

Ultrasonic exploration of the femoral neck is of wide interest as it can provide some information about a potential fracture risk, particularly for osteoporosic patients. *In vivo*, the ultrasonic wave first propagates through soft tissues that can be idealized as a homogeneous fluid. Then, the ultrasonic wave interacts with the bone structure. Transmitted and back-propagated signals are then measured at receivers. A numerical model of this complete chain is useful to understand and control the various parameters involved in this process. The complexity of the bone structure is approached using the elastodynamic finite difference time domain (FDTD) code SimSonic. Due to the small size of the spatial grid needed by FDTD schemes, the propagation between the emitter and the femoral neck may be excessively time and resource consuming. We have developed a coupling between SimSonic and a direct and fast evaluation of the diffraction in homogeneous fluids, based on the numerical discretization of the Rayleigh integral. This approach allows to reduce drastically the total computation time for the complete simulation. Results obtained with this new system are presented, including computation times and computer resources. This approach is particularly useful to simulate experiments with phased arrays, which involve several emissions.

11:40

4aEAa8. Basic examination of noncontact inspection in solid material by using high-intensity aerial ultrasonic waves and optical equipment. Ayumu Osumi (Nihon Univ., 1-8, KandaSurugadai, Chiyoda 101-8308, Japan, oosumi@ele.cst.nihon-u.ac.jp) and Youichi Ito (Nihon Univ., Tokyo, Japan)

Recently, developments have improved methods employing aerial ultrasonic waves for contactless inspection of internal defects in materials such as metals, pipe walls, and fiber-reinforced plastics. Specially, this method is noncontact way differ from conventional ultrasonic inspection that is necessary to contact probe to object. We have developed a new method of aerial ultrasonic inspection that uses high-intensity aerial ultrasonic waves and optical equipments. That is, the object is excited in noncontact way using high-intensity aerial ultrasonic waves and the vibration velocity on the object surface is measured with a laser Doppler galvanometer at same time. We analysis the vibration information and detect defect in materials. We also developed a point-converging acoustic source with a stripe-mode vibration plate to generate the high-intensity aerial ultrasonic waves, an essential component of the method. While the sound source operates at a single resonance frequency, the generated ultrasonic wave has nonlinear acoustic characteristics and generates nonlinear higher harmonics at the focal point because the sound intensity increases by converging a sound wave. Under nonlinear ultrasonic irradiation, the object vibrates at the fundamental frequency and harmonic frequencies corresponding to the ultrasonic waves. Therefore, we also attempted to detect defect in materials for analyzing nonlinear vibration.

THURSDAY MORNING, 6 JUNE 2013 — 512BF, 9:00 A.M. TO 10:20 A.M.

Session 4aEAb

Engineering Acoustics: Acoustics for Navigation

Robert D. White, Chair

Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155

Invited Papers

9:00

4aEAb1. Ultrasonic transducers for navigation. Bernhard E. Boser, Richard J. Przbyla (Berkeley Sensor and Actuator Ctr., Univ. of California, 490A Cory Hall, Berkeley, CA 94720-1770, boser@eecs.berkeley.edu), David A. Horsley, Stefon E. Shelton, and André Guedes (Univ. of California, Davis, CA)

Free space ultrasonic ranging is attractive for applications such as gesture recognition and robotic navigation. Unlike optical ranging technologies, ultrasound based solutions are insensitive to ambient illumination and can therefore be used in- and outdoors. Using time-of-flight, ultrasound rangers work over distances of up to a few meters and achieve sub-mm resolution. Using arrays, objects can be localized in three dimensions. Transducers consist of 400μm aluminum-nitride membranes sandwiched between actuation electrodes batch fabricated on silicon wafers. Unlike capacitive transducers, which require actuation voltages in excess of 100 V, piezoelectric devices are compatible with low-voltage actuation. At the 200 kHz resonance frequency, the wavelength at atmospheric pressure is 2 mm, ideal for compact arrays. The transducers do not dissipate static power and are therefore ideal for battery powered applications. Energy consumption is dominated by the low-noise readout amplifier and is on the order of 1μJ per channel including analog-digital conversion and signal processing, enabling video-rate object tacking at less than 1 mW power dissipation. A prototype system consisting of seven transducers on a 1 mm grid operates up to a 750 mm range and $\pm 35^o$ angle span with $\pm$3.5mm accuracy and $\pm 3^o$ worst case angle error.

9:20

4aEAb2. An infrasound-based avian navigational "map". Jonathan T. Hagstrum (U.S. Geological Survey, 345 Middlefield Rd., MS 937, Menlo Park, CA 94025, jhag@usgs.gov)

The "compasses" (solar, star, geomagnetic) that homing pigeons and other migratory birds use to orient during flight are generally understood, but the "map" sense they need to first determine their homeward direction is not. Atmospheric odor and geomagnetic gradients have been proposed as "map" cues, but are inadequate and remain controversial. Experiments with frosted lenses indicate that sight can also be ruled out. Laboratory tests, however, show that pigeons can detect infrasound (>0.05 Hz), and such signals travel with little attenuation for 1000s of kilometers through the atmosphere. Results from an acoustic ray-tracing program (HARPA) using daily atmospheric profiles are compared with pigeon release data for a number of sites in upstate NY. HARPA runs show that homeward infrasonic cues could have arrived at the sites from directions opposite pigeon departure bearings, especially when these bearings were unusual. Such signals possibly arise from ground-to-air coupling of microseisms or from scattering of microbaroms off terrain features (~0.2 Hz). Pigeons and other birds might use Doppler shifts to determine the directionality of homeward infrasonic cues while flying in circular or other patterns at constant velocity after release; they apparently have built-in airspeed indicators adapted from their olfactory and aural systems.

9:40

4aEAb3. Design, development, and testing of transducers for creating spiral waves for underwater navigation. David A. Brown, Corey Bachand, and Boris Aronov (BTech Acoust. LLC, Adv. Tech. & Manuf. Ctr., Univ. of Massachusetts, 151 Martine St, Fall River, MA 02723, dbAcoustics@cox.net)

The use of spiral waves for underwater acoustic navigation has received much attention in recent years. A spiral wave is characterized as a diverging wavefront that is omnidirectional by magnitude but with a phase that varies linearly by azimuthal angle. Such a signal may be exploited in underwater acoustic navigation when compared with a reference signal of constant phase with respect to

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azimuthal angle. This paper summarizes our sine/cosine spiral wave transducer (SC-SWT) design approach, which creates a spiral wave by generating two orthogonal dipoles driven in phase quadrature using the same cylindrical piezoceramic element. Several acoustic navigation beacon designs that also contain the constant-phase reference transducer have now been fabricated, including variants where spiral and referencing sources have the same effective acoustical origin in the horizontal and vertical planes. Experimental results, test tank calibrations, and problems to overcome are presented.

Contributed Paper

10:00

4aEAb4. Measuring the acoustic response of a compartment fire. Mustafa Z. Abbasi, Preston S. Wilson (Appl. Res. Lab. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E dean Keeton, Austin, TX 78712, mustafa_abbasi@utexas.edu), and Ofodike A. Ezekoye (Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Rescue teams have a small window of time to locate a downed firefighter. Their task is made more difficult due to low visibility, smoke, toxic gases, and high temperatures. In the United States, most firefighters are equipped with a Personal Alarm Safety System (PASS) device that emits an alarm sound, when the firefighter becomes incapacitated. Rescue teams can then follow this sound to the source to locate the downed firefighter. While the PASS device has been enormously successful, anecdotal evidence has shown it fails in some interesting scenarios. For example, cases have been recorded where firefighters inside the building were unable to hear the signal, whereas those outside heard it clearly. To explain these cases, and to improve the signal used by the PASS device, it is necessary to understand sound propagation in the fireground environment. This paper will present acoustic transfer measurements inside a laboratory compartment fire, simulating a fire in a residential structure. The research aims to understand how the developing temperature gradient and smoke layer influences sound propagation. A secondary goal is the development and validation of finite element models of fireground acoustics. [Work supported by U.S. Department of Homeland Security Assistance to Firefighters Grants Program.]

THURSDAY MORNING, 6 JUNE 2013 — 512DH, 9:00 A.M. TO 12:20 P.M.

Session 4aMU

Musical Acoustics: Transient Phenomena in Wind Instruments: Experiments and Time Domain Modeling

Stefan Bilbao, Cochair
Music, Univ. of Edinburgh, Rm. 7306B, JCMB, Kings Bldgs., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom

D. Murray Campbell, Cochair
School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom

Invited Papers

9:00

4aMU1. Modeling pulse-like lip vibrations in brass instruments. Jonathan A. Kemp (Dept. of Music, Univ. of St Andrews, Beethoven Lodge, 65 North St., St Andrews, Fife KY16 9AJ, United Kingdom, jk50@st-andrews.ac.uk) and Richard A. Smith (Smith Watkins Trumpets, Sheriff Hutton, Yorkshire, United Kingdom)

During the starting transient of a note on a brass instrument it can take several cycles of lip vibration before acoustics reflections from the end of the instrument can influence the lip frequency. Under certain conditions, the lip may fail to oscillate at the pitch of the air column resulting in an unwanted pulse-like waveform with relatively low repetition rates (similar to the vocal fry register of phonation in the human voice). This is often observed in the playing of beginners if the lips are insufficiently tense or if the top and bottom lips overlap to a large extent. In this study, the reasons for this behavior will be investigated using modeling techniques with the aim of improving the agreement between physical models and measured transients by including the forces responsible for this effect.

9:20

4aMU2. Transient phenomena in brass instruments. John Chick (School of Eng., Univ. of Edinburgh, Kings Blds., Edinburgh EH9 3JL, United Kingdom, john.chick@ed.ac.uk), Shona Logie (School of Phys., Univ. of Edinburgh, Edinburgh, United Kingdom), Lisa Norman (Reid School of Music, Univ. of Edinburgh, Edinburgh, United Kingdom), and Murray Campbell (School of Phys., Univ. of Edinburgh, Edinburgh, United Kingdom)

The starting transient and the transition between notes are known to be of fundamental musical significance on all instruments. On a brass instrument, the player needs to establish a strongly coupled resonance between the air column and the lips for a note to sound effectively. In the case of a slurred transient, the player must decouple the lips from one resonance before establishing the next. The ease with which this can be achieved depends on several factors including tube length and bore profile, and resonant modes being played. Analysis of measured mouthpiece pressure data and time domain computer modeling have been used to explore transient phenomena in brass instruments, with the aim of identifying desirable playing characteristics of an instrument.

9:40

4aMU3. Transient variation in mechanical action and electric action pipe organs. Alan Woolley (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., King's Bldgs., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, awoolley@staff-mail.ed.ac.uk)

Control of the transients by the player has often been cited as one of the most important characteristics of mechanical pipe organ actions since their reintroduction toward the middle of the last century. Previous research indicates that players do not vary the way that they move the key to a significant extent, except as the result of starting the finger movement from some distance above the key rather than in contact with it. This does not necessarily lead to an audible difference. There are, however, other factors in pipe organ action design, pipe voicing, and the way in which organs are played that may lead to real or apparent transient variation irrespective of the type of action. It is well recorded that it is desirable to stagger the release of a chord starting with the pipes with least wind requirement in order to minimize the effect on the wind chest pressure particularly with traditional pressure regulators remote from the wind chest. This paper investigates some of these mechanisms and compares transient variation on mechanical action organs and electric action organs due to different playing styles.

10:00

4aMU4. Some simulations of the effect of varying excitation parameters on the transients of reed instruments. Fabrice Silva (Laboratoire de Mécanique et d'Acoustique, CNRS-LMA, Marseille, France), Vincent Debut (Appl. Dynam. Lab., Campus Tecnológico e Nuclear, Instituto Superior Técnico/Universidade Técnica de Lisboa, Sacavem, Portugal), Philippe Guillemain, Jean Kergomard, and Christophe Vergez (Laboratoire de Mécanique et d'Acoustique, CNRS-LMA, 31 Chemin Joseph Aiguier, Marseille 13402, France, kergomard@lma.cnrs-mrs.fr)

This paper considers the simulation of self-sustained oscillations in reed and brass instruments, based on a compact continuous-time formulation of the sound production mechanism. The control parameters such as the mouth pressure and the player's embouchure, but also the acoustic resonator and the reed, may vary with respect to time, allowing the analysis of transient and non-stationary phenomena like changes of regime. A particular attention is first given to staccato notes, with comparison of the evolution of the instantaneous frequency in simulations to theoretical and experimental results. This shows the importance of using realistic control parameters on the onset of the oscillations. When the acoustic resonator is modeled using a modal expansion with non-stationary resonance frequencies and damping, it is also possible to simulate and study slurs and musical effects like the wah-wah, gaining some insight on the mechanisms involved.

Contributed Papers

10:20

4aMU5. Modeling articulation techniques in single-reed woodwind instruments. Vasileios Chatziioannou and Alex Hofmann (Inst. of Music Acoust., Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Bldg. M, Vienna 1030, Austria, chatziioannou@mdw.ac.at)

Time-domain simulations of wind instruments can, in principle, deal with non-linear oscillations and are also capable of modeling both the steady-state and the transient behavior of a system. The starting transient is usually an important identifying feature of the instrument that is played. Subtle control of articulation is required from skilled musicians to modulate transients during expressive performance. Focusing on single-reed woodwind instruments, the physical phenomena that underlie different articulation techniques are analyzed. A saxophone player is recorded during portato playing, where articulation is achieved either by the use of the tongue, or by modulating the air flow into the mouthpiece. The bending of the reed and the pressure inside the mouthpiece are measured and a physical model is formulated with the aim to capture the transient effects. Instead of adding new terms (and complexity) to a single mass-spring model, in order to simulate the player's tongue, existing physically meaningful parameters are allowed to vary. In particular, the effect of tonguing is modeled by modulating the equilibrium position of the (lumped) reed and its internal damping, whereas in the case of air-separated tones, only a variation of the blowing pressure is required.

10:40

4aMU6. Measurement setup for articulatory transient differences in woodwind performance. Alex Hofmann, Vasileios Chatziioannou (Music Acoust. (IWK), Univ. of Music and Performance Art Vienna, Anton von Webern Platz 1, Vienna, Select State 1030, Austria, hofmann-alex@mdw.ac.at), Michael Weilguni (Inst. of Sensor and Actuator Systems, Vienna Univ. of Technol., Vienna, Austria), Werner Goebl, and Wilfried Kausel (Music Acoust. (IWK), Univ. of Music and Performance Art Vienna, Vienna, Austria)

To model transient differences caused by varying articulation techniques on single-reed woodwind instruments, human performances have to be measured and analyzed. In a previous study, we investigated differences between tongued and air-separated tones on a saxophone by monitoring inner mouthpiece pressure, mouth pressure, and reed bending during performance. Some of the observed effects (e.g., damping and displacement of the reed) were applied to a physical model. Although tip-opening is an essential parameter of lumped reed models, we were only able to directly compare our measurements to the model by inner mouthpiece air-pressure. In this study, we aim to relate measurable reed-bending to the resulting tip-opening and also determine the sensor-reed's stiffness. A micro-mechanical characterization test system records the static and dynamic mechanical data of the sensor-equipped reeds. Pull and tensile measurements are used to reveal the relation between reed bending and tip opening and will increase our understanding of quasi-static properties of the reed-mouthpiece system. This allows conclusions about the tip opening behavior at transient emergence in expressive single-reed woodwind performance.

11:00

4aMU7. Modes of reed vibration and transient phenomena in free reed instruments. James P. Cottingham (Physics, Coe College, 1220 First Ave., Cedar Rapids, IA 52402, jcotting@coe.edu)

The motion of air-driven free reeds used in the harmonica, accordion, and reed organ is dominated by the fundamental transverse beam mode, but higher transverse modes and the first torsional mode are usually present during steady oscillation, even at low amplitude. In addition, a lateral mode has sometimes been detected, in which the reed tongue oscillation is perpendicular to the transverse oscillation. Interaction of the reed with a resonance in the instrument can result in unusual effects. In the accordion, resonances of the reed cavity can interfere with the reed self-excitation mechanism. In the harmonica, when the reed is nearly closed, a strong aerodynamic instability can in some cases lead to torsional flutter. A characteristic of some free reed instruments is a slow attack, in which the sound builds gradually and often unevenly, with the effect being greater for the longer, lower-pitched reeds. There is evidence that the first torsional mode and the second transverse mode may be significant in initiating reed oscillation, so that reed design enhancing the torsional mode may be helpful in alleviating the problem of slow attack.

11:20

4aMU8. Direct numerical simulations of the recorder in two and three dimensions. Nicholas Giordano (Dept. of Phys., Purdue Univ., 525 Northwestern Ave., West Lafayette, IN 47923, giordano@purdue.edu)

Direct numerical solutions of the compressible Navier-Stokes equations have been used to study various aspects of sound production in the recorder. A custom algorithm implemented on a parallel computer has enabled us to calculate tones and produce visualizations of the air flow near the labium in both two and three dimensions. In three dimensions, we have observed how the attack portion of the tone and the spectrum at long times depends on the relative alignment of the channel and labium. We also describe subtle differences in the process of vortex shedding in two as compared to three dimensions.

11:40

4aMU9. Numerical reproducibility of time-dependent motions of spatial waves in air-columns of wind instruments. Kin'ya Takahashi, Saya Goya, Kana Goya, and Chisato Susaki (The Phys. Labs., Kyushu Inst. of Technol., Kawazu 680-4, Iizuka, Fukuoka 820-8502, Japan, takahasi@mse.kyutech.ac.jp)

Since Schumacher introduced a time-domain model of single-reed instruments and McIntyre *et al.* gave the general concept of time-domain models of wind instruments, the time-domain models, namely, delayed feedback models, have become an important numerical tool for study of wind instruments due to their simplicity, easiness to handle, and reliability. However, those models only reproduce wave oscillations observed in mouthpieces of wind instruments. In this talk, we will propose a numerical technique, which is able to reproduce time dependent motions of spatial waves in an air-column. It is composed of inversed wave propagator matrices combined with the forward and backward Fourier transformations. The resultant spatial waves in the air-column exhibit very similar time-dependent behavior to those observed by an experiment for the clarinet. Actually, backward and forward rounded-off step waves are observed. We will also discuss difference in wave shapes and their time-dependent behavior depending on the shapes of air-columns, cylindrical one like the clarinet, conical one like the saxophone and horn-shaped one like brass instruments. For the conical and horn-shaped air-columns, Helmholtz-like waves are observed rather than the step waves observed for the cylindrical air-column.

12:00

4aMU10. A thermoviscous tube propagation model suitable for time domain analysis. Stephen C. Thompson, Thomas B. Gabrielson (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16803, sct12@psu.edu), and Daniel M. Warren (Knowles Electron., Itasca, IL)

Modeling acoustic propagation in tubes including the effects of thermoviscous losses at the tube walls is important in thermoacoustics, in hearing aid modeling, and in modeling wind musical instruments. Frequency dependent impedances for a tube transmission line model in terms of the so-called thermal and viscous functions are well established, and form the basis for frequency domain analysis of systems that include tubes. However, frequency domain models cannot be used for systems in which significant nonlinearities are important, as is the case with the pressure-flow relationship through the reed in a woodwind instrument. This paper describes a tube model based on a continued fraction expansion of the thermal and viscous functions. The expansion can be represented as an analog circuit model, which allows its use in time domain system modeling. A simple model of a clarinet-like oscillation will be shown.

THURSDAY MORNING, 6 JUNE 2013 — 511BE, 8:55 A.M. TO 12:00 NOON

Session 4aNSa

Noise, Architectural Acoustics, and Psychological and Physiological Acoustics: Effects of Noise on Human Performance and Comfort I

Lily M. Wang, Cochair
Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816

Arianna Astolfi, Cochair
Politecnico di Torino, Corso DC degli Abruzzi, 24, Turin 10124, Italy

Chair's Introduction—8:55

Invited Papers

9:00

4aNSa1. Outline proposal for a good practice guide on the evaluation of human response to vibration from railways in residential environments. Andy Moorhouse, David Waddington, Eulalia Peris, James Woodcock, Calum Sharp, and Gennaro Sica (Acoust. Res. Ctr., Univ. of Salford, Newton Bldg, Salford M20 1JJ, United Kingdom, a.t.moorhouse@salford.ac.uk)

The paper will present outline proposals for a good practice guide for the evaluation of human response to vibration from railways in residential environments. The context is the need to increase the proportion of freight carried by railways in Europe while avoiding additional disturbance from vibration to populations living nearby. Within this context, the European funded project CARGOVIBES is developing the good practice guidelines to assist in the evaluation of potential disturbance. In the guide, it is proposed to include descriptions of the adverse response, primarily annoyance and sleep disturbance. The proposal is to use measured or predicted vibration metrics in conjunction with dose-response relationships to quantify potential adverse impact to residents. To this end, measurement and assessment of vibration will be considered, for example, the equipment, locations, mounting, and a description of the data to be acquired. The latest information on dose-response relationships will then be reviewed. Finally, various national limits for vibration will be evaluated against new scientific evidence on dose-response relationships.

9:20

4aNSa2. Analysis of railway vibration signals using supervised machine learning for the development of exposure-response relationships. Calum Sharp, James Woodcock, Eulalia Peris, Gennaro Sica, Andrew Moorhouse, and David Waddington (Acoust. Res. Ctr., Univ. of Salford, The Crescent, Salford M5 4WT, United Kingdom, c.sharp@edu.salford.ac.uk)

The aim of this work is to investigate the applicability of the use of supervised machine learning methods to classify unknown railway vibration signals within a measurement database. The results of this research will be implemented in the development of exposure-response relationship for annoyance caused by freight and passenger railway vibration, so as to better understand the differences in human response to these two sources of environmental vibration. Data for this research come from case studies comprising face-to-face interviews with respondents and measurements of their vibration exposure collected during the University of Salford study "Human Response to Vibration in Residential Environments." Vibration data from this study are then classified into freight and passenger categories using supervised machine learning methods. Finally, initial estimates of exposure-response relationships are determined using ordinal probit modeling. The results indicate that the annoyance response due to freight railway vibration may be significantly higher than that due to passenger railway vibration, even for equal levels of exposure. The implications of these findings for the potential expansion of freight traffic on rail are discussed. [Work funded by the Department for Environment, Food and Rural Affairs (Defra) UK, and EU FP7 through the CargoVibes project.]

Contributed Papers

9:40

4aNSa3. Study on the annoyance of high frequency noise at industrial workstations. Bozena E. Smagowska (Dept. of Vibroacoustic Hazards, Central Inst. for Labour Protection - National Res. Inst., Czerniakowska str. 16, -, Warsaw, Mazowieckie 00-701, Poland, bosma@ciop.pl)

The aim of the study was subjective assessment of the acoustic environment of persons exposed to high frequency noise at workstations. The study based on the survey regarding the annoyance of this type of noise which was conducted among 52 operators of equipment for the production of platform gratings. The study covered workstations, where acoustic measurements confirmed the presence of high frequency noise (in the frequency range from 10 to 20 kHz). During work at these workstations, the value of the equivalent sound pressure level in one-third octave-bands center frequencies of 10, 12.5, and 16 kHz occurs within the range of 81–103 dB and exceeds the admissible value defined for these frequency bands. The results of the study indicated that 92% of respondents are exposed to noise the whole time of the shift. All respondents wear hearing protection. Most of the employees describe the noise as: buzzing, insistent, high-pitched squeaky, and whistling. Respondents unanimously consider related noise levels as: loud, impeding communication, highly strenuous, and tiring. The highest number of points on a scale corresponding to noise annoyance was achieved by terms: horrible, very, persistently, and firmly.

10:00

4aNSa4. Comparison of discomfort caused by two kind of backup alarm. Laurent Brocolini, Lucie Léger, Etienne Parizet (Laboratoire Vibrations Acoustique, INSA, 25 bis, av. J. Capelle, Villeurbanne F-69621, France, laurent.brocolini@gmail.com), Jean-Marie Verlhac, and Xavier Carniel (Ingénierie Bruit et Vibrations, CETIM, Senlis, France)

Nowadays used on most of construction vehicles, tone backup alarm causes a strong discomfort among resident citizens. To solve this problem, "CETIM" (Mechanical Industries Technical Centre) decided to characterize and test another kind of alarm. This one is called "Cri du Lynx" in french (Lynx scream) because of its particular sound looks like a white noise. A previous study showed that the average sound level of the noise-alarm detectability is 67 dB (A) while it is 64 dB (A) for the tone-alarm (in a 80 dB(A) background noise). However, the noise-alarm could be used if it is less disturbing than the tone-alarm. The present work aims to compare the discomfort caused by exposure to tone and noise back-up alarms. About 50 people rated two times the discomfort caused by 11 sound environments. Each sound environment consisted of a site construction sound environment set at about 65 dB (A) mixed with a sequence of tone or noise alarm set at 56, 59, 62, 65, or 68 dB(A). One of the 11 stimuli had no alarm. Through an ANOVA it can be said if the kind of alarm and/or the sound level has an influence on the discomfort.

10:20

4aNSa5. If a tree falls in a forest, can you hear it? Max P. Weichert (Bionik/Biomimetics in Energy Systems, Carinthia Univ. of Appl. Sci., Emil-von-Behring-Strasse 28, Villach 9500, Austria, max.weichert@edu.fh-kaernten.ac.at)

Anthropogenic noise, from slowly rotating/reciprocating machinery, contributes immensely to environmental low-frequency noise below 100 Hz. This poses an acoustic low-energy problem as our technology offers no effective passive method for remediation. While attention is given to negative impact on human health and ecosystems, we still know and publicly do too little about it. Technical standards in Europe are A-weighted measurements that disregard actual energetic contributions of low frequencies. Different aspects of low-frequency sound have been illuminated by complementing acoustic measurements with findings from technical acoustics, bioacoustics, and human medical science. Sound pressure levels (SPL) were measured from 2.5 Hz to 20 kHz. The measurement setup consisted of SINUS Soundbook quadro (2.3 Hz– 22 kHz; linear@0 dB; $\pm$0.1 dB tolerance; 1 Hz high-pass enabled), 1/2-in. Preamplifier/Free-Field Microphone (3.15 Hz– 20 kHz: $\pm$2.0 dB; 5Hz–10 kHz: $\pm$1.0 dB). Signals were processed with SINUS Driver version 5.1.0.8 and Samurai version 2.0.134. SPL in quiet forest [during 60 s: LZ,eq = 59.3 dB; LA,eq = 29.5 dB(A)] were compared to a quiet university room [during 60 s: LZ,eq = 74.4 dB; LA,eq = 28.2 dB(A)]. Healing properties of felid purrs produced with strong frequencies at an SPL between 30 and 60 dB suggest a general correlation of low-frequency background noise levels and health. Perhaps nature knows better than we do.

10:40

4aNSa6. Prediction of the effectiveness of a sound-masking system in an open-plan office including the Lombard Effect. Yizhong Lei and Murray Hodgson (Mech. Eng., Univ. of British Columbia, -Ste. 2137, 6335 Thunderbird Cres., UBC, Vancouver, BC V6T2G9, Canada, light.lei@hotmail.com)

Sound masking can improve speech privacy in rooms by increasing background-noise levels that mask distracting speech sounds. The Lombard Effect indicates that an increase in background-noise level can increase talker voice levels, reducing speech privacy and the benefit of a sound-masking system. To investigate this, a model of an existing open-plan office was created in CATT-Acoustics and validated. The model was used to predict speech-transmission index (STI) and the effectiveness of a sound-masking system, without and with the Lombard Effect, described by the existing Lombard Voice Model. Predictions were made for ambient-noise levels of 30, 40, and 45 dBA, at various distances from a primary talker, and for 0–4 secondary talkers. With 30-dBA ambient noise, STIs at 1 and 4 m varied with the number of talkers from 0.67 to 0.91 and 0.23 to 0.62 without the sound-masking system, from 0.58 to 0.70 and 0.13 to 0.25 with it but ignoring the Lombard Effect, and from 0.64 to 0.73 and 0.18 to 0.27 with the

Lombard Effect. The Lombard Effect reduced the benefit of the sound-masking system by 3–9% and 7–22%. With higher ambient noise, the system is less effective; the Lombard Effect can almost cancel its benefit, resulting in increased STI (decreased privacy) with the system operating.

11:00

4aNSa7. Acoustical characteristics of Technology Educational Shops in British Columbia. Ahmed Summan and Murray Hodgson (SPPH, UBC, 5613 Montgomery Place, Vancouver, BC V6T2C8, Canada, ahmedsumman@gmail.com)

Technology Educational Shops (TES) are designed to develop high school students' technological literacy. Their acoustical conditions play a dominant role in the quality of these environments. TES are, at the same time, classrooms for learning and industrial workshops for making things. Each use has its own standards governing its acoustical characteristics: ANSI S12.60-2002 for classrooms and the Ondet & Sueur DL2 criteria for workshops. A major conflict could exist by using the same room for two different purposes. This study investigated this conflict by evaluating the acoustical characteristics of 20 unoccupied wood, metal and automotive shops. It conducted measurements of background noise level (BNL), reverberation time (RT), speech intelligibility index (SII), and DL2. Results showed that BNLs and RTs in most TES were higher than the acceptability criteria for unoccupied core learning spaces. SII values indicated bad/poor speech intelligibility for normal and raised voice levels and reasonable/good speech intelligibility for loud and shout voice levels. DL2 values were found acceptable in TES larger than 100 m^2 in floor area. In general, these results indicate the poor acoustical conditions of TES as classrooms, and the need for special sound control measures.

11:20

4aNSa8. Sound quality in small music classrooms. Iara B. Cunha, Tiago Mattos, and Stelamaris R. Bertoli (FEC - Arquitetura e Construção, UNICAMP, Av. Albert Einstein, 951 - Caixa Postal: 6021, Campinas, SP 13083-852, Brazil, iarabcunha@gmail.com)

Good conditions for teaching, learning, and practicing a musical instrument are crucial for a musician's progress. Small rooms for teaching and practicing musical instruments have specific requirements about acoustic performance and strong influence on the perception of the users. Acoustical problems in music classrooms may cause difficulties for teachers to identify mistakes from the young students performance other than those originated from the room itself. For this paper, three music classrooms were evaluated according to reverberation, background noise, and airborne sound insulation between rooms. Thus, reverberation time (RT), background noise level, and standardized level difference (DnT), as a function of frequency and according to ISO 3382-2:2008 and ISO 140-4:1998 standards, were measured and judged. As some results have disagreed from literature recommendations, the main faults of each room were highlighted, considering the type of instrument that is taught, and suggestions for acoustic adjustment were made.

11:40

4aNSa9. Nocturnal vibration from high numbers of freight trains leads to fragmented sleep. Michael G. Smith, Ilona Croy, Oscar Hammar, Mikael Ögren, and Kerstin Persson Waye (Occupational and Environ. Med., Univ. of Gothenburg, Box 414, Gothenburg 40530, Sweden, michael.smith@amm.gu.se)

There are an increasing number of freight trains on the European railway networks, and this growth has been facilitated through use of the available night time periods. Freight trains are particularly problematic with regards to generation of low frequency vibration and noise which has the potential to propagate to nearby homes and influence the sleep of residents. To investigate the potential impact we conducted a laboratory trial on 24 young healthy persons to ascertain physiological and psychological reactions to nocturnal vibration and noise from freight traffic, and to examine differences between gender and noise sensitivity. Nights with low (0.0102 m/s^2) and high (0.0204 m/s^2) peak weighted vibration amplitudes and low (20) and high (36) number of train passages were simulated with noise levels being of the same order between nights. Polysomnography was used to record sleep stage and EEG arousals and awakenings. Event related cardiac activations were analyzed using ECG recordings. Questionnaires were administered in the evenings and mornings to obtain subjective sleep parameters. Sleep was more fragmented during nights with higher vibration amplitudes and number of events. Furthermore, heart rate response was higher in the high vibration condition. Results from the subjective data showed less discrimination between nights.

THURSDAY MORNING, 6 JUNE 2013 — 511CF, 8:55 A.M. TO 10:40 A.M.

Session 4aNSb

Noise, Animal Bioacoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Future of Acoustics

Michael J. Buckingham, Cochair
SIO, UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238

Brigitte Schulte-Fortkamp, Cochair
Technische Universität Berlin, Einsteinufer 25, Berlin 10178, Germany

Chair's Introduction—8:55

Invited Papers

9:00

4aNSb1. Ocean noise: Lose it or use it. William A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238, wkuperman@ucsd.edu)

Ocean noise, natural or man-made, has typically been treated as unwanted interference in the context of detecting signals. However, more recently noise has itself also become a signal of interest in which, for example, ocean or geophysical properties are embedded in the noise field. There is now significant ongoing research in trying to extract information from noise. Much of the latter has utilized either own-ship noise, surface generated noise, biological noise, and/or distant shipping noise. Here we review the evolution of research in ocean noise as it progressed from basic descriptive categorization to attempts to minimize its impact on signal detection, and finally, to its utilization as an environmental descriptor.

9:20

4aNSb2. Ambient noise measurements with deep sound in the Philippine Sea. Michael J. Buckingham and David R. Barclay (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

Most measurements of ambient noise in the deep ocean have been performed using an array of hydrophones located at a fixed depth. Recently, an instrument platform known as Deep Sound has been developed, consisting of a glass sphere containing data acquisition, data storage, and system control electronics, with a pair of vertically aligned hydrophones mounted externally. Deep sound descends under gravity, jettisons a drop weight at a pre-assigned depth, and returns to the surface under buoyancy, traveling in both directions at a nominal 0.6 m/s. Throughout the descent and ascent, the hydrophones record the ambient noise over a bandwidth from 3 Hz to 30 kHz. In April–May 2009, Deep Sound was deployed to a depth of 5500 m in the Philippine Sea. The vertical coherence of the measured noise, from 1 to 10 kHz, matches accurately a simple theory of deep-water, wind-generated ambient noise, provided that the local sound speed is used in evaluating the theoretical coherence function. Moreover, the cross-correlation function of the noise, obtained by taking the Fourier transform of the coherence function, provides the basis of an inversion technique for returning the sound speed profile in the water column. [Research supported by ONR.]

9:40

4aNSb3. Soundscape-focusing on resources. Brigitte Schulte-Fortkamp (Technische Universität Berlin, Einsteinufer 25, Berlin 10178, Germany, b.schulte-fortkamp@tu-berlin.de)

In contrast to many other environmental problems, noise pollution continues to grow, and it is accompanied by an increasing number of complaints from people exposed to the noise. The growth in noise pollution involves direct, as well as cumulative, adverse health effects. It also adversely effects future generations, and has socio-cultural, aesthetic, and economic effects. Therefore, the concept of noise annoyance needed to be broaden to an integrated environmental, psychosocial, and socioeconomic assessment of the community situation to reach a more realistic basis for environmental impact and health risk assessments. Soundscape research represents a timely paradigm shift in that it combines physical, social, and psychological approaches and considers environmental sounds as a “resource” rather than a “waste” to satisfy human needs and wants. Moreover, balancing between the expertise from people living in respective areas and acoustic measurements, architectural planning will lead to a new understanding of a concept of an environment under “noise control” as soundscape suggests exploring noise in its complexity, its ambivalence, and its approach toward sound and quality of life.

10:00–10:40 Panel Discussion

THURSDAY MORNING, 6 JUNE 2013 — 511CF, 11:00 A.M. TO 12:00 NOON

Session 4aNSc

Noise: Children’s Perception of Noise

Irene van Kamp, Cochair
Ctr. for Environ. Health Res., RIVM, P.O. Box 1 Postbus 10, Bilthoven 3720 BA, Netherlands

Janina Fels, Cochair
Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr. 50, Aachen 52074, Germany

Invited Papers

11:00

4aNSc1. Current perspectives on children’s auditory perception and consequences of noise exposure effects. Sofie Fredriksson (Occupational and Environ. Medicine, Univ. of Gothenburg, Box 414, Gothenburg 40530, Sweden, sofie.fredriksson@gu.se), Janina Fels (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany), and Kerstin Persson Waye (Occupational and Environ. Medicine, Univ. of Gothenburg, Gothenburg, Sweden)

Exposure to high sound pressure levels is well known to cause auditory damage, regardless of age. There is however limited knowledge of the effects on hearing due to noise exposure early in life. In addition, no well-established model is used to describe how children perceive and experience their sound environment compared to adults. New studies of children’s hearing have revealed different directivity pattern especially at high frequencies given by the head-related transfer functions due to the anthropometric data of the children and also an ear canal resonance at considerable higher frequencies compared to adults. Recent studies also describe children feeling a great deal of discomfort when exposed to sounds with high frequency characteristics. Children today are exposed to high sound levels from an early age at preschool, school and during leisure time. Few studies have looked at general health effects or hearing in particular. It is

being discussed whether age related hearing loss, regarded as an inevitable part of life, to a large extent may be caused by a lifetime of noise exposure starting early in life. This paper will review available studies on noise induced hearing damage among children and give suggestions for future studies within this field.

11:20

4aNSc2. Laboratory study on effects of environment noise on children's short-term memory. Hui Ma and Shengnan Gong (School of Architecture, Tianjin Univ., No. 92 Weijin Rd., Nankai District, Tianjin 300072, China, mahui@tju.edu.cn)

In laboratory settings 18 children of 6-7 years old were asked to finish series of short-term memory tasks under the different sound situation. The noise stimuli used in the experiment were both road traffic noise and low frequency noise of LAeq.2min = 40, 45, and 50 dB to simulate the indoor sound condition when the children at school or at home. The result showed both road traffic noise and low frequency noise had influence on children's task achievement and significant adverse effects on subjective noise annoyance evaluation. Comparing with the linear dose-response relationship of road traffic noise, the effects brought by low frequency noise to children's short-term memory work and noise annoyance evaluation was more complicated. There was an interesting trend that higher noise annoyance was evaluated, the higher task marks was obtained. As for the gender difference, boys showed more sensitivity to low frequency noise than girls.

11:40

4aNSc3. The effects of noise disturbed sleep on children's health and cognitive development. Irene van Kamp (Ctr. for Environ. Health Res., RIVM, P.O. Box 1 Postbus 10, Bilthoven, Utrecht 3720 BA, Netherlands, irene.van.kamp@rivm.nl), Anita Gidlöf-Gunnarsson, and Kerstin Persson Waye (Occupational Medicine and the Environment, Gothenburg Univ., Gothenburg, Sweden)

Undisturbed sleep is essential for physiological and psychological health. Children have a special need for uninterrupted sleep for growth and cognitive development. Noise is an environmental factor that affects most children. In addition to noise in schools and preschools, many are exposed to potentially disturbing traffic related noise at night. The knowledge of how children's health, wellbeing and cognitive development is affected by noise disturbed sleep due to road traffic is very incomplete. Nor do we know how children are able to handle noisy situations (coping) and if a learned noise-related behavior in the long term has a negative influence on children's health and learning. The need for a restorative home environment can be particularly important when the child is simultaneously exposed to noise in the school environment. Moreover, it has been shown that although children are less sensitive for awakenings and sleep cycle shifts due to nighttime exposure, they are more sensitive for physiological effects such as blood pressure reactions and related motility during sleep. This paper aims to review existing knowledge on how children's health and cognitive development are affected by noise in the home and school environment, with special focus on the importance of noise-disturbed sleep.

THURSDAY MORNING, 6 JUNE 2013 — 519B, 9:00 A.M. TO 11:40 A.M.

Session 4aPA

Physical Acoustics: Nonlinear Acoustics I

Kent L. Gee, Chair
Brigham Young Univ., N243 ESC, Provo, UT 84602

Contributed Papers

9:00

4aPA1. Experimental verification of a wave-vector-frequency-domain nonlinear acoustic model. Yun Jing (Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695, yjing2@ncsu.edu) and Jon Cannata (HistoSonics, Ann Arbor, MI)

In this paper, a recently developed wave-vector-frequency-domain method for nonlinear wave propagation is verified by underwater experiments. A specially designed focused transducer was used to generate short high intensity pulses. 2D scans were conducted at a pre-focal distance, which were later used as the input to the numerical model to predict the acoustic field at focal and post-focal zones. Adaptive attenuation was introduced to reduce the Gibbs effect. Graphic processing units (GPU) were also employed to speed up the computation. Good agreement was observed between the simulation and experiment.

9:20

4aPA2. Ballistic shock wave localization estimation of shooter position and velocity using difference of time of arrival DTOA algorithm in orthogonally arranged discrete acoustic arrays. Murray S. Korman (Dept. of Phys., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu) and Antal A. Sarkady (Dept. of Elec. and Comput. Eng., U.S. Naval Acad., Annapolis, MD)

A mathematical algorithm was developed using the difference in time of arrival, DTOA, of the ballistic shock wave cone at each position of an N-element array. The array is thought to be made up of orthogonally arranged discrete wide-band point like microphone or piezo-electric elements. The algorithm is used to estimate the parameter space involving both the displacement of the shooter with respect to the array and the velocity of the projectile. The algorithm utilizes a nonlinear least squares parameter fit of a

difference in time of arrival equation that involves a lengthy Taylor series expansion of the exact "difference in time of arrival" theoretical equation. Results are presented showing that the model has good versatility in estimating displacement (location) and velocity in simulated trials that are presented. The method does not require any muzzle blast wave information or knowledge of the absolute time of arrival of the shock wave. Therefore, the orthogonal array located far from the shooter will be able to estimate the localization parameters as long as the projectile miss distance is not so large that ambient noise does not mask the detection of the shock wave pressure signature.

9:40

4aPA3. Compressively excited acoustic wave propagation in a three-dimensional channel bifurcated by an elastic partition. Katherine Aho and Charles Thompson (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, katherine_aho@student.uml.edu)

Fluid motion resulting from compression of the exterior walls of a three-dimensional channel that is axially partitioned by a flexible membrane is examined. The axial variation in the acoustic impedance of the partition gives rise to a pressure gradient across the partition and generates an evanescent field at its surface. The effects of these evanescent modes and their impact on spatial wavenumber selectivity are of particular interest. The Green's function for the pressure is given for the case of low fluid compressibility and channel high aspect ratio. In this limit, it is shown the evanescent modes may be modeled in terms of generalized functions. The complete solution of the pressure field inside the channel will be shown. Implication of the results on the development a MEMS transducer in which the displacement of membrane is used as the method of transduction is presented. [NSF Grant 0841392.]

10:00

4aPA4. Propagation of N-waves in a turbulent and refracting atmosphere with ground effects (laboratory-scale experiment). Sébastien Ollivier, Edouard Salze, and Philippe Blanc-Benon (LMFA - UMR CNR 5509, Univ. Lyon 1, Centre Acoustique - Ecole Centrale de Lyon, 36 avenue Guy de Collongue, Ecully 69134, France, sebastien.ollivier@univ-lyon1.fr)

Sound propagation in a refracting atmosphere leads to the formation of a geometrical shadow zone close to the ground, and of an illuminated zone above a limiting ray. Without turbulence, acoustic waves can be diffracted into the shadow zone close to the ground. When propagating through a turbulent atmosphere, it is known that waves can be distorted, scattered, or focused. In order to investigate how turbulence modifies the pressure field into the shadow zone, a well controlled laboratory-scale experiment has been performed. An electrical spark source is used to generate short duration (20 μs) and high pressure (1500 Pa) N-waves. A convex surface models the effect of an upward refracting atmosphere, and a heating grid generates thermal turbulence (1% fluctuations of refraction index). To compute statistics of wave parameters variation, seven 1/8 in. microphones have been used to record 2000 waves at each position after propagation through the turbulent field. Wave parameters (peak pressure, rise time) obtained with turbulence are compared to data obtained without turbulence. Results show that turbulence scatters sound into the shadow zone, which increases significantly the noise level.

10:20

4aPA5. The influence of a focused acoustic field on mass-transfer processes at a heterogeneous boundary. Dmitry Kasyanov (Radiophysical Res. Inst., 25/12a Bolshaya Pecherskaya St., Nizhny Novgorod 603950, Russian Federation, da_kasyanov@nirfi.sci-nnov.ru)

An experiment is conducted on estimating the velocity of a Schlichting boundary flow arising when a focused field falls on a rigid boundary in a liquid. Also temperature increase into the acoustic boundary layer is experimentally estimated in this situation using optic means. The velocity of a small-scale Schlichting flow is determined by an indirect method from the characteristics of the cocurrent Rayleigh flow using the particle image velocimetry method. The velocity of the Schlichting flow attained in experiments gives the possibility of significantly accelerating mass-transfer processes at a heterogeneous boundary, which is confirmed by experimental results on acoustic intensification of rapid growth of salt monocrystals conducted under strictly controlled laboratory conditions. It is shown the increase in the temperature of the boundary layer is insufficient for significant retardation of the crystal growth.

10:40

4aPA6. Statistical properties of nonlinear N-wave propagating in thermal or kinematic turbulence. Petr V. Yuldashev (Dept. of Acoust., Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation and Lab. de Mécanique des Fluides et d'Acoustique, UMR CNRS 5509, Ecole Centrale de Lyon, 119991, Russian Federation, Moscow, Leninskie Gory, Moscow 119991, Russian Federation, petr@acs366.phys.msu.ru), Sébastien Ollivier (Lab. de Mécanique des Fluides et d'Acoustique, UMR CNRS 5509, Université de Lyon, Ecole Centrale de Lyon, Ecully, France), Vera Khokhlova (Dept. of Acoust., Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation), and Philippe Blanc-Benon (Lab. de Mécanique des Fluides et d'Acoustique, UMR CNRS 5509, Ecole Centrale de Lyon, Ecully, France)

Nonlinear propagation of high amplitude N-wave through turbulent layer is studied using 2D KZK-type nonlinear parabolic equation. The incident acoustic wave is assumed to have a plane wavefront and the waveform is a classical symmetrical N-wave. The turbulent layer is synthesized using a method of random Fourier modes. The modified von Karman spectra with the same values of outer and inner scales are considered for both scalar-type (temperature fluctuations) and vector-type (velocity fluctuations) turbulent fields. The rms value of the refraction index fluctuations μ is varied as a parameter. It is shown that statistical characteristics of N-wave propagating in vector-type or scalar-type turbulent fields are equivalent when μ in scalar-type turbulence is almost two times greater. The distance of most probable occurrence of caustics obtained in the KZK simulations, which account for diffraction effects is demonstrated to be inversely proportional to μ while the geometrical acoustics approach predicts the inverse proportionality to 2/3 power of μ. An effect of the initial N-wave amplitude on the peak pressure in random caustics is analyzed. The enhancement of focusing efficiency is observed for moderate initial amplitudes, whereas strong nonlinearity is shown to reduce pressure amplitudes. [Work supported by PICS RFBR 10-02-91062/CNRS 5603 grants.]

11:00

4aPA7. The numerical simulation of propagation of intensive acoustic noise. Igor Demin, Gurbatov Sergey, and Pronchatov-Rubtsov Nikolay (Acoustics, Nizhny Novgorod State Univ., 23, Gagarin Ave., Nizhny Novgorod 603950, Russian Federation, phdem56@gmail.com)

The propagation of intensive acoustic noise is of fundamental interest in nonlinear acoustics. Some of the simplest models describing such phenomena are generalized Burgers' equations for finite amplitude sound waves. An important problem in this field is to find the wave's behavior far from the emitting source for stochastic initial waveforms. The method of numerical solution of generalized Burgers equation proposed is step-by-step calculation supported on using Fast Fourier Transform of the considered signal. The general idea is to keep only Fourier image of concerned signal and update it recursively (in space). For simulating the wave evolution we used 4096 (212) point realizations and took averaging over 1000 realizations. Also the object of the present study is a numerical analysis of the spectral and bispectral functions of the intense random signals propagating in non-dispersive nonlinear media. The possibility of recovering the input spectrum from the measured spectrum and bispectrum at the output of the nonlinear medium is discusses. The analytical estimations are supported by numerical simulation. For two different types of primary spectrum evolution of jet noise were numerically simulates at a short distance and assayed bispectrum and a spectrum analysis of the signals.

11:20

4aPA8. Radiation of finite-amplitude waves from a baffled pipe. K. J. Bodon, Derek C. Thomas, Kent L. Gee, Rachael C. Bakaitis (Physics, Brigham Young Univ., Provo, UT 84602, joshuabodon@gmail.com), David T. Blackstock, and Wayne M. Wright (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The radiation of finite-amplitude waves from the open end of a baffled, circular pipe is considered as a direct continuation of work begun by Blackstock and Wright more than three decades ago [Kuhn *et al.*, J. Acoust. Soc. Am. **63**, (1978), S1, S84]. Band-limited Gaussian noise, as well as 1, 1.5, and 2 kHz sinusoidal pulses, with initial sound pressure levels ranging from

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552 to 1186 Pa, have been propagated down a 6.1 m pipe, whose open end (5.1 cm inner diameter) has been placed off-center in a large rectangular baffle. As the steepened or shock-like waves exit the pipe, the measured waveforms are comprised of sharp impulses that are delta function-like in nature, particularly on axis. Although linear piston theory predicts similar waveform shapes, there is also evidence that nonlinear propagation of these impulses, which can exceed 150 Pa near the pipe opening, is occurring. http://asadl.org/jasa/resource/1/jasman/v63/iS1/pS84_s5

THURSDAY MORNING, 6 JUNE 2013 514ABC, 8:55 A.M. TO 12:00 NOON

Session 4aPPa

Psychological and Physiological and Animal Bioacoustics: Biomechanics of Hearing

Sunil Puria, Chair
Mech. Eng., Stanford Univ., 496 Lomita Mall, Stanford, CA 94305

Chair's Introduction—8:55

Invited Papers

9:00

4aPPa1. Mechano-acoustical measurement and modeling of the outer and middle ear. W. Robert J. Funnell (BioMedical Eng. and Otolaryngol. – Head & Neck Surgery, McGill Univ., 3775, rue University, Montreal, QC H3A 2B4, Canada, robert.funnell@mcgill.ca)

Mechano-acoustical measurement and modeling have evolved together. Most early measurements of the behavior of the outer and middle ear produced either spatial averages or single-point observations, which were amenable to modeling with uniform transmission lines and lumped circuits. A major step forward was the measurement of displacement patterns on the eardrum, which called for the use of finite-element models. Other major experimental steps forward included measuring spatial sound-pressure distributions, 3-D displacement patterns, and intracochlear pressures. Use of the finite-element method made it desirable to obtain detailed 3-D shape measurements, which were made much easier by the introduction of magnetic-resonance microscopy and x-ray microCT. The finite-element method has also made it possible to exploit measurements of material properties, and several different approaches have been used recently for making such measurements. The greatest challenges may be in dealing with very small dimensions and non-linear visco-elastic behavior. There is a need for more and better 3-D multipoint vibration measurements, and for material-property measurements that are more localized and that span a broader frequency range. Important directions for modeling include better use of available shape and material-property data, more attention to experimental animals and to variability, and better integration with cochlear models.

9:20

4aPPa2. Carhart's notch: A window into mechanisms of bone-conducted hearing. Namkeun Kim, Charles R. Steele, and Sunil Puria (Dept. of Mech. Eng., Stanford Univ., 496 Lomita Mall, Stanford, CA 94305, chasst@stanford.edu)

Otosclerosis is a disease process of the ear that stiffens the stapes annular ligament and results in footplate immobilization. This produces a characteristic loss in bone-conducted (BC) hearing of about 20 dB between 1 and 2 kHz, known as "Carhart's notch," for which the specific mechanisms responsible have not yet been well understood. In this study, it is hypothesized that this observed pattern of hearing loss results from interactions between compressional and inertial mechanisms of BC hearing. Differences in the basilar-membrane velocity between a normal and otosclerotic human ear were calculated in response to compressional vibration of the cochlear capsule, translational vibration of the skull bone in various directions, and combinations of the two, using an anatomically accurate 3-D finite element model of the middle ear, cochlea, and semicircular canals. Compressional and inertial BC stimuli were found to both be necessary to capture the full behavior of clinical data, with the compressional component dominating below 0.75 kHz, the inertial component dominating above 3 kHz, and the notch between 1 and 2 kHz resulting from the suppression of an ossicular resonance due to stapes fixation. [Work supported by grant R01-DC07910 and R01-DC05960 from the NIDCD of NIH.]

9:40

4aPPa3. Fast waves, slow waves and cochlear excitation. Elizabeth S. Olson (OTO/HNS & Biomed. Eng., Columbia Univ., 630 W.168th St., P&S 11-452, New York, NY 10032, eao2004@columbia.edu)

In idealized cochlear models [especially, Peterson and Bogert, J. Acoust. Soc. Am. (1950)], intracochlear pressure is decomposed into two modes, the compression pressure (fast mode) and the traveling wave pressure (slow mode). Because the cochlear fluid is nearly incompressible, only the slow mode leads to significant motion. In the real cochlea, additional fast modes exist. These evanescent modes are similar to the traveling wave mode in driving significant fluid and tissue displacement. They are present in the region of the cochlear windows, where the anatomy is not the ideal symmetric structure of basic cochlear models. Evanescent modes also have emerged in experimental work in which cochleostomies are made in the apex. At high stimulus level fast modes are able excite hair cells, leading to auditory nerve responses. However, fast mode motions do not seem to be amplified by the cochlear amplifier. This observation supports the concept that the amplifier relies on traveling wave curvature, as has been proposed in cochlear models [for example, Yoon *et al*., Biophys. J. (2011)]. I will review experimental and theoretical results on the fast and slow waves, and propose experiments in which wavelength-based theories of amplification could be tested.

10:00–10:20 Break

10:20

4aPPa4. Comparative auditory biomechanics probed by otoacoustic emissions. Christopher Bergevin (Phys. & Astronomy, York Univ., 4700 Keele St., Petrie 240, Toronto, ON M3J 1P3, Canada, cberge@yorku.ca), Wei Dong (Columbia Univ., New York, NY), Laurel Carney (Univ. of Rochester, Rochester, NY), David S. Velenovsky, Kevin E. Bonine (Univ. of Arizona, Tucson, AZ), and James L. Jarchow (Sonora Veterinary Group, Tucson, AZ)

Since Kemp's discovery in 1978, otoacoustic emissions (OAEs) have provided valuable scientific and clinical tools for the study of the ear. For example, OAEs can provide objective measures of sensitivity and selectivity over the frequency range of "active" hearing. Given the universality of OAEs across the kingdom Animalia, comparative studies can reveal how various morphological factors affect peripheral auditory transduction and thereby what information is encoded for higher level cognition. Motivated by the complexity of cochlear mechanics and the many unknowns that currently exist, the present study describes OAEs stemming from two non-mammalian groups whose auditory periphery is relatively simpler than that of mammals: several lizard genera (Heloderma, Tiliqua, Agama, and Tupinambis) that exhibit significant relative differences in tectorial membrane structure, and a highly vocal bird species (*Melopsittacus undulatus*). By utilizing recent improvements in OAE measurement and analysis strategies combined with quantitative anatomical measures (e.g., number of hair cells), these data shed new light upon emission generation mechanisms and how such tie back to a given species' ability to encode ecologically relevant sounds. Furthermore, these data serve to inform theoretical models of auditory biophysics by clarifying what roles various morphological features do (or do not) play.

10:40

4aPPa5. Active processes and sensing in the cochlea. Karl Grosh and Julien Meaud (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109-2125, grosh@umich.edu)

One key question in the biophysics of the mammalian cochlea is determining the relative contribution to cochlear amplification by the two active processes present in the outer hair cell, namely prestin-based somatic motility and hair bundle (HB) motility. In the biological cochlea, these two effects are intimately coupled as HB force generation is linked to calcium-dependent adaptation of the transduction current and somatic force generation is driven by the depolarization caused by the same transduction current. To separate these effects, we construct a global mechanical-electrical-acoustical mathematical model of the cochlea. The global cochlear model is coupled to linearizations of nonlinear somatic motility and HB motility. We find that the active HB force alone is not sufficient to power high frequency cochlear amplification while somatic motility can perform this task. We discuss the limitations to this mathematical approach along with existing seminal experiments and proposed experiments (both in the cochlea and in the auditory nerve) to map future directions for uncovering the micromechanical contributions to the system level response of the cochlea. Describing the relation between the microfluidic flow stimulating the inner hair cell HB and the active processes in the cochlea and is an important future research direction.
[Support: NIH-NIDCD.]

11:00

4aPPa6. Weak lateral coupling between stereocilia of mammalian cochlear hair cells requires new stimulus methods to study the biomechanics of hearing. K. Domenica Karavitaki (Neurobiology, Harvard Med. School, 220 Longwood Ave., Goldenson 443, Boston, MA 02115, dkaravitaki@hms.harvard.edu), Paul D. Niksch, and David P. Corey (Neurobiology, Harvard Med. School and Howard Hughes Medical Inst., Boston, MA)

The forces felt by different transduction channels in a bundle depend critically on how well stereocilia remain cohesive during deflection. In the bullfrog saccule, sliding adhesion mediated by horizontal top connectors (HTC) confers coherent motion to hair cell stereocilia and parallel gating to all transduction channels. In cochlear inner and outer hair cells (IHCs and OHCs), the mature complement of HTC is established by postnatal day 12; they extend between adjacent stereocilia of both rows and columns. Contrary to our expectation that bundle cohesion should be robust in all directions, our experiments on gerbil cochleas show that lateral coupling among stereocilia of the tallest row is weak. These findings suggest that the function and molecular composition of the HTC in the cochlea are different from those in bullfrog hair cells. They also raise concern for current stimulus methods, which involve glass probes that are often small compared to the bundle width. Our data suggest that only stereocilia in contact with the probe are stimulated, and delivery of the stimulus to the remaining stereocilia is weak and inhomogeneous. To mimic the OHC stimulus delivered by the overlying tectorial membrane *in vivo*, we are developing new stimulation technologies.

11:20

4aPPa7. The elusive hair cell gating spring, a potential role for the lipid membrane. Jichul Kim, Peter M. Pinsky, Charles R. Steele, Sunil Puria (Dept. of Mech. Eng., Stanford Univ., Stanford, CA), and Anthony J. Ricci (Dept. of Otolaryngol.-HNS, Stanford Univ., 496 Lomita Mall, Stanford, CA 94305, tricci@ohns.stanford.edu)

Deflection of auditory hair cell hair bundle results in a nonlinear (i.e., non Hookean) force-displacement relationships whose molecular mechanism remains elusive. A gating spring model posits that mechanosensitive channels are in series with a spring such that channel opening puts the activation gate in series with the spring, thus reducing spring extension until further stimulation is provided. Here we present a theoretical analysis of whether the lipid membrane might be the source of nonlinearity. A hair bundle kinematic model is coupled with a lipid membrane model that includes a diffusible compartment into which the tip-link embeds and a minimally diffusive reservoir pool. Using physiological parameters, this model was capable of reproducing nonlinear force-displacement plots, including a negative stiffness component but required a standing tip-link tension. In addition, this model suggests the mechanotransducer channel is most sensitive to curvature forces that are located within 2 nm of the tip-link. [Work supported in part by Grant Nos. R01-DC07910 and R01-DC03896 from the NIDCD of NIH and by The Timoshenko fund from Mechanical Engineering Department at Stanford University].

11:40–12:00 Panel Discussion

Session 4aPPb

Psychological and Physiological Acoustics: Binaural Hearing (Poster Session)

Pavel Zahorik, Chair
Psychol. and Brain Sci., Univ. of Louisville, Life Sci. Bldg. 347, Louisville, KY KY

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.

4aPPb1. Amplitude modulation detection by human listeners in reverberant sound fields: Effects of prior listening exposure. Pavel Zahorik and Paul W. Anderson (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Life Sciences Bldg. 347, Louisville, KY, pavel.zahorik@louisville.edu)

Previous work [Zahorik *et al.*, POMA, **15**, 050002 (2012)] has reported that for both broadband and narrowband noise carrier signals in a simulated reverberant sound field, human sensitivity to amplitude modulation (AM) is higher than would be predicted based on the acoustical modulation transfer function (MTF) of the listening environment. These results may be suggestive of mechanisms that functionally enhance modulation in reverberant listening, although many details of this enhancement effect are unknown. Given recent findings that demonstrate improvements in speech understanding with prior exposure to reverberant listening environments, it is of interest to determine whether listening exposure to a reverberant room might also influence AM detection in the room, and perhaps contribute to the AM enhancement effect. Here, AM detection thresholds were estimated (using an adaptive 2-alternative forced-choice procedure) in each of two listening conditions: one in which consistent listening exposure to a particular room was provided, and a second that intentionally disrupted listening exposure by varying the room from trial-to-trial. Results suggest that consistent prior listening exposure contributes to enhanced AM sensitivity in rooms, but that it is not the sole determinant of the enhancement. [Work supported by the NIH/NIDCD.]

4aPPb2. Spatial consistency as a cue for segregation and localization. Brian Simpson (Battlespace Acoust., Air Force Res. Lab., 2610 Seventh St., Area B, Bldg. 441, Wright-Patterson AFB, OH 45433, brian.simpson@wpafb.af.mil), Robert Gilkey (Psychology, Wright State Univ., Dayton, OH), Eric Thompson (Ball Aerosp. and Technologies Corp., Wright-Patterson AFB, OH), Douglas Brungart (Audiol. and Speech Ctr., Walter Reed National Military Med. Ctr., Bethesda, MD), Nandini Iyer, and Griffin Romigh (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH)

Many real-world auditory scenes are dynamic and complex, with multiple sounds that may change location over time. In this experiment, we examined the ability of listeners to localize a spatially consistent target sound in a dynamic, spatially varying auditory scene. The target and masker stimuli were composed of sequences of 60-ms bursts of uncorrelated noise (2 to 16 bursts in duration) and differed only in their degree of spatial consistency. Specifically, each target burst within a sequence came from the same spatial location (which varied from trial to trial), whereas each masker burst within a sequence came from a different, randomly chosen spatial location. The listener's task was to localize the spatially-consistent sequence. Localization errors decreased by approximately 11° with each doubling of the sequence duration, and approached quiet performance with 16-burst sequences. Adding a second masker increased localization errors by approximately 14° overall. These results suggest that spatial information can be combined across multiple observations over time to identify and localize a spatially consistent target in a dynamic auditory scene. These data will be discussed in terms of the information obtained from each burst and the manner in which the information is combined across bursts.

4aPPb3. Head movement during horizontal and median sound localization experiments in which head-rotation is allowed. Daisuke Morikawa, Yuki Toyoda, and Tatsuya Hirahara (Faculty of Eng., Toyama Prefectural Univ., 5180 Kurokawa, Imizu, Toyama 939-0398, Japan, t074001@st.pu-toyama.ac.jp)

We measured subjects' head movements during horizontal and median sound localization experiments in which head-rotation was allowed in order to know how they move their heads to localize sound in a head rotation condition. The head movements in a head-rotation condition were measured while localizing 500-Hz low-pass noise, 12-kHz high-pass noise, and white noise. With regard to horizontal plane, sound localization became easier with head-rotation than head-still condition. All subjects turned their heads toward the presented sounds, yet they did not necessarily turn their heads to face the sound. The amount of head rotation was small for WN compared to that for LPN and HPN. Sound localization also became easier with head-rotation than head-still condition for the median plane. All subjects swung their heads right and left centering on 0°, no matter what the stimulus elevation angle was.

4aPPb4. Integration of auditory input with vestibular and neck proprioceptive information in the interpretation of dynamic sound localization cues. Janet Kim (Health and Rehabilitation Sci. Program, Western Univ., 62 Essex St, London, ON N6G 1B2, Canada, jkim2223@uwo.ca), Michael Barnett-Cowan (Brain and Mind Inst., Western Univ., London, ON, Canada), and Ewan Macpherson (National Ctr. for Audiol., Western Univ., London, ON, Canada)

To determine the front/back location of a sound source via head rotation, the auditory system must integrate sensorimotor information about head motion with the dynamic acoustic cues resulting from motion of the source relative to the head. In order to determine the influence of vestibular and proprioceptive cues on processing of dynamic acoustic cues, we measured, in active, passive, and counter-rotation conditions, listeners' ability to discriminate front/rear locations of low-frequency sounds not accurately localizable without head motion. Targets were presented over headphones during head rotations using dynamic virtual auditory space methods. In the active condition, the subject performed a head-on-body rotation, which provided vestibular and neck proprioceptive information. In the passive condition, proprioceptive information was minimized by whole-body rotation with no neck movement using a motorized rotating chair. In the counter-rotation condition, the subject performed a head-on-body rotation while the body was counter-rotated, which minimized vestibular input by keeping the head still in space. Dynamic acoustic cues corresponded to the head-on-body angle. Discrimination was accurate in active and passive conditions, but near chance under counter-rotation, suggesting that vestibular inputs are

necessary and sufficient to inform the auditory system about head movement, whereas proprioceptive cues are neither necessary nor sufficient.

4aPPb5. Falling stars: Acoustic influences on meteor detection. Darlene Edewaard and Michael S. Gordon (Dept. of Psych., William Paterson Univ., 300 Pompton Rd., William Paterson U., Wayne, NJ 07470, edewaardd@student.wpunj.edu)

As particles enter the earth's atmosphere they produce a burst of electromagnetic energy, including visible and radio-wave emissions. Consequently, just as meteors can be detected visually in the night sky they can be "heard" using radio telescopes. The current project investigated the potential influence of these audio signals on meteor detection. Anecdotally, and in related research, it has been found that auditory signals can enhance or even alter visual perception of objects. The current project examined the specific effects of accompanying auditory signals on the detection of meteors. Meteors present an interesting case of audiovisual integration in that detection paradigms often entail extended vigilance and extremely brief, yet brilliant astronomical events. Experiments specifically investigated how auditory signals that varied in spectra influenced changes in visual magnitude and duration judgments of meteors. In addition, research targeted how extraneous auditory cues during a vigilant meteor search might contribute to false judgments. Results are described in terms of audiovisual integration and the relation of perceptual mechanisms to meteor detection.

4aPPb6. Time-to-arrival discrimination of multiple sound sources. Michael S. Gordon, Darlene Edewaard, and Matthew Pacailler (Psychology, William Paterson Univ., 300 Pompton Rd., Wayne, NJ 07470, gordonm10@wpunj.edu)

Previous research in auditory detection of time-to-arrival (TTA) has tended to focus on single sound sources approaching a listener. However, visual studies of TTA have suggested that there may be different perceptual strategies employed with respect to single versus multiple objects on approach. The current research is designed to directly address the capacity and informational support of listeners to make determinations on the TTA of multiple sound sources. Initial experimentation evaluated the capacity for discrimination between a stationary and moving sound source. Additional studies evaluated multiple moving sound sources, with direct manipulation of their spectra to determine how various acoustic qualities contribute to TTA determinations. Results suggest the relative influence of intensity, frequency, and sound source dynamics as they facilitate competitive discriminations between approaching sound sources. Additional conclusions demonstrate changes in the use of spectral cues for lower and higher levels of auditory clutter as they might impact TTA.

4aPPb7. Performance of a highly directional microphone array in a multi-talker reverberant environment. Sylvain Favrot, Christine R. Mason, Timothy M. Streeter (Dept. of Speech, Lang. & Hearing Sci. and Hearing Res. Ctr., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, sfav@bu.edu), Joseph G. Desloge (Sensimetrics Corp., Malden, MA), and Gerald Kidd (Dept. of Speech, Lang. & Hearing Sci. and Hearing Res. Ctr., Boston Univ., Boston, MA)

A visual guided hearing aid (VGHA) has recently been developed, which uses an eye tracker to steer the "acoustic look direction" (ALD) of a beamforming microphone array. The current study evaluates the performance of this highly directional microphone in providing spatial release from masking (SRM) under acoustically dry and reverberant conditions. Four normal-hearing subjects participated in a speech intelligibility test with collocated and spatially separated speech maskers when listening either through the microphone array or through KEMAR to simulate "natural" binaural conditions. The results indicated that near-normal SRM was achieved by listening through the VGHA in both environments. In the acoustically dry condition, SRM was similar to the measured signal-to-noise ratio (SNR) gain from the microphone array. However, in the reverberant condition, subjects showed significantly greater SRM than predicted from the measured SNR gain from the array. This is consistent with the measured improvement in SNR for the early part of the room impulse response for the target but not for the spatially separated maskers. This indicates that in some reverberant conditions the microphone array provides substantial source selection benefits.

4aPPb8. Head tracking and source localization in reverberant cocktail party scenarios. Anthony Parks and Jonas Braasch (Program in Architectural Acoust., Rensselaer Polytechnic Inst., 126 2nd St. Apt. 12, Troy, NY 12180, abstractpoetry@gmail.com)

From experience and observation, it is known that listeners actively explore their auditory environment through a variety of head movement strategies. Beyond the resolution of front-back confusions, little is known about the mechanisms by which head movement enables listeners perform a broad range of auditory scene analysis tasks. In this experiment, an attempt was made to look at one of these tasks: how well listeners can track a source by head movement. A three-talker paradigm was utilized in a headphone-based head-tracked virtual environment spatialized with HRTFs. The task involves a target that moves from trial to trial with two stationary interferers (male target, two female interferers, all speaking phonetically balanced sentences). The listener is prompted to move her head such that she believes she is facing the target. The two independent conditions are three levels of reverberation (anechoic, strong early reflections, strong late reflections) and seven target azimuth angles (from +45 to −45 degrees in steps of 15 degrees), while the measured responses include facing accuracy as a percentage and number of times the playback button is hit before each trial is completed (to indicate task difficulty). Results are discussed within the context of both room acoustics and perception.

4aPPb9. Differences in masked localization of speech and noise. Inseok Heo (Elec. and Comput. Eng., Univ. of Wisconsin - Madison, Madison, WI), Lynn Gilbertson, An-Chieh Chang, Jacob Stamas, and Robert Lutfi (Dept. of Commun. Sci. and Disord., Univ. of Wisconsin - Madison, Madison, WI 53705, lrjohnson1@wisc.edu)

Dichotic masking studies using noise are commonly referenced in regard to their implications for "cocktail party listening" wherein target and maskers are speech. In the present study masker decision weights (MDWs) are reported suggesting that speech and noise are processed differently in dichotic masking. The stimuli were words or Gaussian-noise bursts played in sequence as masker-target-masker triads. The apparent location of words (noise bursts) from left to right was varied independently and at random on each presentation using KEMAR HTRFs. In the two-interval, forced-choice procedure listeners were instructed to identify whether the second-interval target was to the left or right of the first. For wide spatial separations between target and masker noise-MDWs were typically negative, indicating that target location was judged relative to the masker. For small spatial separations between target and masker noise-MDWs were typically positive, suggesting that target location was more often confused with the masker. For both spatial separations, however, word-MDWs were close to zero, implying that the masker served to distract attention from the target without itself being given significant weight. The results are consistent with an interpretation in which spectral dissimilarities among words generally serve to reduce confusions and relative comparisons among words.

4aPPb10. Sound-localization performance with the hearing protectors. Veronique Zimpfer (Acoust. and Protection of Soldier, ISL, Inst. of Saint Louis, BP 70034, Saint Louis 68301, France, veronique.zimpfer@isl.eu) and David Sarafian (Unite Percept., IRBA, Institut de Recherche Biomédicale des Armées, Brétigny sur Orge, France)

Hearing-protection system that provide level-dependent sound attenuation can protect the ear against potentially damaging sounds (such as loud impulsive noises), while at the same time allowing the perception of moderate-level signals (such as speech). Such systems come in two forms: passive (nonlinear-attenuation earplugs) and active (talk-through system). This study sought to quantify the effect of these systems on spatial hearing. To this aim, sound-localization performance was measured in twenty subjects, with and without ear protectors on. Five protectors (two passives and three actives) were tested. The results showed significant increases in the proportions of errors during the use of one of the systems tested. To clarify the origin of this effect, "protected head-related transfer functions" (PHRTFs), i.e., HRTFs obtained with the ear-protectors on, were measured in the horizontal plane for each of the systems tested. The comparisons of these measures between PHRTFs with HRTFs were found to be in agreement with the subjective tests.

4aPPb11. Effects of targeted pinna occlusion on pinna/spectral cues to localization in the median plane. Alan Musicant (Psychology, Middle Tennessee St. Univ., Box X063, Murfreesboro, TN 37132, Alan.Musicant@mtsu.edu) and Robert Baudo (Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Auditory localization accuracy (in humans) in the median sagittal plane has been attributed, by some authors, to an effect of "spectral notches" that occurs in the frequency region of 4–8 kHz. Another possibility for decrements in vertical plane localization accuracy has been overlooked. Roffler and Butler (1967) and Hebrank and Wright (1974) both demonstrated that removal or absence of sound frequencies above about 8 to 10 kHz led to decrements in vertical plane localization accuracy. They did this by using carefully selected types of band pass or band limited noise. A reduction in accuracy of auditory vertical localization by occluding all or part of the pinna has been known for many years [Gardner and Gardner (1973)]. We have previously reported results that demonstrate disruption in accuracy with various partial pinna occlusions [ARO (2012)] that differs from results reported by Gardner and Gardner. We now have data that seems to indicate that the reduction in localization accuracy occurs, in part, because of disturbances in high frequency regions (above about 8 to 10 kHz) and that disruptions in "spectral notches" (4–8 kHz) has little to no effect upon vertical plane localization.

4aPPb12. Perceived elevation cued by images rotating in horizontal planes. Tianshu Qu, Haoze Sun, Ning Wang, Xihong Wu (Key Lab. of Machine Percept. (Ministry of Education), Center for Information Science, Peking Univ., Beijing 100871, China, qutianshu@gmail.com), and William M. Hartmann (Dept. of Phys. and Astronomy, Michigan State Univ., East Lansing, MI)

The sense of elevation perceived by human listeners is normally attributed to high-frequency spectral structure above 4 kHz, caused by anatomical filtering. Our research began with the conjecture that elevation information might be available below 4 kHz when it is linked to a quasi-continuous set of azimuthal cues through measured (KEMAR) head related transfer functions. In an elevation discrimination test, listeners heard two successive azimuthal rotations of 90 degrees, each in a different horizontal plane. The elevations of the horizontal planes differed by as little as 10 degrees and as much as 110 degrees. Listeners reported which rotation had the higher elevation. Results of the rotation experiments were compared with the results from experiments with fixed azimuths, similar to those of Algazi *et al*. [J. Acoust. Soc. Am. **109**, 1110–1122 (2001)]. A rotation from 0 to 90 degrees led to negligible improvement compared to a fixed azimuth of 45 degrees. By contrast, a rotation from 45 to 135 degrees appeared to be particularly advantageous. [Work supported by the National Natural Science Foundation of China Grant Nos. 61175043 and the AFOSR.]

4aPPb13. On the ecological interpretation of limits of interaural time differences sensitivity. William M. Hartmann, Brad Rakerd, and Eric J. Macaulay (Phys.-Astronomy, Michigan State Univ., East Lansing, MI 48824, hartman2@msu.edu)

Human listeners, and other animals too, use interaural time differences (ITD) to localize pure tones, but this ability abruptly diminishes as the frequency of a pure tone increases. The diminished sensitivity appears to serve a useful function. It prevents the confusion that would otherwise arise from the large interaural phase differences that occur at high frequency as sound waves diffract around the head. Possibly this benefit offers an ecological explanation for the diminished sensitivity of the nervous system. However, comparison of the frequency dependence of ITD sensitivity, as measured in headphone experiments, and the frequency dependence of the physical phase shifts, as measured in an anechoic room, reveals a bad match between these two functions. The decrease in neural sensitivity to ITD is seen to be far too rapid, casting doubt on this form of ecological reasoning. If one wants to maintain an ecological context, it is more plausible to argue that our binaural architecture, with its neurophysiological limitations, evolved when our head diameters were smaller by as much as 50%. [Work supported by the AFOSR, 11NL002.]

4aPPb14. Threshold interaural time differences and the centroid model of sound localization. William M. Hartmann (Phys. and Astronomy, Michigan State Univ., East Lansing, MI 48824, hartman2@msu.edu) and Andrew Brughera (Hearing Res. Ctr., Boston Univ., Boston, MA)

The centroid display model of sound lateralization hypothesizes an array of brain-stem cells with wide ranges of best frequencies (f) and best interaural time delays (ITD, τ). The cells are distributed according to function $p(f,\tau)$, and images are lateralized according to the centroid of an excitation pattern on this array, the rate-ITD function. The centroid display was tested by calculations using model cells for the medial superior olive, as the origin of the rate-ITD function. The cells had excitatory inputs, membrane potential increments, and time constants established by physiological measurements. Cells were driven with realistic frequency-dependent synchrony. The predictions were compared to the measured frequency dependence of ITD thresholds for human listeners. It was found that for high frequencies, 750 Hz and greater, the model could successfully account for the thresholds by making appropriate adjustments to $p(f,\tau)$. For lower frequencies, the model greatly underestimated thresholds for any reasonable $p(f,\tau)$ because integration over the wide range of τ reduced the variability in the rate-ITD function to unreasonable values. It is concluded that the centroid model fails to account for human thresholds at low frequency. [Work supported by the AFOSR and NIDCD.]

4aPPb15. Sound source localization: Bandwidth and envelope. William Yost and Xuan Zhong (Speech and Hearing Sci., ASU, P.O. Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Human listeners were asked to locate six sound sources separated by 15o in the right quarter field. Sound sources were located in a sound-deadened room (reverberation time <90 ms) at the height of the listener's pinna 1.67 meters from the listener. In experiment 1, eight, 200-ms pure tones covering the frequency range from 250 to 7011 Hz were presented. In experiment 2, 200-ms noise bursts with different bandwidths (1/6, 1/3/, 1, and 2 octaves) at three center frequencies (250, 2000, and 4000 Hz) were presented. In experiment 3, 200-ms, 4000-Hz tones were presented with transposed envelopes with rates of 50, 100, 150, and 250 Hz. Several indicators of sound source localization performance were measured including root-mean-square (rms) error in degrees. RMS error decreased with increasing bandwidth from approximately 20 degrees for pure tones to approximately 6 degrees for 2-octave wide noises. RMS error depended on center frequency much more for narrow bandwidths than for broader bandwidths. RMS error decreased slightly from 50-Hz rate of modulation to 250-Hz rate of modulation. The data suggest that stimulus bandwidth is the primary variable effecting sound source localization performance in the free-field. [Research supported by an AFOSR Grant.]

4aPPb16. The Haas Effect—Lateral extent and perceptual weighting of localization cues. M. Torben Pastore and Jonas Braasch (Architectural Acoust., Rensselaer Polytechnic Inst., 4 Irving Place, Troy, NY 12180, m.torben.pastore@gmail.com)

The Haas effect is a well-known manifestation of the precedence effect. Originally, Haas measured the echo threshold as a function of the primary auditory event and its single reflection being equally loud. What is not well known is the lateral position of the lead/lag pair as a function of inter-stimulus interval and level difference between lead and lag. How robust the Haas effect is in the localization dominance region was investigated, adjusting the level difference between lead and lag, and using 200 ms band-passed noise presented dichotically over headphones. In addition, the onset and offset cues were removed for half the trials and left intact for the other half to investigate the roles of onset and offset cues versus ongoing cues. Lateral displacement of the auditory event was recorded with an acoustic pointer. Analysis of these results help to reveal the perceptual weighting of localization cues in the Haas effect.

4aPPb17. An original paradigm to investigate pure informational masking using complex tones. Axelle Calcus (Fonds de la Recherche Scientifique, F.R.S.-FNRS, 50, av. F. Roosevelt, Brussels 1050, Belgium, acalcus@ulb.ac.be), Trevor Agus (Institut d'Etude de la Cognition, D.C. Ecole Normale Supérieure, Paris, France), Cécile Colin (UNESCOG - CRCN, Université Libre de Bruxelles, Brussels, Belgium), Régine Kolinsky (Fonds de la Recherche Scientifique, F.R.S.-FNRS, Brussels, Belgium), and Paul Deltenre (Laboratoire de Neurophysiologie Sensorielle et Cognitive, Hôpital Brugmann, Brussels, Belgium)

Most usual speech masking situations induce both energetic and informational masking. Energetic masking (E.M.) arises because both signal and maskers contain energy in the same critical bands. Informational masking (I.M.) prevents the listeners from disentangling acoustical streams even

when they are well separated in frequency, and is thought to reflect central mechanisms. In order to quantify I.M. without E.M. contamination in complex auditory situations, target and maskers can by presented dichotically. However, this manipulation provides the listeners with important lateralization cues, which dramatically reduces I.M. Therefore, the current study aimed at restoring a fair amount of I.M. using complex tones in a new dichotic paradigm. Regularly repeating signals and random-frequency multitone maskers were presented dichotically, but switched from one ear to the other within a 10s sequence. Switches could either appear at a slow or rapid rate. We compared listeners' detection performance in these switching situations to that elicited in traditional diotic and dichotic situations. Results showed that the amount of I.M. induced when signal and maskers were rapidly switching throughout a sequence was significantly higher than in classical dichotic situations, and appeared to be comparable to the diotic listening situation. Therefore, this paradigm provides an original tool to evaluate auditory perception in situations of pure I.M. using complex tones.

4aPPb18. Release from masking through spatial separation in distance in hearing impaired listeners. Adam Westermann and Jörg M. Buchholz (National Acoust. Labs., Australian Hearing, 126 Greville St., Chatswood, NSW 2067, Australia, adam.westermann@nal.gov.au)

It is widely accepted that speech intelligibility improves as a speech signal and interfering masker are separated spatially in azimuth. In a previous study [Westermann *et al*. IHCON (2012)] a similarly strong improvement was found for normal hearing (NH) listeners when target and masker are separated in distance. In this study speech reception thresholds (SRTs) were measured for 16 hearing impaired (HI) listeners using the Listening in Spatialized Noise-Sentences Test (LiSN-S) and the Coordinate Response Measure (CRM). Acoustic scenarios were auralized via headphones using binaural room impulse responses recorded in an auditorium. In the first scenario, the target was presented at a distance of 0.5 m from the center of the listener's head and the interferer at a distance of 0.5 m or 10 m. In a second setup, the interferer's location was fixed and the target's location was varied. HI listeners showed a substantial release from masking as target and interferer were separated in distance. This effect was consistent for both LiSN-S and CRM, but less pronounced than for NH listeners. This study suggests that distance related cues play a significant role when listening in complex environments and are also to some extent available to HI listeners.

4aPPb19. Auditory streaming in cocktail parties and the extent of binaural benefit. Esther Schoenmaker and Steven van de Par (Acoust. Group, Univ. of Oldenburg, Carl von Ossietzkystrasse 9-11, Oldenburg D-26129, Germany, esther.schoenmaker@uni-oldenburg.de)

Studies that investigate the advantage of spatial separation of speakers in a cocktail party generally focus on one of two processing strategies. The first assumes a top-down mechanism in which the listener focuses attention on the known location of a target speaker. Glimpses of target speech are collected and combined to form an auditory stream. The second strategy makes use of interaural differences in perceptual input and exploits these in order to suppress interfering sounds. Equalization-Cancelation (EC) models typically follow this approach. In order to investigate the contributions of both mechanisms, a headphone experiment was conducted that explores auditory streaming based on binaural cues. Sequences of logatomes spoken by one target and two interfering speakers were presented. In this experiment a new type of stimuli was introduced in which the possibility to use binaural masking release cues was eliminated for each time-frequency interval (glimpse) while the localization cues of the dominating source were preserved. Thus, listeners could attend to spatially separated glimpses, but no EC processing was possible. The effect of the availability of masking release cues on successful streaming will be analyzed and discussed.

4aPPb20. Exploring auditory gist: Comprehension of two dichotic, simultaneously presented stories. Nandini Iyer, Eric R. Thompson, Brian D. Simpson (Air Force Res. Lab., 2610 Seventh St., Bldg. 441, Area B, Wright Patterson Air Force Base, OH 45433, Nandini.Iyer@wpafb.af.mil), Douglas S. Brungart, and Van Summers (Walter Reed National Military Med. Ctr., Bethesda, MD)

Cherry (1953) showed that when listeners were asked to selectively attend to one ear in a dichotic listening task, they were able to identify gross attributes of the signal in the unattended ear, suggesting that listeners may be able to capture the "gist" of an auditory stream even when they are asked to ignore it. This experiment explored the extraction of auditory "gist" by investigating the amount and nature of the semantic information stored in memory for later recall. In the experiment, listeners heard two dichotically presented stories; they were directed to: (1) listen to one of the two stories and answer yes-no questions about that story (Directed condition), (2) not directed (Undirected condition) and answer questions about one or both stories, and (3) listen to one of the stories and answer questions about the unattended story (Misdirected condition). Results suggest that listeners can recall the main ideas of both stories in the undirected attention condition significantly better than chance, but that their performance falls substantially below the level achieved in the directed attention condition. These findings are consistent with studies of visual gist processing, suggesting that global features, rather than details, are perceived even before attention is focused on the auditory streams.

4aPPb21. Factors influencing target detectability in realistic listening scenarios. Tobias Weller, Virginia Best, and Jörg M. Buchholz (National Acoust. Labs., 126 Greville St., Chatswood, NSW 2067, Australia, tobias.weller@nal.gov.au)

In psychoacoustics, there is an increasing demand for more realistic testing environments that better capture the real-world abilities of listeners and their hearing devices. However, there are significant challenges involved in controlling the detectability of relevant target signals in realistic environments. We conducted an extensive detection study in a simulated real-world environment to understand some of the important dimensions influencing detection. A multi-talker cafeteria scene was generated using Room simulation software and played back by means of a 3-D loudspeaker array. Detection thresholds for the target word "two" were measured adaptively for eight different target directions in the horizontal plane. Performance was then measured for fixed signal-to-noise ratios around these thresholds to obtain a psychometric function for each direction. To examine the effect of target-location uncertainty, psychometric functions were also measured with randomized target directions. Detection thresholds depended on the target direction, consistent with changes in signal-to-noise ratio caused by the head shadow. Target-location uncertainty increased thresholds globally by a small amount. These findings provide a framework for controlling the detectability of target sounds in future experiments aimed at measuring localization, identification, awareness, etc. in realistic listening environments.

4aPPb22. Spectral integration of interaural time differences in auditory localization. Nicolas Le Goff (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Lyngby 2800, Denmark, nlg@elektro.dtu.dk), Jörg M. Buchholz (National Acoust. Labs., Chatswood, NSW, Australia), and Torsten Dau (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, Lyngby, Denmark)

This study investigates how the auditory system integrates spatial information across frequency. In experiment 1, discrimination thresholds for interaural time differences (ITDs) were measured as a function of both reference ITD and center frequency (CF) of noises with bandwidth of one ERB. In addition, discrimination thresholds were also measured as a function of CF for different values of interaural coherence (IC) typical of sounds in realistic acoustic environments. For both high ICs and small reference ITDs, discrimination thresholds were lowest for CFs between 700 and 1000 Hz. For smaller ICs and larger reference ITDs, this dominance region shifted towards lower CFs. A conceptual localization model was developed that used the variance of the ITD thresholds to optimally weight the contribution of the individual frequency bands before spectral integration. In experiment 2, the model was tested by asking listeners to align a broadband noise signal with an ITD that was fixed across frequency onto a broadband noise target with different ITDs in individual 1 ERB-wide subbands. The results were consistent with both the model predictions and the shift of dominance range observed in experiment one.

4aPPb23. Lateralization of noise targets with interaural level differences presented within a noise interferer. Darrin Reed and Steven van de Par (Acoust. Group, Univ. of Oldenburg, Achternstrasse 23, Oldenburg 26122, Germany, darrinreed@hotmail.com)

The interaural level difference (ILD) of a lateralized target source is reduced when the target is presented together with background noise containing no ILD. It is unknown whether listeners simply use this reduced

aggregate ILD or are still able to utilize the target ILD in a lateralization task. Behavioral experiments revealed that the temporal asynchrony between the onsets/offsets of the target and the background noise resulted in the population of listeners actually perceiving a larger ILD than the target ILD. For synchronous onsets/offsets, however, the perceived ILD depended on the coherence of the background noise. With coherent background noise, the population of subjects perceived a reduced ILD near the aggregate ILD. In contrast, the population of subjects made a reasonable estimate of the target ILD when the background noise was diffuse. Implementation of an Equalization Cancellation model and taking the compensatory level equalization that yields the lowest output as an estimate for the ILD results in the reduced ILD of the aggregate stimulus being reported regardless of the background noise coherence. However, application of an appropriate normalization factor to the model's output results in a dependence on background noise coherence for ILD lateralization as seen in the behavioral experiments.

4aPPb24. Shifts in the judgment of distance to a sound source in the presence of a sonic crystal. Ignacio Spiousas, Pablo E. Etchemendy, Esteban Calcagno, and Manuel C. Eguia (Laboratorio de Acústica y Percepción Sonora (LAPSo), Universidad Nacional de Quilmes (UNQ), Roque Sáenz Peña 352, Bernal, Buenos Aires B1876BXD, Argentina, ispiousas@unq.edu.ar)

The ability of subjects to estimate the distance to a sound source in a room relies on the integration of a number of different cues: sound intensity level, direct-to-reverberant energy ratio, spectral content and binaural cues, among others. This work examines how the perception of auditory distance is modified for a particular sound field: the transmitted field of an acoustic source through a sonic crystal slab in a semi-reverberant room. A series of experiments were performed comparing the egocentric distance to a sound source passing and not passing through the sonic crystal, using an acoustical virtual environment whereby some of the auditory distance cues could be manipulated. The results obtained show that the presence of the sonic crystal introduces significant shifts on the auditory distance perception. These shifts are correlated with the spatial and spectral variation of the acoustical properties of the sonic crystal. Also, it was possible to determine the relative influence of the manipulated cues (intensity, binaural, and reverberation).

4aPPb25. Validating a binaural head for use in jury testing. Jeremy E. Charbonneau, Colin Novak, and Helen Ule (Mech. Eng., Univ. of Windsor, 1560 Dougall Ave., Windsor, ON N8X1S1, Canada, charbo6@uwindsor.ca)

A test procedure for use in loudness perception tests must be created to completely describes a phenomenon while at the same time minimizing jury listening fatigue. One contributor to this fatigue is the amount of time necessary for the test subject to experience all the required signals. Head and torso simulators have been used for years as a means to reliably quantify the acoustic performance of a product while avoiding the influence of listener bias and fatigue. This procedure not only controls the test parameters but also removes any human error that may occur. The purpose of this investigation is to qualify a head and torso simulator for use in loudness investigations. The objective of this experiment is to correlate the results from using this equipment to human subject results for high resolution experiments on directionality of loudness. Comparisons are also made from the directionality results at various listening angles including a listener facing a sound source.

4aPPb26. Development of an underwater binaural head model. Christopher A. Bailey and Neil L. Aaronson (Natural Sci. and Mathematics, The Richard Stockton College of NJ, 54 Mark Dr., High Bridge, NJ 08829, challenbailey@gmail.com)

The human brain has difficulty localizing sound in aquatic environments, where the acoustical properties of water greatly impede the mechanisms by which the brain interprets binaural signals. In this experiment, a hollow steel sphere with antipodal hydrophones is exposed to noise bursts in an underwater environment. The sphere can be filled with various materials to alter the apparatus' rigid qualities. Interaural time and level differences (ITDs and ILDs, respectively) are calculated from recordings of these noises and compared to a theoretical model for sound propagation around a non-rigid head in an effort to better characterize binaural hearing in underwater surroundings. While the theoretical behavior of sound diffracting around a rigid head has been well documented, the similar problem involving a flexible head has largely been left to experimental methods due to the computational complexity of the task. In the current study, a new computational model, capable of predicting ITDs and ILDs for sounds encountering a non-rigid sphere in diverse environments, is used. Both the model and the experiment will be introduced in this presentation. The findings have significant implications for the future development of reliable methods for improving sound localization in underwater environments, for instance for recreational divers.

4aPPb27. Can monaural temporal masking asymmetry explain the transient and/or ongoing precedence effect? Richard L. Freyman, Charlotte Morse-Fortier, Amanda M. Griffin (Dept. of Commun. Disord., Univ. of Massachusetts, 358 N. Pleasant St., Amherst, MA 01003, rlf@comdis.umass.edu), and Patrick M. Zurek (Sensimetrics Corp., Malden, MA)

Investigations of the precedence effect show that interaural differences within the first of two pairs of brief binaural sounds contribute more to lateralization than those within the second pair. The present study asked whether this phenomenon could be explained by asymmetries in monaural masking, and compared the results to a second experiment investigating the same question with the "ongoing" precedence effect. Leading and lagging stimuli were binaural pairs of 1-ms frozen noise bursts, with a 2-ms delay between pairs, and had ITDs of +500 and −500 ms, respectively. Detection thresholds for lead or lag in the presence of the other were assessed monaurally using a 4AFC task. Results showed that threshold for the lead was 12 dB better than that for the lag, on average. Ongoing stimuli were created by repeating the transient stimuli 63 times, with a new noise token on each repeat. Although the leading burst in each binaural pair contributes more to lateralization than the lagging burst, monaural thresholds for the leading bursts were not better than those for the lagging bursts. The results suggest that the transient precedence effect, but not the ongoing precedence effect, might be explained by temporal masking asymmetry. [Work supported by NIH DC01625].

4aPPb28. Effects of the stimulus spectrum on temporal weighting of binaural differences. G. Christopher Stecker (Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, cstecker@uw.edu)

The influence, or "perceptual weight" of binaural information typically varies over the duration of a brief sound, as characterized by the temporal weighting function (TWF). Here, TWFs were measured for binaural lateralization of Gabor click trains (GCT) varying in carrier frequency from 1 to 8 kHz, and of broadband noise-burst trains (NBT) with repeated ("frozen") or newly sampled ("fresh") noise across bursts. Interclick intervals (ICI) ranged from 2 to 10 ms. On each of many trials, human listeners judged the lateral position of a singly presented GCT or NBT, and indicated the position on a touch-sensitive display. Lateral positions varied with the overall interaural time (ITD, ranging +/− 500 μs) and level (ILD, ranging +/−5 dB) differences applied to each stimulus. Additional random variation in ITD (+/−100 μs) and ILD (+/−2dB) was applied independently to each click within a train. TWFs were calculated by multiple linear regression of normalized position judgments onto the individual click ITD and ILD values, and indicated large ICI-dependent weights on the initial click ("onset dominance"), elevated weights near offset, and lower weights for interior clicks. Flatter TWFs with reduced onset/offset weights were observed for "fresh" NBT stimuli than for GCT or "frozen" NBT stimuli. The results corroborate previous reports of temporal asymmetries in the binaural processing of periodic stimuli across frequency. [Work supported by R01 DC011548.]

4aPPb29. Frequency domain binaural model with front-back discrimination capability using artificial neural network. Tsuyoshi Usagawa, Takuro Tomita, and Yoshifumi Chisaki (Dept. of Comput. Sci. and Elec. Eng., Kumamoto Univ., 2-39-1 Kurokami, Chuo-Dist., Kumamoto 860-8555, Japan, tuie@cs.kumamoto-u.ac.jp)

Conventional binaural model or binaural hearing assistance systems have a well known ambiguity in front-back discrimination, which is called as "front-back confusion" or "cone of confusion" in psychoacoustics. It is known that spectral cue of sound provides keys to solve this confusion in binaural listening condition and the peaks and notches of spectral components play main role to estimate the vertical angle in sagittal coordinate. In this paper, a frequency domain binaural model with front-back discrimination method using artificial neural model is proposed and examined for various

HRTF catalogs. The performance of the model is examined for simulated conditions using various HRTF catalogs and the results show similar discrimination capability even if the learning process should be done for each catalogs. The experimental results using a dummy head is also provided.

4aPPb30. Simulation of the head-related transfer functions using cloud computing. Tomi Huttunen, Kimmo Tuppurainen, Antti Vanne (Kuava Ltd., Microkatu 1, Kuopio FI-70211, Finland, tomi.huttunen@kuava.fi), Pasi Ylä-Oijala, Seppo Järvenpää (Dept. of Radio Sci. and Eng., Aalto Univ., Helsinki, Finland), Asta Kärkkäinen, and Leo Kärkkäinen (Nokia Res. Ctr., Helsinki, Finland)

Due to the complexity of measurements for obtaining individual head-related transfer functions (HRTFs), numerical simulations offer an attractive alternative for generating large HRTF data bases. In this study, HRTFs are simulated using a fast multipole boundary element method (BEM). The BEM is well suited for the HRTF simulations. Namely, only the surface of the model geometry is discretized which simplifies the pre-processing compared to other full-wave simulation methods (such as finite element and finite difference methods). The BEM is formulated in frequency domain and the model is solved separately for each frequency. Since a large number of frequencies is needed in wide-band HRTF simulations, the BEM simulation greatly benefits from distributed (or parallel) computing. That is, a single computing unit takes care of a single frequency. In this study, a distributed BEM using cloud computing is introduced. Simulations are computed in a public cloud (Amazon EC2) using a realistic head and torso geometry (3D laser scanned geometry of Bruel & Kjaer HATS 4128 mannequin). The frequency range of the simulations is from 20 to 20000 Hz. The feasibility of cloud computing for simulating HRTFs is examined and first analysis results for the simulated HRTFs are shown.

4aPPb31. Effect of distant-variant/invariant head-related transfer functions on perception of a proximal sound source in virtual auditory space. Makoto Otani, Fuminari Hirata, Kazunori Itoh, Masami Hashimoto, and Mizue Kayama (Shinshu Univ., 4-17-1 Wakasato, Nagano 380-8553, Japan, otani@cs.shinshu-u.ac.jp)

A virtual auditory space can be presented to a listener based on binaural synthesis using head-related transfer functions (HRTFs) that are obtainable by measurements or numerical simulations. Due to hardware complexity, HRTF measurements are typically made for a fixed source distance though they are used in binaural synthesis for variable source distances. However, it is known that HRTFs depend on source distance especially for proximal sources for distance less than 1 m. So it is possible that binaural synthesis with HRTFs for a fixed source distance may result in degradations for proximal sound image perception. In this paper, experiments were performed to examine how the use of distant-variant or -invariant HRTFs affect the perception of a proximal sound source in a virtual auditory space in which the listener's motion is compensated by head tracking. HRTFs for source distances up to 1 m, in 5 cm steps, are numerically simulated using the boundary element method. Results show the difference between presented and perceived source distances being significantly smaller when distance-variant HRTFs were used. This indicates that the use of HRTFs corresponding to actual sound source position leads to accurate perception of a proximal source.

4aPPb32. The role of spatial detail in sound-source localization: Impact on head-related transfer function modeling and personalization. Griffin D. Romigh (711th Human Performance Wing, Air Force Res. Labs., 4064 chalfonte, beavercreek, OH 45440, griffin.romigh@wpafb.af.mil), Douglas S. Brungart (Audiol. and Speech Ctr., Walter Reed National Military Med. Ctr., Bethesda, MD), Richard M. Stern (Elec. and Comput. Eng., Carnegie Mellon Univ., Pittsburgh, PA), and Brian D. Simpson (711th Human Performance Wing, Air Force Res. Labs., Dayton, OH)

Current techniques designed to personalize generic head-related transfer functions (HRTFs) have some capacity to quickly customize spatial auditory displays, but these techniques generally fall short of the level of realism and performance provided by fully individualized HRTF measurements. This residual performance deficit reflects inaccuracies due to vast amounts of spatial and spectral variation that occurs across the measured HRTFs of individual listeners. Some of this variation encodes perceptually-important directional information, but a substantial proportion does not. Kulkarni and Colburn (1998) showed that perceptually irrelevant spectral variation could be eliminated by smoothing the HRTF magnitude with a truncated Fourier-series expansion. The present study investigates a related method for smoothing the spatial variation contained in the HRTF by utilizing a truncated spherical harmonic expansion. The impact of spatial smoothing was evaluated by comparing localization performance with individualized HRTFs which were fully represented or had various degrees of spatial smoothing. Results indicate that a highly-smoothed fourth-order spherical harmonic representation can produce localization accuracy comparable to that of a full individualized HRTF. Analysis of the resulting simplified HRTF representations also uncovered a number of interesting relationships across different individuals which may provide new insights for the development of future HRTF personalization and estimation techniques.

4aPPb33. The relation between the information delivered by head-related transfer function and human spatial hearing. Vladimir Tourbabin and Boaz Rafaely (Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva, 84105, Israel, tourbabv@ee.bgu.ac.il)

The human auditory system is capable of performing various tasks related to spatial hearing including sound localization and source segregation. The performance of these tasks depends on many factors, including the complexity of the sound field; the way in which the information from the sound field is transferred to the ears; and the ability of the binaural hearing system to extract the required information. This study focuses on the role of the transfer system represented by the HRTFs that relate the sound field to the signals at the ears. Previously, a measure for the information delivered by the HRTFs was proposed, and the role of HRTFs in human sound localization in the horizontal and median planes was investigated. In the current study, the role of HRTFs in human sound localization is further investigated by analyzing localization in the entire three-dimensional space. Then, the proposed measure is used to investigate the role of HRTFs in source segregation as part of a spatial-release from masking task. The results show that the information delivered by the HRTFs can account for the increase in intelligibility as a function of the spatial separation between the desired source and the masker reported in the literature. However, the decrease in intelligibility with increasing number of spatially separated maskers seems to be related to the binaural hearing system, and cannot be explained by the information in the HRTFs.

4aPPb34. Simplification of head-related impulse response in early reflection simulation. Liang Zhang (Acoust. Lab., Phys. Dept., School of Sci., South China Univ. of Technol., Guangzhou, China) and Xiao-li Zhong (Acoust. Lab., Phys. Dept., & State Key Lab. of Subtropical Bldg. Sci., South China Univ. of Technol., Bldg. No.18, Wu Shan Rd. No. 381, Guangzhou, Guangdong 510641, China, xlzhong@scut.edu.cn)

In virtual auditory environment, early reflections are usually simulated by the image-source method, and binaural signals are synthesized by convolving the input stimulus with corresponding head-related impulse responses (HRIRs). Considering the limited resolution of the human hearing, this work investigates the minimal length of HRIRs needed in early reflection simulation via a simple model consisting of a single direct sound and a reflection. The direct sound is synthesized using 512-point HRIR and fixed at the position directly in front of the subject, while the reflection is synthesized by HRIRs at various directions with four time-domain lengths (512, 256, 128, and 64 points, at a sampling frequency of 44.1 kHz) as well as five time delays relative to the direct sound (from 10 to 50 ms at intervals of 10 ms). A three-interval, two-alternative forced-choice paradigm is employed in this work. Results indicate that for most spatial directions the HRIR with a length of 64-point is perceptually adequate in early reflection simulation. [Work supported by State Key Laboratory of Subtropical Building Science, South China University of Technology, Grant No. 2013KB23.]

4aPPb35. An efficient finite-impulse-response filter model of head-related impulse response. Junfeng Li, Jian Zhang (Inst. of Acoust., Chinese Acad. of Sci., No. 21, Beisihuan Xilu, Beijing 100190, China, junfeng.li.1979@gmail.com), Shuichi Sakamoto, Yoiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Yonghong Yan (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Head-related impulse responses (HRIRs) play a crucial role in binaural 3-D audio rendering. The HRIRs with a couple of hundred-sample lengths result in the high computation cost for the real-time 3-D audio applications especially when multiple sound sources are rendered simultaneously. To overcome this problem, various modeling approaches have been reported to reduce the number of parameters of HRIRs without sacrificing the quality of rendered sounds. In this research, an efficient finite-impulse-response (FIR) model is first studied, which is essentially based on the concept of the minimum-phase modeling technique. In this method, the measured HRIRs are represented by the interaural time delay (ITD) and the magnitude spectra that are approximated by two FIR filters. To investigate the accuracy dependence of this modeling approach on the order of FIR filter, two psychoacoustic experiments on sound localization and sound quality were conducted by comparing the synthesized stimuli with the measured HRIRs and those with the FIR models of different orders. Experimental results indicated that the measured hundred-sample-length HRIRs can be sufficiently modeled by the low-order (a dozen of coefficients) FIR model from the perceptual point of view. The derived low-order FIR model can be further applied to real-time 3-D audio applications.

4aPPb36. Estimation of spectral notch frequencies of the individual head-related transfer function from anthropometry of listener's pinna. Yohji Ishii and Kazuhiro Iida (Faculty of Eng., Chiba Inst. of Technol., 2-17-1 Tsudanuma, Narashino 275-0016, Japan, s0972004QT@it-chiba.ac.jp)

Listener's own head-related transfer functions (HRTFs) are necessary for accurate sound image reproduction. The HRTFs of other listeners often cause the front-back confusion and the errors in elevation perception. It is, however, impractical to measure the HRTFs of any listener for any sound source direction because the measurement requires special apparatus and much time. On the other hand, the estimation of the entire spectrum information of listener's own HRTF still remains as an unsolved difficult issue. One of the authors has shown that the simplified HRTFs, which is recomposed only of the first spectral peak around 4 kHz (P1) and the lowest two spectral notches (N1 and N2) above P1, extracted from the listener's measured HRTFs in the median plane, provide almost the same localization accuracy as the measured HRTFs. While the frequency of P1 is almost constant independent of the sound source elevation and the listener, those of N1 and N2 are highly dependent on both the elevation and the listener. The present study proposes a method, which estimates the frequencies of N1 and N2 in the median plane for the individual listener from the anthropometry of the listener's pinna, and examines the validity of the method.

4aPPb37. Further evidences of the contribution of the ear canal to directional hearing: Design of a compensation filter. Andrea Martelloni (Inst. of Sound and Vib. Res., Univ. of Southampton, Milan, Italy), Davide A. Mauro (Institut TELECOM, TELECOM Paris Tech, CNRS-LTCI, Via Comelico 39/41, Milan 20135, Italy, mauro@di.unimi.it), and Antonio Mancuso (Lab. di Informatica Musicale (LIM), Università degli Studi di Milano, Milan, Italy)

It has been proven, and it is well documented in literature, that the directional response in HRTFs comes largely from the effect of the pinnae. However, few studies have analyzed the contribution given by the remaining part of the external ear, particularly the ear canal. This work investigates the directionally dependent response of the modeled ear canal of a dummy head, assuming that the behavior of the external ear is sufficiently linear to be approximated by an LTI system. In order to extract the ear canal's transfer function, two critical microphone placements (at the eardrum and at the beginning of the cavum conchae) have been used. The system has been evaluated in several positions, along the azimuth plane and at different degrees of elevation. The results point out a non-negligible directional dependence that is well within the normal hearing range; based on these findings, physical models of the ear canal have been analyzed and evaluated. We have also considered the practical application to binaural listening, and the coloration originated by the superimposition of the contribution of two ear canals (the listener's and the dummy head's). A compensating FIR filter with arbitrary frequency response is discussed as a possible fix.

THURSDAY MORNING, 6 JUNE 2013 — 512CG, 9:00 A.M. TO 12:00 NOON

Session 4aSA

Structural Acoustics and Vibration: Applications in Structural Acoustics and Vibration II

Sabih I. Hayek, Cochair
Eng. Sci., Penn State, 953 McCormick Ave., State College, PA 16801-6530

Robert M. Koch, Cochair
Chief Technol. Office, Naval Undersea Warfare Ctr., Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Contributed Papers

9:00

4aSA1. Analysis of the acoustic scattering from a submerged bilaminar plate. Sabih I. Hayek (Eng. Sci. and Mech., Penn State Univ., 953 McCormick Ave., State College, PA 16801-6530, sihesm@engr.psu.edu) and Jeffrey E. Boisvert (NAVSEA, Div. Newport, Newport, RI)

The acoustic scattering from a submerged finite bilaminar rectangular elastic plate is modeled using the exact theory of three-dimensional elasticity. The two layers of the composite bilaminar plate have the same lateral dimensions, but have different thicknesses and material properties. The plate is set in an infinite rigid baffle and is coupled to a different acoustic medium on its two surfaces. The plate is insonified by an acoustic plane wave. The farfield backscattered and forward scattered waves are computed for various layer thicknesses and elastic material properties in contact with air or water. First, the scattering from a uniform finite, baffled, steel plate elastic plate was computed. A bilaminar plate was also analyzed with one layer made of steel and the other made of a damped elastomer. The backscattered and forward scattered acoustic farfield spectra versus frequency were computed at the on-axis point receiver due to a normally incident plane wave. The directivity functions for a normally incident plane wave insonifying the steel- or elastomer-side of the plate were computed for a range of frequencies, with either water or air backing. [Work supported by NAVSEA Division Newport under the ASEE Summer Faculty Program.]

9:20

4aSA2. Numerical modeling of the radiation by a submerged fluid-filled cylindrical shell: Observation of the S0, A0, and A waves. Serguei Iakovlev (Dept. of Eng. Mathematics and Internetworking, Dalhousie Univ., 1340 Barrington St., Halifax, NS B3J 1Y9, Canada, serguei.iakovlev@dal.ca), Hugo A. F. A. Santos (Dept. of Civil Eng. and Architecture, Tech. Univ. of Lisbon, Lisbon, Portugal), Benjamin Schulman, and Kyle Williston (Dept. of Eng. Mathematics and Internetworking, Dalhousie Univ., Halifax, NS, Canada)

We consider a submerged fluid-filled cylindrical shell subjected to an external acoustic pulse and analyze the structure of the field radiated by the shell into the fluids, both external and internal. We first propose a computationally efficient semi-analytical model of the interaction based on the Reissner-Mindlin shell theory combining some of the classical methods of mathematical physics with the finite-difference methodology, and then use the model to simulate the interaction. We demonstrate that the model accurately reproduces the wave structure of the radiated fields seen in the experiments for submerged evacuated shells, namely, both the symmetric Lamb waves S0 and the pseudo-Rayleigh waves A0. It is further observed that the internal and external wave patterns associated with the A0 waves exhibit the same alternation of the equiphase lines as the one seen in the experiments for a plate loaded by the fluid on both sides, a result that seems to be particularly relevant in the context of very limited number of experimental images of the radiated field for shells loaded by fluid from both inside and outside. Not less interestingly, we also demonstrate that the Scholte-Stoneley, or A, wave is also reproduced by the model, and we offer some insights into the non-observability of this wave for certain types of cylindrical shells reported in earlier experimental studies.

9:40

4aSA3. Modeling of wave propagation in drill strings using acoustic transfer matrix method. Je-Heon Han, Yong-Joe Kim (Mech. Eng., Texas A&M Univ., 3123 TAMU, College Station, TX 77840, jeep2000@tamu.edu), and Mansour Karkoub (Mech. Eng., Texas A&M Univ. at Qatar, Doha, Qatar)

In order to understand critical vibrations of a drill bit such as stick-slip and bit-bounce and their wave propagation characteristics through a drillstring system, it is critical to model the torsional, longitudinal, and flexural waves. The objective is to model these waves propagating through the drillstring in a computationally efficient way. Here, a modeling method based on an acoustic transfer matrix between two sets of wave variables at the ends of a cylindrical pipe is proposed. For a drillstring system with multiple pipe sections, the total acoustic transfer matrix is calculated by multiplying all individual matrices of which each is obtained for an individual pipe section. Since drillstring systems are typically extremely long, conventional numerical analysis methods such as FEM require a large number of meshes, which makes it difficult to analyze these drillstring systems. On the contrary, the "analytical" acoustic transfer matrix method requires significantly low computational costs. For the validation, experimental and numerical data are obtained from a laboratory measurement and by using a commercial FEM package, ANSYS, respectively. They are compared to the modeling results obtained by using the proposed method. It is shown that the modeling results are well matched with the experimental and numerical results.

10:00

4aSA4. Some research of mapped radiation modes and its application in analyzing the radiation surface of vibrating structures. Haijun Wu, Weikang Jiang, and Siwei Pan (State Key Lab. of Mech. System and Vib., Shanghai Jiao Tong Univ., 800 Dong Chuan Rd., Shanghai 200240, China, haijun.wu.cn@gmail.com)

Acoustic radiation modes divide the velocity patterns into groups with effective and ineffective radiation efficiencies. It suggests a promising way in the computation of sound power and near field acoustic holography. However, obtaining the radiation modes based on eigenvalue analysis suffers from issues of CPU time and memory, especially for large-scale problems. Based on the idea of equivalent source method and boundary integral equation, it is mathematically and numerically proved that the spherical basis functions is a set of linearly independent patterns on an arbitrary surface. They are termed as mapped radiation modes. The analytical sound power of a structure vibrating in its mapped radiation modes is obtained. An efficient and accurate method to compute the sound power based on the mapped radiation modes is proposed. Based on the relationship between the mapped radiation modes of a vibrating structure and its field sound pressure distribution patterns on a sphere, an improved near field acoustic holography is also developed. It does not need the inverse process but requires microphones to locate at the integral points on a sphere. Numerical examples are presented to validate advantages of applying mapped radiation modes in the sound power evaluation and near field acoustic holography.

10:20

4aSA5. Experimental investigation into sound and vibration of a torpedo-shaped structure under axial force excitation. James Leader, Jie Pan (School of Mech. and Chemical Eng., Univ. of Western Australia, 3 Brahea Place, Mt Claremont, Perth, WA 6010, Australia, 20351548@student.uwa.edu.au), Paul Dylejko (Defence Sci. and Technol. Organisation, Perth, Western Australia, Australia), and David Mathews (HMAS Stirling, Defence Sci. and Technol. Organisation, Perth, WA, Australia)

In this study, the sound radiation patterns and vibration characteristics of a torpedo-shaped structure are determined experimentally using a proof mass actuator to allow pure axial excitation of the model. Using this method, the second energy path found in previous designed structures is eliminated. Input power and driving forces are measured using four force transducers and four accelerometers, while the vibration response and mode shapes are measured using an array of accelerometers. The sound pressure and its directivity are captured by a spatially distributed microphone array inside an anechoic chamber. Motivations for this work are to investigate the effect of the complex boundary constraints: a semispherical head and conical tail on the two meter long model when compared to existing analytical solutions for simple geometries, and later the measurement will be performed in an underwater experiment to contrast the effect of fluid loading.

10:40

4aSA6. Results of an implementation of the dual surface method to treat the non-uniqueness in solving acoustic exterior problems using the boundary element method. Ralf Burgschweiger (PG Computational Acoust., Beuth Hochschule für Technik Berlin, Luxemburger Str. 10, Berlin 13353, Germany, burgi@beuth-hochschule.de), Ingo Schäfer (Underwater Acoust. and Marine Geophys. Res. Inst. (FWG), Wehrtechnische Dienststelle für Schiffe und Marinewaffen (WTD71), Kiel, Germany), Adel Mohsen (Eng. Mathematics & Phys. Dept., Eng. Faculty, Cairo Univ., Cairo, Egypt), Rafael Piscoya, Martin Ochmann (PG Computational Acoust., Beuth Hochschule für Technik Berlin, Berlin, Germany), and Bodo Nolte (Underwater Acoust. and Marine Geophys. Res. Inst. (FWG), Wehrtechnische Dienststelle für Schiffe und Marinewaffen (WTD71), Kiel, Germany)

The problem of non-uniqueness (NU) of the solution of exterior acoustic problems when using the boundary element method (BEM) is well known. Methods like the Burton-Miller technique or the CHIEF method are used to solve this challenge at the expense of more complex procedures for handling hypersingular integrals and/or higher computing times due to higher complexity of the algorithm or additional equations. The dual surface method, commonly used for electromagnetic problems, was adapted for acoustic radiation and scattering problems. The basic principles of methods to solve the NU problem are outlined and results for different models and solution procedures are presented, taking into account quality, solution time, and the numerical advantages when using iterative solvers.

11:00

4aSA7. The peculiarities of the non-axisymmetric frequency spectra of finite elastic cylinders. Dmytriy Libov (Theor. and Appl. Mech., Kiev National Taras Shevchenko Univ., 64 Volodymyrska St., Kiev 04214, Ukraine, dmytro.libov@univ.net.ua)

A rigorous solution of three-dimensional boundary-value problem concerning the forced vibrations of a finite, elastic, isotropic cylinder is constructed analytically by means of the superposition method. With this solution, the resonances in non-propagating waves were investigated.

Particularly, a survey of the frequency spectrum for an aluminum cylinder, vibrating with the circumferential order two, reveals the existence of a localized resonance, usually referred to as an end resonance, well below the cut-off frequency of the lowest real dispersion branch of an infinite cylinder. This phenomenon demonstrates the remarkable differences between the axisymmetric and non-axisymmetric end resonances of elastic cylinders. Comparison of the theoretical results with the experiments published elsewhere reveals an excellent agreement.

11:20

4aSA8. Sound generated by a wing with a flap interacting with an eddy. Avshalom Manela (Aerosp. Eng., Technion, Technion City, Haifa 32000, Israel, avshalom@aerodyne.technion.ac.il) and Lixi Huang (Mech. Eng., Univ. of Hong Kong, Hong Kong, Hong Kong)

Acoustic signature of a rigid wing, equipped with a movable downstream flap and interacting with a line vortex, is studied in a two-dimensional low-Mach number flow. The flap is attached to the airfoil via a torsion spring, and the coupled fluid-structure interaction problem is analyzed using thin-airfoil methodology and application of the Brown and Michael equation. It is found that incident vortex passage above the airfoil excites flap motion at the system natural frequency, amplified above all other frequencies contained in the forcing vortex. Far-field radiation is analyzed using Powell-Howe analogy, yielding the leading order dipole-type signature of the system. It is shown that direct flap motion has a negligible effect on total sound radiation. The characteristic acoustic signature of the system is dominated by vortex sound, consisting of relatively strong leading and trailing edge interactions of the airfoil with the incident vortex, together with late-time wake sound resulting from induced flap motion. In comparison with the counterpart rigid (non-flapped) configuration, it is found that the flap may act as sound amplifier or absorber, depending on the value of flap-fluid natural frequency. The study complements existing analyses examining sound radiation in static- and detached-flap configurations.

11:40

4aSA9. Localization and identification of three-dimensional sound source with beamforming based acoustic tomography. Hao Ding, Huancai Lu, Chunxiao Li, Jiangming Jing, Dongting Mei, and Guozhong Chai (Zhe Jiang Univ. of Technol., Chaowang Rd. 18th, HangZhou 310014, China, haodinggo@gmail.com)

Beamforming based commercial planar microphone array could only localize and identify the sound source when the distance between source and array is known. This paper presents a beamforming based acoustic tomography (BBAT) method to locate and identify the source in 3D space, say, the BBAT method can not only locate the source in X-Y plane that is parallel to the array, but also the depth Z of the source. In this method, the sound field is reconstructed on the virtual planes at different distances along depth direction (Z direction). The maximum response of sound field on every virtual reconstruction plane is tracked, where the largest value among those maximum responses appears at Z direction is the depth of the source. The location of source at X and Y directions can then be easily identified based on beamforming principle, which is utilized by commercial planar array. The BBAT method is evaluated theoretically by simulation of monopole source, the experimental evaluation is done as well in anechoic chamber. The results from both simulation and experiment indicate that this method is capable to locate and identify sound source in 3D space. However, it cannot recognize the sound source located in front of the planar array or behind because of the genetic limitation of 2D planar array in identification of source depth.

THURSDAY MORNING, 6 JUNE 2013 — 515ABC, 9:00 A.M. TO 11:40 A.M.

Session 4aSCa

Speech Communication: Auditory Feedback in Speech Production I

Anders Lofqvist, Cochair
Haskins Labs., 300 George St., New Haven, CT 06511

Charles R. Larson, Cochair
Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208

Invited Papers

9:00

4aSCa1. Individual differences in auditory-motor integration revealed by speech fluency manipulations. Torrey M. Loucks (Speech and Hearing Sci., Univ. of Illinois, 901 S. Sixth St., Champaign, IL 61820, tloucks@illinois.edu), Heecheong Chon (Div. of Speech-Language Pathol., Chosun Univ., Gwangju, South Korea), Shelly Kraft (Commun. Sci. and Disord., Wayne State, Detroit, MI), and Nicoline Ambrose (Speech and Hearing Sci., Univ. of Illinois, Champaign, IL)

A role for auditory feedback in maintaining fluency appears less specific than for pitch control, as one example, but delayed auditory feedback (DAF) clearly provides a potent manipulation of fluency. As most speakers are susceptible to DAF, we predicted DAF is particularly suited to identifying individual differences in auditory-motor integration. We conducted a series of studies to probe susceptibility to DAF-induced disfluency in 60 normally fluent speakers during conversation and oral reading. We further contrasted DAF effects on fluency with dual-task effects on fluency. During conversation and reading under DAF (250 ms delay), multivariate cluster classification indicated speakers show high, low, or intermediate susceptibility to disfluency. In contrast, dual-task effects on fluency appeared bimodal with individuals showing high or low susceptibility. DAF susceptibility was not related to dual-task disfluency in 41/60 speakers, but the remaining speakers were disfluent under DAF and dual-task conditions. When the DAF paradigm was extended to adults who stutter, most were classified as highly susceptible. The findings provide compelling evidence that individual differences need to be considered in auditory-motor integration research. Fluency is influenced by both auditory feedback and cognitive factors related to attention, which can inform theories of normal and disordered speech.

9:20

4aSCa2. Experience-dependent learning effects in speech production with spectrally degraded feedback. Elizabeth D. Casserly (Linguistics, Indiana Univ., Memorial Hall Rm. 322, Bloomington, IN 47406, casserly@indiana.edu) and David B. Pisoni (Psychol. & Brain Sci., Indiana Univ., Bloomington, IN)

This study examined the speech of normal-hearing adult participants before and during their use of a portable, real-time vocoder (PRTV). The PRTV continuously transforms environmental acoustics, including speakers' own speech feedback, via a real-time simulation of cochlear implant processing. The impacts of this substantial spectral degradation on speech production were measured in three groups of subjects: group 1 received altered acoustic feedback for one continuous 55 min session; group 2 experienced the feedback transformation for one session of 6 h total; and group 3 wore the PRTV for four consecutive sessions of 4 h each, for a total of 16 h of experience. Speakers in each group were recorded producing 114 isolated English words and 24 sentences both before their feedback manipulation began and at periodic intervals during their experimental session(s). Acoustic-phonetic analyses of the speech produced by subjects in all three groups revealed substantial effects of the spectral feedback degradation in several domains, including fluency/speaking rate, vocal affect, and vowel quality. Speakers were able to adjust and recover quickly in some of these areas, such as affect, while other changes, such as those in vowel quality and speaking rate, remained despite 16 h of experience with the acoustic transformation.

9:40

4aSCa3. Speech production in amplitude-modulated noise. Ewen N. MacDonald (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ctr. for Hearing and Speech Sci., Bldg. 352, Ørsteds Plads, Kgs Lyngby DK-2800, Denmark, emcd@elektro.dtu.dk) and Stefan Raufer (Institut für Hörtechnik und Audiologie, Jade Hochschule Oldenburg, Oldenburg, Germany)

The Lombard effect refers to the phenomenon where talkers automatically increase their level of speech in a noisy environment. While many studies have characterized how the Lombard effect influences different measures of speech production (e.g., F0, spectral tilt, etc.), few have investigated the consequences of temporally fluctuating noise. In the present study, 20 talkers produced speech in a variety of noise conditions, including both steady-state and amplitude-modulated white noise. While listening to noise over headphones, talkers produced randomly generated five word sentences. Similar to previous studies, talkers raised the level of their voice in steady-state noise. While talkers also increased the level of their voice in amplitude-modulated noise, the increase was not as large as that observed in steady-state noise. Importantly, for the 2 and 4 Hz amplitude-modulated noise conditions, talkers altered the timing of their utterances, reducing the energetic overlap with the masker by approximately 2%. However, for the 1 Hz amplitude-modulated condition, talkers increased the overlap by approximately 4%. Overall, the results demonstrate that talkers are sensitive to the temporal aspects of noisy environments and will alter their speech accordingly.

10:00

4aSCa4. Auditory plasticity and sensorimotor learning in speech production. Douglas M. Shiller (École d'orthophonie et d'audiologie, Université de Montréal, P.O. Box 6128, succursale Centre-ville, Montreal, QC H3C 3J7, Canada, douglas.shiller@umontreal.ca), Daniel R. Lametti, and David J. Ostry (Dept. of Psych., McGill Univ., Montreal, QC, Canada)

Numerous studies have shown the speech motor system to be highly flexible and responsive to changes in sensory input, revealing a central role for both auditory and somatosensory feedback in the acquisition and maintenance of speech motor control. Consistent with these studies, models of speech production have highlighted the role of accurate, stable sensory representations that serve, in part, as the goals of speech movements. A separate (and considerable) body of work has demonstrated that auditory-sensory representations of speech sounds are not perfectly stable, but rather exhibit rapid adaptation to changing input conditions in both children and adults. The plasticity of auditory representations has important implications for the control of speech production, both in early speech motor development and in the sensory-based maintenance of speech accuracy that characterizes adult speech motor control. In this talk, I will describe a series of studies that explore the link between sensory and motor plasticity in the speech motor system. The studies combine the paradigm of sensorimotor adaptation (altering auditory feedback during speech production) with measures and manipulations of auditory-perceptual representations of speech sounds. The results reveal not only that auditory speech targets are flexible under conditions of altered auditory feedback, but that changes in sensory representations can have a direct impact on speech motor learning and performance.

10:20–10:40 Break

10:40

4aSCa5. The relationship between vocal pitch feedback error and event-related brain potentials. Jeffery A. Jones, Nichole Scheerer, and Anupreet Tumber (Psych. & Laurier Ctr. for Cognit. Neurosci., Wilfrid Laurier Univ., 75 University Ave. W., Waterloo, ON N2L 3C5, Canada, jjones@wlu.ca)

Understanding the neural processing of auditory feedback during speech is essential to the development of a comprehensive model of speech motor control. Currently, the relationship between the magnitude of errors detected in feedback and the evoked neural responses is unclear. We exposed speakers to sudden changes in vocal pitch that ranged from 0 to 400 cents in magnitude. Vocal responses and auditory event-related potentials (ERPs: P1-N1-P2-N2 components) were measured. Results showed that vocal response magnitudes were relatively consistent when speakers were exposed to small feedback perturbations (<250 cents). Larger perturbations (>300 cents) caused decreased vocal response magnitudes. P1 amplitudes showed a non-specific increase when feedback was perturbed. N1 amplitudes demonstrated more specificity: smaller feedback perturbations evoked one size of response, while larger feedback perturbations elicited a larger response. P2 amplitudes increased with increases in the feedback perturbation magnitude. Moreover, a reliable relationship existed between vocal response magnitude and P2 amplitude: vocal response magnitude and P2 amplitude increased in response to perturbations between 50 and 250 cents, and then decreased in response to larger perturbations. ERPs allow us to hypothesize the stages of processing. Results will be discussed with respect to perceptual and production thresholds and implications for speech motor control.

11:00

4aSCa6. Sensorimotor integration during human self-vocalization: Insights from invasive electrophysiology. Jeremy Greenlee, Roozbeh Behroozmand, Nandakumar Narayanan (Neurosurgery, Univ. of Iowa, 1827 JCP, 200 West Hawkins Dr., Iowa City, IA 52241, jeremy-greenlee@uiowa.edu), Jonathan R. Kingyon (Dept. of Bioengineering, Univ. of Iowa, Ames, IA), Charles Larson (Speech and Commun. Disord., Northwestern Univ., Evanston, IL), Hiroyuki Oya, Hiroto Kawasaki, and Matthew A. Howard (Neurosurgery, Univ. of Iowa, Ames, IA)

Effective human speech requires the neural integration of ongoing vocal production with the auditory and somatosensory feedback signals that are produced. We are using invasive electrophysiology techniques in patient volunteers undergoing neurosurgical treatment in order to gain insights into these mechanisms and underlying neural circuits. By using multi-contact electrode arrays chronically implanted over the perisylvian temporal lobe auditory cortex (e.g., area PLST) and the inferior frontal gyrus (IFG), we can examine local field potentials and frequency-specific responses from cortical areas important for both vocal production and speech sound processing. Our initial studies have found that during self-vocalization, focal areas within higher order auditory cortex on the superior temporal gyrus (STG) show response modulation compared to the responses of the same areas during passive listening. Manipulation of the auditory feedback that a speaker receives during vocalization (e.g., pitch-shifted or delayed auditory feedback) leads to further modulation of these PLST sites. Measures of functional connectivity including electrical stimulation tract tracing or phase-synchrony analysis demonstrate that portions of PLST are functionally connected to regions of the IFG. These findings support forward models for vocal control in which efference copies of premotor cortex activity modulate sub-regions of auditory cortex.

11:20

4aSCa7. Speech motor learning alters auditory and somatosensory event-related potentials. Takayuki Ito, Joshua H. Coppola (Haskins Labs., 300 George St., New Haven, CT 06511, taka@haskins.yale.edu), and David J. Ostry (McGill Univ., Montréal, QC, Canada)

Speech motor learning is dependent upon changes to motor function, but it also results in changes to sensory systems. However, the neural mechanisms of sensory plasticity associated with speech motor learning are little understood. We here examined whether auditory and somatosensory cortical processes are changed in conjunction with speech motor learning. We tested native speakers of American English. Altered auditory feedback (AAF) training was used as a motor learning task. As subjects repeated aloud the speech utterance "head," the produced sound was feedbacked through headphones while the first formant of /ea/ was gradually decreased over 50 repetitions and held at a maximum change for 110 repetitions. In order to evaluate the effects of the resulting adaptation on cortical sensory processes, we recorded auditory and somatosensory event-related potentials (ERPs) using 64-channel electroencephalography before and after AAF training. Auditory ERPs were elicited by using the synthesized vowel sound "e." Somatosensory ERPs were elicited by facial skin deformation. We found changes to auditory and somatosensory ERPs following AAF training in individuals who showed adaptation to altered auditory feedback. The changes in ERPs were correlated with the amount of adaptation. This suggests that speech motor learning alters somatosensory and auditory cortical processing.

THURSDAY MORNING, 6 JUNE 2013 — 516, 9:00 A.M. TO 12:00 NOON

Session 4aSCb

Speech Communication: Voice and F0 Across Tasks (Poster Session)

Marc Garellek, Chair
Dept. of Linguist, UCLA, Los Angeles, CA 90095

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.

4aSCb1. The relative contribution of rhythm, intonation, and lexical information to the perception of prosodic disorder. Paul Olejarczuk and Melissa A. Redford (Linguistics, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, paulo@uoregon.edu)

Acoustic studies suggest that children with autism spectrum disorder (ASD) produce atypical rhythm and intonation [Diehl and Paul (2011)]. Behavioral studies indicate that children with ASD combine prosody with lexical content in atypical ways [Peppé *et al.* (2007)]. The current study assessed the relative contribution of rhythm, intonation, and language context to perception of prosodic disorder. Short excerpts were taken from narratives produced by 18 children with ASD and 18 typically developing controls. Prior study indicated that listeners easily distinguished groups on the basis of these excerpts. Here, the excerpts were resynthesized to control for voice quality and to allow for selective inclusion of F0, duration, intensity, and lexical information. Experiment 1 investigated listeners' ability to distinguish the groups based on delexicalized samples that preserved only rhythm (duration + intensity), only intonation, or a combination of both. Experiment 2 investigated the contribution of lexical information to the judgments, and the interaction of lexical information with intonation. Results indicated that (1) listeners were less able to distinguish between groups in the Intonation Only condition, and (2) intonation had a negligible effect on performance when lexical content was present. We conclude that rhythm cues and lexical information contribute more to perceived disorder than intonation.

4aSCb2. Intonation perception in English: Effects of stimulus amplitude and listeners' language background. Katherine Morrow and Chang Liu (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, katherinemorrow@utexas.edu)

The contour of fundamental frequency (F0) of the final word is the primary acoustic cue for intonation production and perception in English utterances. On the other hand, speakers of Mandarin Chinese may have to use

other acoustic cues such as amplitude and duration rather than F0 contours to differentiate intonation contrasts since F0 contours carry lexical meaning in Mandarin Chinese. The goal of this study was to examine the role of the final word amplitude in intonation perception of English sentences. The final word amplitude was manipulated at three levels relative to the carrier sentence: −6, 0, and +6 dB. F0 contours of the final word were controlled continuously from falling to rising patterns. Listeners' task was to identify the sentence intonation: question or statement. Preliminary results showed the intonation boundary shifted from slightly falling F0 contours to slightly rising F0 contours as the final word amplitude decreased for Chinese listeners, but the boundary did not change for the three amplitudes for English listeners. These results imply that Chinese listeners may use the final word amplitude as a secondary cue to perceive intonation contrasts in English, while English listeners may primarily rely on F0 contours for intonation perception.

4aSCb3. Towards a vocal typology for American English. Tyler McPeek and James Harnsberger (U. Florida, 4131 Turlington Hall, Gainesville, FL, tylermcpeek@ufl.edu)

Prior work in the field of speaker identification has shown that individual voices are not uniformly dissimilar from one another: when misidentified, the errors are not randomly distributed but, in fact, indicate the existence of vocal stereotypes, or groups of voices that share identifiable features. Popular labels for such groups can include "rich," "droning," "gravelly," and many others. In this study, 100 American English male and female voices were separately rated for interspeaker similarity and the ratings used to posit nine vocal types for each gender. Acoustic properties corresponding to speaking rate, pitch variability, and mean pitch were the most predictive in classifying voices into types in discriminant analyses using a total of 23 measures. For female voices, chronological age strongly influenced the resulting taxonomy. Male voice types were not blocked by age, and unlike the female voices, a single type constituted a plurality (26%) of the voices in the database. The implications of this work for the modeling of other indexical properties of speech will be discussed, along with its implications in the applied area of forensic voice identification.

4aSCb4. Perceptual sensitivity to a model of the source spectrum. Marc Garellek (Dept. of Linguist, UCLA, Los Angeles, CA), Robin A. Samlan, Jody E. Kreiman, and Bruce R. Gerratt (Dept. of Head and Neck Surgery, UCLA, 31-24 Rehab. Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu)

A psychoacoustic model of the source spectrum has been proposed in which four spectral slope parameters describe perception of overall voice quality: H1-H2 (the difference in amplitude between the first and second harmonics), H2-H4, H4-2000 Hz (i.e., the harmonic nearest 2000 Hz), and 2000–5000 Hz. The goals of this study are to evaluate perceptual sensitivity in the mid-to-high frequency range of the model and determine how sensitivity to one parameter varies as a function of another. To determine listener sensitivity to slope changes for each parameter, just-noticeable differences were obtained for series of stimuli based on synthetic copies of one male and one female voice. Twenty listeners completed an adaptive up-down paradigm. To provide a baseline of listener sensitivity to each spectral slope parameter, the synthetic voices were manipulated so that spectral slope varied by 0.5 dB increments for each parameter while other parameters remained constant. We then assessed how listener sensitivity to a given harmonic slope parameter changes when the others covary. These results will help assess the validity of the model and determine what sources of cross-voice variability in spectral configuration are perceptible.

4aSCb5. Biomechanical models of damage and healing processes for voice health. Alba Granados, Jonas Brunskog, and Finn Jacobsen (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Blg. 352, room 111, Kongens Lyngby 2800, Denmark, algra@elektro.dtu.dk)

In voice-loading occupations, employees are required to use their voice for continuous and large periods of time, which might lead to voice problems. In this work, anomalous vocal-fold vibrations due to long-time high voice-load are investigated. Laryngeal endoscopic high-speed images within the vocal-fold plane are available. These data are used to improve existing continuum biomechanical models of the vocal-folds by analyzing the injury processes. The project is expected to result in methods that objectively demonstrate the impact of high voice-load on voice. A detailed description of the currently developing work will be presented, including a rigorous analysis of the hypothesized injury processes of the vocal folds.

4aSCb6. Perceptual consequences of changes in epilaryngeal area and shape. Robin A. Samlan and Jody E. Kreiman (Dept. of Head and Neck Surgery, UCLA School of Med., 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu)

Decreasing epilaryngeal area has been shown to increase glottal flow pulse skewing and harmonic amplitudes [Titze, J. Acoust. Soc. Am. **123**, 2733 (2008)]. It is not known, however, whether listeners perceive voice quality changes when epilaryngeal area is altered, or if perceived quality is different if the area change occurs at the ventricular folds or aryepiglottic (AE) folds. In this study, a kinematic vocal tract model was used to create five epilaryngeal cavity shapes resulting from constriction and retraction of the ventricular and AE folds. Four voice sources simulating varying degrees of vocal deviation were filtered through the five shapes for a total of 20 stimuli. Fourteen listeners completed a sort and rate task. Results were analyzed using multidimensional scaling (MDS). Altering the epilaryngeal cavity shape resulted in voice quality differences, and perceptual distances differed by voice source. AE fold constriction was perceived most differently from other shapes for all talkers. Ventricular fold constriction was perceived most similar to AE constriction for 3 of the 4 voice sources. Glottal flow and acoustic differences for each epilaryngeal shape will be described and related to the perceived differences in voice quality.

4aSCb7. A physiologically and perceptually motivated voice source model. Gang Chen (Elec. Eng., Univ. of California, Los Angeles, Department of Electrical Engineering, UCLA, 63-134 Engr IV, Los Angeles, CA 90095-1594, gangchen@ee.ucla.edu), Marc Garellek (Linguist, Univ. of California, Los Angeles, Los Angeles, CA), Jody Kreiman, Bruce R. Gerratt (Head and Neck Surgery, Univ. of California, Los Angeles, Los Angeles, CA), and Abeer Alwan (Elec. Eng., Univ. of California, Los Angeles, Los Angeles, CA)

Many glottal source models have been proposed, but none has been systematically validated perceptually. Our previous work showed that model fitting of the negative peak of the flow derivative is the most important predictor of perceptual similarity to the target voice. In this study, a new voice source model motivated by high-speed laryngeal videoendoscopy is proposed to capture perceptually-important source shape aspects. Six voice source models (the proposed model, two previous models developed at UCLA, as well as the Fujisaki-Ljungkvist, Liljencrants-Fant, and Rosenberg models) were fitted to 40 natural voices obtained by inverse filtering and analysis-by-synthesis (AbS). We generated synthetic copies of the voices using each modeled source pulse, with all other parameters held constant, and then conducted a visual sort-and-rate task in which listeners assessed the extent of perceived match between the original natural voice samples and each copy. Model fitting results showed that the proposed model provides a more accurate fitting to the AbS-derived source than the other models. Perceptual experiments showed that the proposed model provides a close match to the original natural voice. Perceptual studies examining the extent to which each model matches the target tokens will also be reported. [Work supported by NSF grant IIS-1018863 and NIH/NIDCD grant DC01797.]

4aSCb8. Variation of maximal Lyapunov exponent with voice disorders of pilots. Robert Ruiz (LA.R.A, Univ. of Toulouse, Toulouse, France), Philippe Plantin de Hugues (B.E.A, Bureau d'Enquêtes et d'Analyses pour la sécurité de l'aviation civile, Le Bourget, France), and Claude Legros (LA.R.A, Univ. of Toulouse, 5 allées A.Machado, Toulouse 31058 cedex 1, France, legros@univ-tlse2.fr)

The maximal Lyapunov exponent λ is a signature of chaos in the field of nonlinear dynamics. Analysis of vowels uttered by a troubled speaker can show irregularities and instabilities due to nonlinearities of the phonatory system. Therefore λ can be studied for the research of voice acoustic features able to present variations due to psychophysiological disturbances. These ones belong to the aeronautical context. Two pilots' voices have been recorded at stopovers during short-haul rotations on a day. A day of driving was used as an experimental material to study a similar workload. After being woken up in a laboratory sleep inertia experiment, another pilot is recorded. Finally, the Cockpit Voice Recorder of a crashed airplane

provided a real-case corpus for the study. Vowels are extracted in all recordings and their maximal Lyapunov exponent is estimated. Regardless of whether or not the chaotic behavior of the voice, results show a large dispersion, little variations with different directions from the normal state of the speaker to the end of the recordings. On the basis of these experiments, the number of speakers involved, the choice of the calculation parameters, the phonetic material used, λ has a low sensitivity to the aeronautical psychophysiological disturbances.

4aSCb9. Effects of supraglottic compressions on the aerodynamics and acoustics of excised canine larynges. Fariborz Alipour and Eileen Finnegan (Commun. Sci. & Disord., Univ. of Iowa, 250 Hawkins Drive, 334 WJSHC, Iowa City, IA 52242, alipour@iowa.uiowa.edu)

The purpose of this study was to examine the aerodynamic and acoustic effects due to supraglottic compressions, which may be seen in some dysphonic patients. Canine larynges were prepared and mounted and vocal fold oscillations were generated and controlled by the flow of air through the glottis. Glottal adduction was accomplished by rotating the arytenoids with a suture passed behind the vocal folds to simulate lateral cricoarytenoid muscle action. Supraglottic medial and anterior-posterior compressions were accomplished by manual squeezing at the arytenoid level and alternating between the rest and compressed conditions. The raw data, including EGG, subglottal pressure, flowrate, and microphone signals, were recorded on a DAT tape and later digitized and processed with Matlab. A video image of the superior aspect of the larynx was recorded using a stroboscopic light during the whole experiment. Results indicated that the excised larynges oscillated better and easier without the false vocal folds, but generated louder sound with false vocal folds. Medial compression always resulted in increased subglottal pressure, decreased flow rate and most often increased the sound intensity, but decreased EGG closed quotient. Both of these compressions had negative effects on the amplitude of EGG signal, suggesting disruption of vocal fold contact.

4aSCb10. The quantal larynx revisited. Scott Moisik (Linguist, Univ. of Victoria, P.O. Box 3045, Victoria, BC V8W 3P4, Canada, srmoisik@uvic.ca) and Bryan Gick (Linguist, Univ. of British Columbia, Vancouver, BC, Canada)

Quantal effects signify nonlinear relations in the properties of speech sounds, traditionally emphasizing articulatory-acoustic relations [Stevens, J. Phonet. *17*, 3–45 (1989)]; these relations hold important clues to how continuous phonetic parameters map onto discrete phonological categories. Stevens described quantal states in laryngeal speech function, showing how the vocal fold abduction-adduction continuum can be partitioned into breathy, modal, and pressed phonatory quanta. This account, however, relies on a one-dimensional conceptualization of the larynx, which recent developments in laryngeal phonetic theory reveal to be inadequate [Edmondson and Esling, Phonology *23*, 157–191 (2006)]: a more realistic model must include the epilarynx. We reopen the issue of quantal laryngeal speech behavior in the context of recent research demonstrating quantal biomechanical properties in labial articulation [Gick *et al*., Can. Acoust. *39*, 178–179 (2011)]. Our "whole larynx" approach [Moisik and Esling, ICPhS, 1406–1409 (2011)] countenances epilaryngeal influence on laryngeal articulatory and phonatory possibilities through quantal biomechanical and aero-mechanical effects. We demonstrate, through computer simulation, three novel cases of laryngeal quantality: (1) vocal-ventricular interaction in glottal stop, glottalization, and laryngealization; (2) aryepiglotto-epiglottal stricture in pharyngeal consonants; (3) epilaryngeal predisposition for growl-like or harsh phonation.

4aSCb11. Individual control of singing voice based on cepstrum manipulation. Kenji Ikeda, Kota Nakano, Masanori Morise, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is002081@ed.ritsumei.ac.jp)

Desktop music (DTM) software is used to synthesize various instrumental sounds, whereas it was difficult to synthesize the natural singing voice because the singing voice is more complicated than other instrumental sounds. In the past, it had been object that we output more natural singing voice by analyzing the singing voice and extracting the spectrum envelope with high accuracy. Recently, since the algorithm is improved, the singing voice synthesis software is used. In the conventional method, it had been focused changing the original voice personality by controlling some parameters of spectrum envelope. However, the method can synthesize the singing voice by only particular voice personality. The quality of the synthesized singing voice also depends on accurate control of voice personality. In this study, we attempt to control user's impression directly and investigate the control method that is focused on the singer's individuality in the singing voice. In this paper, we propose a voice personality control method based on mapping the timbre of target singer in the cepstrum domain and demonstrate the effectiveness of the proposed method. As a result of subjective experiments, we confirmed that the proposed method can control the voice personality of singers with high quality.

4aSCb12. Modeling vocal fold asymmetries with coupled Van der Pol oscillators. Jorge C. Lucero (Dept. Comput. Sci., Univ. of Brasilia, Campus Universitario Darcy Ribeiro, Brasilia, Distrito Federal 70910-900, Brazil, lucero@unb.br) and Jean Schoentgen (Lab. of Signals, Images and Telecommunication Devices, Université Libre de Bruxelles, Brussels, Belgium)

Models of the glottal sound source are being developed to extend a recent synthesizer of disordered voices [Fraj *et al*., J. Acoust. Soc. Am. **132**, 2603–2615 (2012)]. The synthesizer was based on a nonlinear wave-shaping algorithm, which generates a glottal excitation to a concatenated-tube representation of the trachea and vocal tract. The purpose of the present work is to incorporate a physics-based model of the vibrating vocal folds in order to increase the anatomical fidelity of the synthesizer. Further, the model will permit to characterize left-right fold asymmetries and explore the effect of those asymmetries on the resultant vocal timbre. In this report, the vocal folds are represented as a system of two coupled Van der Pol oscillators with noise terms and a detuning factor between their natural frequencies. Regions of phase locked and unlocked oscillations are determined and illustrated with bifurcation diagrams. Also, the effect of frequency detuning on the resultant frequency jitter is analyzed. The results are discussed in terms of their implications for modeling abnormal vocal fold behavior. [Work supported by CNPq (Brazil) and FNRS (Belgium).]

4aSCb13. The correlation between perceptual saliency and acoustic parameters of dysarthrias. Emily Wang (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 1611 West Harrison, Ste. 530, Chicago, IL 60612, emily_wang@rush.edu) and Leo Verhagen (Neurological Sci., Rush Univ. Medical Ctr., Chicago, IL)

Dysarthria is a group of speech disorders resulting from neurological disturbances in central or peripheral systems. There are six single types of dysarthria and all present with deviations at both segmental and suprasegmental level. However, it is unclear what matters more to the listener: the deficits at the segmental or suprasegmental level. In this study, reading samples were collected from subjects with any of the three types of dysarthria: scanning speech of ataxic dysarthria, spastic dysarthria, and hypokinetic dysarthria. All had slow speaking rate, monopitch, and monoloudness. Acoustic analyses were used to examine changes at both segmental and suprasegmental level. At the segmental level, parameters obtained include word and syllable per minute, vowel F1 and F2, syllable, word, sentence, and pause duration, mean F0 and vF0 at sentence and paragraph level. Peak F0 and vowel duration of stressed and unstressed vowels were also obtained. Perception experiment was conducted. Pitch contours were extracted and tested separately from those unmanipulated stimuli. Listeners made forced choice for rate and speech naturalness for the former and for overall speech intelligibility, speech rate, and speech naturalness for the latter. Effective size was used to determine the contributions of parameters at the segmental and suprasegmental level.

4aSCb14. Vowel synthesis using a vocal tract mapping interface and simulation study of inverse mapping. Kohichi Ogata and Tomohiro Hayakawa (Grad. School of Sci. and Technol., Kumamoto Univ., 2-39-1 Kurokami, Chuo-ku, Kumamoto 860-8555, Japan, ogata@cs.kumamoto-u.ac.jp)

We have developed a vocal tract mapping interface to produce vowel sounds. The interface provides an easy and effective setting of the vocal tract shape with a simple mouse click on its interface window. A vocal tract shape at the mouse position on the interface window is calculated by interpolation based on the prepared vocal tract shapes. In this paper, the features and advantages of the interface are shown through examples of the generated vowel sounds and vocal tract shapes. In addition, the inverse estimation of the vocal tract shape from formant frequencies is studied as one of its applications. In this method, the vocal tract shape is estimated by searching the point inside the region after

determining a possible region that includes the solution. The possible region on the interface window is determined based on the changes in the formant frequencies. The usefulness of the proposed method is shown through simulation results.

4aSCb15. Disruptive effect of unattended noise-vocoded speech on recall of visually presented digits: Interaction between the number of frequency bands and languages. Kazuo Ueda, Yoshitaka Nakajima (Dept. Human Sci., Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, ueda@design.kyushu-u.ac.jp), Kana Doumoto (Unit of Perceptual Psych., Kyushu Univ., Fukuoka, Japan), Wolfgang Ellermeier, and Florian Kattner (Institut für Psychologie, Technische Universität Darmstadt, Darmstadt, Germany)

To assess the effects of degraded irrelevant speech on the serial recall of visually presented digits, noise-vocoded speech was generated in Japanese and German. Effects of the participants' native language were also examined by studying 40 Japanese and 40 German listeners. The number of frequency bands used in vocoding and the language (native or not) the irrelevant sound was derived from affected performance significantly. The participants' native language had a greater disruptive effect than the non-native language, particularly in conditions in which intelligibility was moderate. Speech sounds appear to have been processed automatically although the participant was instructed to neglect them. This must have required some amount of cognitive resources, which could have been used for the recall task otherwise. This automatic interference was stronger when the native language was used, probably because it contained perceptual cues that were more difficult to degrade.

4aSCb16. Evaluation of bone-conducted ultrasonic hearing-aid regarding transmission of phonetic features. Takayuki Kagomiya and Seiji Nakagawa (Health Res. Inst., National Inst. of Adv. Sci. and Technol., 1-8-31, Midorigaoka, Ikeda, Osaka 563-8577, Japan, t-kagomiya@aist.go.jp)

Human listeners can perceive speech signals in a voice modulated ultrasonic carrier from a bone-conduction stimulator, even if the listeners are patients with sensorineural hearing loss. Considering this fact, we have been developing a bone-conducted ultrasonic hearing aid (BCUHA). The purpose of this study is to evaluate the ability of BCUHA to transmit Japanese distinctive features to the recipient. For this purpose, a series of mono-syllable intelligibility experiments was conducted. A series of sequential information transfer analyses (SINFA) were carried out to analyze what kind of articulatory features were well transmitted. Results of the SINFA showed that: in vowel perception, "openness" and "frontness" were well transmitted, while in consonant perception, Japanese "you-on" (palatalized sound) feature was well transmitted; however, transmission of other features like articulatory position or manner was limited. These results indicated that the BCUHA has sufficient frequency resolution to transmit vowel information, while some signals are masked by carrier sound. To improve this problem, further investigation and development is required.

4aSCb17. Perceptual evaluation of the functional and aesthetic degradation of speech by wind noise during recording. Iain R. Jackson, Paul Kendrick, Trevor J. Cox, Bruno M. Fazenda, and Francis F. Li (Acoust. Res. Ctr., The Univ. of Salford, Newton Bldg., Salford M5 4WT, United Kingdom, t.j.cox@salford.ac.uk)

This paper will present results from a systematic investigation into functional and aesthetic audio quality of speech recordings degraded by wind noise. The major source of wind noise tested comes from velocity fluctuations interacting with the transducer, generating pressure fluctuations at the microphone diaphragm. To better understand the effect of this type of noise, a perceptual experiment was designed to assess task performance and perceptions of quality when speech and simulated wind noise are presented together. A wind noise simulator was developed, which produces realistic audio from anemometer data, to allow the noise to be isolated from other ambient sounds, and also enable salient parameters to be controlled. Two key components of wind noise in recordings were evaluated, the average level and its temporal variance or "gustiness." Eight levels of wind noise were factorially combined with three levels of gustiness. Each of these permutations was then presented with one of 24 randomly assigned, grammatically correct, nonsense sentences. Participants were asked to type the sentence they heard, rate the difficulty of the task, and indicate overall quality of the clip. Each sentence contained four keywords—correct identification of which was used for scoring performance.

4aSCb18. Effect of source viewing on the unmasking of speech. Alan Chorubczyk, Shin-Ichi Sato, and Maira Cardozo (Sound Eng., Tres de Febrero Univ., Amenabar 1819, Ciudad Autónoma de Buenos Aires 1428, Argentina, alanchoru@gmail.com)

In this work, it is analyzed if the subject visual awareness of the speaker presence would result in an enhancement of the speech understanding capability. For this purpose, a pair comparison test with different scales of speech intelligibility conditions with and without source viewing was conducted.

4aSCb19. Detection of obstructive sleep apnea by estimation of oral and nasal cavity cross-section areas from acoustic recordings of snore. Hsu-Kang Huang, Yi-Wen Liu (Elec. Eng., National Tsing Hua Univ., 101 Kuang-Fu Rd. Sec 2, Delta Bldg. Rm. 828, Hsinchu 30013, Taiwan, ywliu@ee.nthu.edu.tw), and Rayleigh Ping-Ying Chiang (ENT, Shin Kong Hospital, Taipei, Taiwan)

Obstructive sleep apnea (OSA) refers to the condition in which a person's breathing is paused while asleep, or the airflow is decreased, due to obstruction in the upper respiratory airway. In severe cases, OSA can cause complete arousals and deprive the patient from normal sleep. Surgical intervention is sometimes recommended, but accurate identification of the site of obstruction can be difficult. In the present study, we devised signal processing methods to estimate the site and the severity of airflow obstruction from recordings of sounds of snore. The vocal tract, the oral and the nasal cavity are modeled as three branches joining at the pharynx. Each branch consists of cylindrical segments whose cross-section areas can vary during snoring. Estimation of these cross-section areas consists of two steps: First, an auto-regressive moving-average method is applied to find the linear coefficients of a pole-zero model that optimally accounts for the recorded sound. Then, the Levinson-Durbin algorithm is applied to convert the coefficients to ratios of cross-section areas between adjacent segments. The present method is applied to a set of recorded snore samples during clinically confirmed apnea episodes, and results are compared with those of simple snore. Effectiveness of the method is analyzed statistically.

4aSCb20. Study of unvoiced fricative speech production: Influence of initial conditions on flow development. Yo Fujiso, Annemie Van Hirtum (GIPSA-Lab, Grenoble Univ., 11 rue des Mathématiques, Grenoble Campus, Saint Martin d'Hères 38402, France, yo.fujiso@gipsa-lab.grenoble-inp.fr), Kazunori Nozaki, and Shigeo Wada (Grad. School of Eng. Sci., Osaka Univ., Toyonaka-city, Japan)

Human unvoiced fricative speech sounds such as [s] and [f] are produced by a complex fluid-structure interaction. Indeed, a moderate Reynolds number ($100 \leq Re \leq 10000$) turbulent jet is issued from a constriction somewhere in the vocal tract formed between the hard palate and an articulator such as tongue, teeth, or lips. By using simplified in-vitro replicas representing parts of the human vocal tract, some physical phenomena relevant to the unvoiced fricative speech production can be reproduced and more easily understood. The current study focuses on the influence of initial conditions on flow development by performing flow measurements and Large Eddy Simulation on a rectangular channel containing a tooth-shaped obstacle.

4aSCb21. Measurements of the aero-acoustic properties of the vocal folds and vocal tract by broad and narrow band probes during phonation into controlled acoustic loads. Noel N. Hanna, John Smith, and Joe Wolfe (School of Phys., The Univ. of New South Wales, Sydney, NSW 2052, Australia, n.hanna@unsw.edu.au)

The aeroacoustic properties of the vocal folds and tract are difficult to measure directly. Here, they were measured using broad- and narrow-band excitation at the mouth during phonation into various acoustic loads, including a non-resonant load provided by an acoustically infinite waveguide with cross section comparable with that of the tract. The tract is treated as a duct terminated by the larynx. Mechanical properties of the walls and terminations were determined using a microphone array [Dickens *et al.* (2007)]. The vocal fold response was monitored with an electroglottograph and wall motion was measured electromechanically. The impedance spectra show negative resistance bands at frequencies near those of phonation, consistent with regeneration at the folds. The walls give inertances consistent with thicknesses of order 1 cm and compliances consistent with distributed stiffnesses of about 100 kN/m^3 [Hanna *et al.* (2012)]. The duct resonant properties are consistent

with losses several times higher than the viscothermal losses at smooth rigid walls. Dickens *et al.*, J. Acoust. Soc. Am. **121**, 1471–1481 (2007). Hanna *et al.*, in *Proceedings of the Australian Acoustical Society Conference* (2012).

4aSCb22. Flow development in the uniform glottis and viscosity effects.
Lewis Fulcher (Phys. & Astronomy, Bowling Green State Univ., Bowling Green, OH 43403, fulcher@bgsu.edu) and Ronald Scherer (Commun. Sci. & Disord., Bowling Green State Univ., Bowling Green, OH)

Thirty-two pressure distributions at minimal diameters of d = 0.005, 0.0075, 0.01, 0.02, 0.04, 0.08, and 0.16 cm have been measured at a number of transglottal pressures of interest for phonation. Care is taken to identify those portions of the pressure distributions within the glottis that include substantial regions of uniform decrease with axial distance. These portions are further examined to identify their components that have a linear dependence on the volume velocity and those that have a quadratic dependence on the volume velocity. An analysis based on the Navier Stokes equation creates a natural framework for investigating corrections to the parabolic profile of fully developed flow, which leads to the Poiseuille formula. For glottal diameters between 0.0075 and 0.02 cm the Poiseuille formula is a good approximation. Overall, an inverse 2.59 power law to describe the diameter dependence of the linear coefficients is found to be superior to the inverse cube dependence of the Poiseuille formula. Glottal flow resistance is used as a means of comparing the accuracy of the two power laws.

THURSDAY MORNING, 6 JUNE 2013 — 510A, 9:00 A.M. TO 12:00 NOON

Session 4aSP

Signal Processing in Acoustics: Sensor Array Beamforming and Its Applications

Jens Meyer, Cochair
mh Acoust., 38 Meade Rd., Fairfax, VT 05454

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

Contributed Papers

9:00

4aSP1. Circular harmonics beamforming with spheroidal baffles.
Stewart Holmes (Lloyd's Register ODS, Strandvejen 104A, Hellerup 2900, Denmark, stewart.holmes@lr-ods.com)

Circular microphone arrays have been well studied when mounted on cylindrical and spherical baffles. In this paper, the oblate and prolate spheroids are explored as a baffle for a microphone array system. It is shown that the prevailing methods for analysis of circular arrays may be easily adapted to this class of baffle, and that the use of these baffles represent a continuously variable geometry with edge cases that are the cylindrical and spherical baffles, and the array in empty space with no baffle. The performance—in the form of directivity index, maximum side-lobe level and main beam resolution—of spheroidally baffled arrays is analyzed with respect to errors created by transducer noise, positioning error, and modal aliasing, for both delay-and-sum and eigenbeamforming arrays. It is shown that the noise and position errors depend strongly on the spheroidal eccentricity while the aliasing error is fairly independent of baffle shape. Simulations of example arrays are used to show that while the prolate spheroidal baffle is of little advantage compared to current systems, the oblate spheroidal baffle can be used to create a significantly smaller array with only a relatively minor performance degradation.

9:20

4aSP2. Spatial sound pick-up with a low number of microphones.
Julian D. Palacino and Rozenn Nicol (FT/OLNC/OLPS/COMSERV/SVQ//TPS, Orange Labs, 2 Av Pierre Marzin, Lannion 22307, France, julian.palacino@orange.com)

Portable audio devices have become more and more popular during the last decade. Recent advances in audio compression and electronic miniaturization allow people to keep all their music and movies at hand, at all times. Nowadays people are using their smartphones everywhere as photo or video cameras. In order to provide a consumer possibility of 3D audio recording adapted to these kinds of devices, we developed a compact microphone array able to pick-up a full 3D sound scene, using less than four microphones. To compensate for the low number of microphones, which results in poor spatial selectivity, spatial post processing is applied to the microphone signals and improves the sound source localization. Another advantage is that the post processing makes the sound reproduction flexible. The 3D audio scene can be converted into any format to be rendered to any equipment and any device. The paper will describe various recording setups in combination with the associated post processing. As a first assessment, their performances in terms of sound localization accuracy will be compared using a new set of objective criteria descriptors.

9:40

4aSP3. Lattice theory models for space-time sampling of acoustic signals.
Kaushallya Adhikari and John R. Buck (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, kadhikari@umassd.edu)

The space-time sampling of underwater acoustic signals by both fixed and towed arrays of sensors can be modeled by lattice theory. The sampling schedule of a fixed array produces a rectangular space-time lattice while the sampling schedule of an array towed at a uniform velocity produces a trapezoidal lattice. Changing the velocity of a towed array corresponds to changing the skewness of a trapezoidal lattice. Willis and Bresler [IEEE Info. Theory (1997)] established an upper bound on the space-time-bandwidth product of a signal that can be sampled time sequentially by a lattice without aliasing. This upper bound provides a valuable perspective on the tradeoff among the temporal intersample interval, the interelement spacing of sensors, the velocity of the towed array, and the spectral support of signal that can be sampled without aliasing. A towed array with the same interelement spacing and temporal intersample interval can sample signals with broader spectral support than a fixed array. Alternately, a towed array can sample a signal with the same spectral support at a slower rate than an equivalent fixed array decreasing load on the data processing system. [Work supported by ONR.]

10:00

4aSP4. Frequency-sum beamforming in an inhomogeneous environment.
Shima H. Abadi, Matthew J. VanOverloop, and David R. Dowling (Mech. Eng., Univ. of Michigan, 2010 W.E.Lay Automotive Lab. 1231 Beal Ave., Ann Arbor, MI 48109, shimah@umich.edu)

The arrival directions of ray paths between a sound source and a receiving array can be determined by beamforming the array-recorded signals. And, when the array and the signal are well matched, directional resolution increases

with increasing signal frequency. However, when the environment between the source and the receivers is inhomogeneous, the recorded signal may be distorted and beamforming results may be increasingly degraded with increasing signal frequency. However, this sensitivity to inhomogeneities may be altered through use of an unconventional beamforming technique that manufactures higher frequency information by summing frequencies from lower-frequency signal components via a quadratic (or higher) product of complex signal amplitudes. This presentation will describe frequency-sum beamforming, and then illustrate it with simulation results and near-field acoustic experiments made with and without a thin plastic barrier between the source and the receiving array. The experiments were conducted in a 1.0-meter-deep and 1.07-m-diameter cylindrical water tank using a single sound projector, a receiving array of 16 hydrophones, and 100 micro-second signal pulses having nominal center frequencies from 30 to 120 kHz. The results from frequency-sum beamforming will be compared to the output of conventional delay-and-sum beamforming for different center frequencies. [Work sponsored by ONR and NAVSEA.]

10:20

4aSP5. Heading and hydrophone data fusion for towed array shape estimation. Jonathan Odom and Jeffrey Krolik (Elec. and Comput. Eng., Duke Univ., P.O. Box 90291, Durham, NC 27708, jonathan.odom@duke.edu)

This paper addresses the problem of towed array shape estimation for passive, horizontal sonar arrays. Beamforming and localization techniques significantly degrade when an assumed linear array bends due to tow platform maneuvers or ocean currents. In this paper, heading sensors along the array and acoustic hydrophone data are jointly used to estimate the shape of the array. Previously, heading data have been filtered using a dynamical motion model to reduce noise during turns. In recent work, a time-varying noise field directionality estimate that incorporates a dynamical model for the acoustic field provides a second, albeit biased, estimate of the array shape. In this paper, these two estimates are combined via adaptive weights to obtain improved shape estimates during maneuvers. A multi-source simulation is used to demonstrate the robustness of the combined array shape estimate when compared to the separate heading or acoustic sensor based techniques.

10:40

4aSP6. Least squares versus non-linear cost functions for a virtual artificial head. Eugen Rasumow, Matthias Blau (Institut für Hörtechnik und Audiologie, Jade Hochschule WOE, Ofener Str. 16/19, Oldenburg D-26121, Germany, eugen.rasumow@jade-hs.de), Simon Doclo (Institut für Physik, Carl von Ossietzky Universität, Oldenburg, Germany), Martin Hansen (Institut für Hörtechnik und Audiologie, Jade Hochschule WOE, Oldenburg, Germany), Steven van de Par (Institut für Physik, Carl von Ossietzky Universität, Oldenburg, Germany), Dirk Püschel (Akustik Technologie Göttingen, Soundtec, Oldenburg, Germany), and Volker Mellert (Institut für Physik, Carl von Ossietzky Universität, Oldenburg, Germany)

In order to take into account spatial information into binaural recordings, it is common practice to use so-called artificial heads. Disadvantageously artificial heads are inherently non-individual and bulky devices. Alternatively, the individual frequency-dependent directivity pattern of human head related transfer functions (HRTFs) can also be approximated by a microphone array with appropriate filters [Rasumow *et al*. (2011)]. Such a setup may be referred to as a virtual artificial head (vah). The filters for the application of the vah can be derived by minimizing a narrow band cost function including regularization constraints. As a first approach, it is appropriate to apply a least-squares cost function. The major advantage is its closed form solution [cf. Rasumow *et al*. (2011)], whereas from a psychoacoustically point of view, it seems more reasonable to minimize the dB-error instead. The latter cost function must, however, be minimized iteratively. We propose a minimization procedure for and present first results regarding the subjective appraisal of binaural filters derived using both cost functions. Future work includes the extension of this work to binaural cost functions.

11:00

4aSP7. Signal processing for hemispherical measurement data. Markus Müller-Trapet and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany, mmt@akustik.rwth-aachen.de)

To realistically model the sound propagation in rooms, a detailed knowledge of the reflection properties of the surrounding surfaces is required. In this context, the reflection properties include both the sound absorption as well as scattering. In order to be able to measure the angle-dependent reflection properties of surfaces in-situ, a hemispherical microphone array was recently designed and built. For a reduction of the required hardware an efficient, rotationally symmetric sampling was chosen, so that 28 microphones on two concentric semicircles are employed to measure a total of over 3000 positions on a hemisphere in just over 15 minutes. This contribution will give an overview over the required signal processing steps to process the measurement data from such a microphone array. Special emphasis will be placed on the determination of the microphone positions and the special case of data available on a hemispherical surface. Also, the sound field model used to determine the impedance on the surface will be explained. Preliminary results will be presented.

11:20

4aSP8. Fly-over aircraft noise measurement campaign at Montreal-Trudeau airport using a microphone array. Jean-Francois Blais (Acoust. and Vib., Bombardier Aerosp., P.O. Box 6087, Station Ctr.-Ville, Montreal, QC H3C 3G9, Canada, jean-francois.blais@aero.bombardier.com), Cédric Camier (GAUS, Univ. of Sherbrooke, Sherbrooke, QC, Canada), Mathieu Patenaude-Dufour, Robby Lapointe (Acoust. and Vib., Bombardier Aerosp., Montreal, QC, Canada), Jonathan Provencher, Thomas Padois, Philippe-Aubert Gauthier, and Alain Berry (GAUS, Univ. of Sherbrooke, Sherbrooke, QC, Canada)

Engines being quieter due to high by-pass ratios, the airframe noise, produced for instance by landing gears or high-lift devices, has become a significant contributor to the total noise radiated by aircraft during approach and landing. As part of the investigations carried out to understand noise generation mechanisms, the beamforming techniques developed over the last decade and applied to microphone array measurements have shown to be effective tools for localization and quantification of these aerodynamic noise sources. In order to validate their in-house beamforming softwares, Bombardier Aerospace and the Groupe d'Acoustique de l'Université de Sherbrooke have conducted a 5-day measurement campaign in June 2012. The 95-microphone array was located on the roof of a building next to the Montreal-Trudeau airport. Aircraft position was determined by two high-definition cameras, both synchronized with the microphone array by inter-range instrumentation group time codes generators. This paper summarizes the measurement campaign. The aircraft tracking tool and the beamforming algorithms used to characterize the noise sources are presented. Several Bombardier CRJ fly-overs were recorded during this test. Beamforming results obtained for different airlines are compared in order to evaluate the repeatability of the method.

11:40

4aSP9. Accuracy of head-related transfer functions synthesized with spherical microphone arrays. Cesar Salvador, Shuichi Sakamoto, Jorge Trevino (Grad. School of Information Sci./Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai 980-8579, Japan, salvador@ais.riec.tohoku.ac.jp), Junfeng Li, Yonghong Yan (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and Yôiti Suzuki (Grad. School of Information Sci./Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan)

The spherical harmonic decomposition can be applied to present realistically localized sound sources over headphones. The acoustic field, measured by a spherical microphone array, is first decomposed into a weighted sum of spherical harmonics evaluated at the microphone positions. The resulting decomposition is used to generate a set of virtual sources at various angles. The virtual sources are thus binaurally presented by applying the corresponding head-related transfer functions (HRTF). Reproduction accuracy is heavily dependent on the spatial distribution of microphones and virtual sources. Nearly uniform sphere samplings are used in positioning the microphones so as to improve spatial accuracy. However, no previous studies have looked into the optimal arrangement for the virtual sources. We evaluate the effects of the virtual source distribution on the accuracy of the synthesized HRTF. Furthermore, our study considers the impact of spatial aliasing for a 252-channel spherical microphone array. The microphone's body is modeled as a human-head-sized rigid sphere. We evaluate the synthesis error by comparison with the target HRTF using the logarithmic spectral distance. Our study finds that 362 virtual sources, distributed on an icosahedral grid, can synthesize the HRTF in the horizontal plane up to 9 kHz with a log-spectral distance below 5 dB.

Session 4aUWa

Underwater Acoustics: Detection and Localization

Yong Min Jiang, Cochair
NATO STO CMRE, Viale San Bartolomeo 400, La Spezia 19126, Italy

Julien Bonnel, Cochair
ENSTA Bretagne, 2 rue François Verny, Brest cedex 9 29806, France

Contributed Papers

9:00

4aUWa1. Using warping processing to range bowhead whale sounds from a single receiver. Julien Bonnel (LabSTICC (UMR CNRS 6285), ENSTA Bretagne, 2 rue François Verny, Brest cedex 9 29806, France, julien.bonnel@ensta-bretagne.fr) and Aaron Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA)

In certain shallow water environments, the acoustic propagation of low-frequency marine mammal calls can be well-modeled as a discrete set of normal modes. Each mode propagates with a different group velocity, and thus in principle the range of the call can be inferred by comparing relative arrival times of the modal arrivals. Traditionally, several time-synchronized hydrophones are required to spatially filter out individual modes in order to measure relative arrival times. In this presentation a nonlinear signal processing method classed "warping" is used to identify individual mode arrival times on a single receiver, even when the mode arrivals are overlapping in time. Warping processing is limited to frequency-modulated sources with monotonic increases or decreases of frequency with time. It is thus applicable to whale calls that consist of simple frequency-modulated upsweeps or downsweeps. Once the modes are separated, the source range can be estimated using conventional modal dispersion techniques. This method is applied on several bowhead whale vocalizations recorded near Kaktovik (Alaska) in 2012. Bowhead whale calls are ranged up to 35 km under median ambient noise conditions. These single-receiver range estimates are consistent with estimated ranges previously obtained via other methods [Work supported by North Pacific Research Board and Shell Exploration and Production Company.]

9:20

4aUWa2. Source signature characterization and feature based detection of open-circuit SCUBA regulators. Kay L. Gemba and Eva-Marie Nosal (Dept. of Ocean and Resources Eng., Univ. of Hawai'i at Manoa, 2500 Campus Rd., Honolulu, HI 96822, gemba@hawaii.edu)

A goal of the Center for Island, Maritime, and Extreme Environment Security is to develop passive acoustic methods to monitor harbor and near-shore environments. To study detection of open circuit SCUBA divers, several regulator configurations were recorded and characterized under ideal conditions in a pool environment. Sound Pressure Levels (SPL) and Sound Spectral Levels were calculated over the 4 kHz to 80 kHz band. Results include SPL range over all recordings along with variation of SPL due to change in SCUBA tank pressure and regulator flow rate. Regulator broadband signatures were used to test three diver detection algorithms under varying synthetic and real ambient noise conditions. The first detector is a broad band energy detector exploiting *a priori* knowledge of the underlying signal length. The other two detectors are an envelope and a cepstral detector which estimate the divers' fundamental breathing frequency and require at least two breaths for a positive result. In order to maximize SNR, regulator signatures were band pass filtered to exploit their respective dominant features. Receiver operating characteristic curves were calculated to compare detector performances. [Work funded by the U.S. Department of Homeland Security through the Center for Island, Maritime, and Extreme Environment Security.]

9:40

4aUWa3. Data-based sensitivity kernel in a highly reverberating cavity. Selda Yildiz, Christian Marandet, Sandrine T. Rakotonarivo (Marine Physical Lab., Scripps Inst. of Oceanogr./UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, syildiz@ucsd.edu), Philippe Roux (Institut des Sci. de la Terre, Université Joseph Fourier, Grenoble, France), and W. A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr./UCSD, La Jolla, CA)

Our goal is to acoustically localize medium inhomogeneities, (i.e., scatterers) in a very complex medium without having to resort to constructing an accurate propagation model. Instead, we use a data-based sensitivity kernel approach to characterize medium changes in this complex medium which, in this study, is a highly reverberating cavity. The efficacy of the method is confirmed in an experiment with a moving aggregate of ping-pong balls inside a fish tank of 5.6 m diameter and water depth of 1.05 m in the ~10 kHz frequency regime; acoustic sources and receivers are on the periphery of the tank. Using a sensitivity kernel constructed from field data for scatterers at a sparse set of known positions, we demonstrate that we can localize other scatterers at unknown positions.

10:00

4aUWa4. Active detection of a moving target in a waveguide with strong masking echoes. Yoann Benoit and Claire Prada (Institut Langevin, ESPCI ParisTech, CNRS UMR 7587, 1 rue Jussieu, Paris 75006, France, claire.prada-julia@espci.fr)

In shallow water, active detection of a small moving target can be difficult because of strong echoes from large fixed obstacles. To cancel strong unwanted echoes, differences between successive acquisitions can be achieved, however they are very sensitive to fluctuations. A projection method combined with a fast acquisition technique is proposed as a robust alternative. An ultrasonic experiment is presented: a 64 transducers linear vertical array is used to detect a small target moving above a large obstacle in a waveguide. To reduce acquisition time, eight groups of adjacent elements transmit linear frequency modulations with increasing delays in a single emission. The 8x64 array response matrix is then obtained by correlations and time windowing. The projection is achieved between two acquisitions obtained while the target is moving, in order to remove the obstacle's contribution. Namely, the second acquired matrix is projected on the space orthogonal to the eight singular vectors of the first acquired matrix. Then, it is shown that the first singular vector of the projected matrix focuses on the second target's position. Comparisons are made with the decomposition of the time reversal operator in differential mode and conventional beamforming.

10:20

4aUWa5. Underwater source localization using a hydrophone-equipped glider. Yong Min Jiang and John Osler (Ctr. for Maritime Res. and Experimentation, Viale San Bartolomeo 400, la Spezia, (SP) 19126, Italy, jiang@cmre.nato.int)

Buoyancy-driven underwater gliders are autonomous underwater vehicles that were originally developed to collect oceanographic data. CMRE is studying the use of this technology for the characterization of denied areas, including alternate sensor payloads and applications. During the Rapid

Environmental Assessment phase of the Noble Mariner 2012 NATO exercise, Conducted in Gulf of Lions in September 2012, an omnidirectional hydrophone was mounted on a shallow water glider to sample the spatial distribution of the acoustic and oceanographic fields at different ranges and depths. This paper presents a study of the potential to localize acoustic sources by using the acoustic and environmental data collected by the glider. During the experiment, a bottom moored acoustic source was deployed in an area with benign bathymetry. Continuous wave and frequency modulated pulses were broadcast for approximately 6 h. The glider was flying along predefined tracks and the distances from the source were typically from 5 to 9 km. A Ray tracing model is used to evaluate the arrival structures of the acoustic signal, and to estimate the source location. The impact of the range dependent water column sound speed profile on the uncertainty of the source localization is also discussed.

10:40

4aUWa6. Three-dimensional localization of multiple sources in an uncertain ocean environment. Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no) and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper develops a Bayesian focalization approach to simultaneous three-dimensional localization of multiple sources in shallow water with uncertain environmental properties, for application to horizontal line array (HLA) data. The algorithm maximizes the posterior probability density over unknown environmental (seabed and water column) and source parameters (includes source-bearing) using an adaptive hybrid optimization algorithm. Maximum-likelihood expressions for source strengths and noise variance are used that allows these parameters to be sampled implicitly rather than explicitly. An extension to the algorithm that optimizes for *a priori* unknown number of sources, based on minimizing the Bayes information criterion, is developed and presented. The algorithm is applied to simulated multi-frequency data in a continental shelf environment, and to data recorded on a HLA deployed on the seafloor in an experiment conducted in the Barents Sea.

11:00

4aUWa7. Small boat localization using adaptive three-dimensional beamforming on a tetrahedral and vertical line array. John Gebbie, Martin Siderius (Elec. and Comput. Eng. Dept., Portland State Univ., 1900 SW 4th Ave., Ste.160, Portland, OR 97201, jgebbie@ece.pdx.edu), Peter Nielsen, James H. Miller (Res. Dept., STO-CMRE, La Spezia, Italy), Steven Crocker (Sensors & SONAR Systems Dept., NUWC, Newport, RI), and Jennifer Giard (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Passive acoustic detection and localization of small surface craft has a number of practical applications, such as monitoring and protecting sensitive marine habitats. Moored passive equipment can be cumbersome to deploy and communicate with, so AUV-mounted devices are being investigated as an alternative. The GLASS'12 experiment was designed to assess the feasibility of using a hybrid autonomous underwater vehicle outfitted with a compact volumetric nose array as a data collection platform. The array consisted of five vertical elements and 4 in a tetrahedral arrangement, and the hybrid underwater vehicle had the capability operating in either glider or propeller-driven modes. The rigid design of the array minimized element location mismatch and enabled the use of aggressive adaptive beamforming in 3-D. This facilitated isolation of broadband multipath arrivals originating from the motor of a small rubber boat. Cross-correlation of beams enabled the time-lag between the arrivals to be measured, which, in turn yielded information about the target range. The underlying formulation bears similarity to the passive fathometer [J. Acoust. Soc. Am. **120**(3) (2006)], which exploits surface wave noise rather than ship noise. This presentation will focus on the array beamforming and potential applications for localization and environmental sensing.

11:20

4aUWa8. Separation of moving ship striation patterns using physics-based filtering. Yann Le Gall and Julien Bonnel (ENSTA Bretagne, Lab-STICC (UMR CNRS 6285), 2, rue François Verny, Brest 29806, France, yann.le_gall@ensta-bretagne.fr)

When a ship is moving toward an acoustic receiver in an oceanic waveguide, the time-frequency representation of the recorded signal exhibits a striation pattern that can be useful in numerous applications such as ship localization or geoacoustic inversion. If there are many ships, the striation patterns add up and they must be separated if one wants to study them separately. In this paper, a physics-based filtering scheme for passive underwater acoustics has been developed. The algorithm allows separating the time-frequency striations of two different moving ships. The proposed method considers filtering the 2D Fourier transform of the received spectrogram. The filter design is based on the waveguide invariant principle and on some *a priori* knowledge on the oceanic waveguide. The noise nature on the spectrogram is taken into account by introducing a nonlinearity to the filtering scheme. The algorithm thus corresponds to a nonlinear homomorphic filter. The method is validated on both simulated data and experimental marine data. This filtering scheme offers good prospects for all applications using ship noise and a single receiver.

Session 4aUWb

Underwater Acoustics and Acoustical Oceanography: Propagation and Scattering

Kyle M. Becker, Chair
Ocean Acoust. Program, Office of Naval Res., 875 N. Randolph St., Arlington, VA 22203-1995

Contributed Papers

9:00

4aUWb1. Experimental verification of enhanced sound transmission from water to air at low frequencies. David C. Calvo, Michael Nicholas, and Gregory J. Orris (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, david.calvo@nrl.navy.mil)

Laboratory-tank measurement of enhanced sound transmission from water to air at low frequencies is presented. Findings are consistent with the theory of anomalous transparency of the water-air interface in which almost all of the acoustic energy emitted by a shallow submerged source is emitted into the air [Godin, Phys. Rev. Lett. **97** (2006)]. The classical picture in the water remains very much the same: the monopole source suffers a radiation efficiency decrease due to interference with the strong surface reflection [McDonald and Calvo, J. Acoust. Soc. Am. **122** (2007)]. For source depths progressively less than a fraction of an acoustic wavelength in water, the measured radiation pattern in the air becomes progressively omnidirectional. The wider radiation pattern owes itself to the conversion of inhomogeneous (evanescent waves) into propagating waves that fill the angular space outside the usual 13-degree cone. On-axis point measurements using a microphone in air and a hydrophone in water, along with the measured directivities, are consistent with previously published power transmission ratios from water to air. Asymptotic expressions for the radiated field in the air are also presented. [Work sponsored by the Office of Naval Research.]

9:20

4aUWb2. Measured scattering of a first-order vortex beam by a sphere: Cross-helicity and helicity-neutral near-forward scattering and helicity modulation. Viktor Bollen, David J. Zartman, Timothy M. Marston, and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, viktor.bollen@wsu.edu)

The wavefield of a traveling wave acoustic vortex beam has an axial null and an angular phase ramp. An appropriately phased four-element transducer array can be used to generate a first order vortex beam [Hefner and Marston, J. Acoust. Soc. Am. **106**, 3313–3316 (1999)]. The direction of the phase ramp determines the helicity of the beam. Superposition of signals from an appropriately positioned four-element receiver array gives a helicity selective detector and commutation of diagonal source elements can be used to reverse the source helicity [Marston and Marston, J. Acoust. Soc. Am. **127**, 1856 (2010)]. These techniques were used to investigate the near forward scattering by a small sphere placed on or near a beam's axis. The forward scattering vanishes in the on-axis case [Marston, J. Acoust. Soc. Am. **124**, 2905–2910 (2008)]. As the sphere is moved off axis the scattering to a helicity neutral receiver is found to increase linearly in the displacement with a first order phase swirl as a function of the sphere coordinates. For cross-helicity detection (detection opposite the beam's helicity) as required by symmetry, the signal is approximately quadratic in the displacement with a second-order phase swirl. [Work supported by ONR.]

9:40

4aUWb3. A numerically stable rational approximant for the split-step Padé propagator. David E. Roberts (34 Craiglockhart Loan, Edinburgh, Scotland, EH14 1JS) and David J. Thomson (733 Lomax Rd., Victoria, BC V9C 4A4, Canada, drdjt@shaw.ca)

The split-step Padé algorithm due to Collins [J. Acoust. Soc. Am. **93**, 1736–1742 (1993)] provides a fast and accurate method for solving the parabolic equation (PE). The formal solution to the PE propagator, which involves the pseudo-differential operator $(1+X)^{1/2}$, is replaced by an $[n/n]$-Padé rational approximant. This approximant can be expanded either as a product or a sum of rational-linear terms, each term leading to a tridiagonal system of equations in X which is readily solved numerically. To ensure adequate suppression of undesirable contributions from the evanescent part of the spectrum ($X < -1$), stability constraints must be imposed. In contrast to this approach, we follow the suggestion of Lu and Ho [Optics Lett. **27**, 683–685 (2002)] and examine the use of an $[n-1/n]$-Padé approximant that inherently dampens these evanescent components of the spectrum. An algorithm for generating the necessary coefficients is described. Transmission losses computed using both rational approximants are compared for a typical shallow-water configuration.

10:00

4aUWb4. Three-dimensional acoustic propagation under a rough sea surface. Megan S. Ballard (Appl. Res. Labs. at the Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78665, suedewitt@gmail.com)

A three-dimensional propagation model using stepwise coupled modes is applied to calculate the acoustic field under a rough sea surface. The model is formulated in a cylindrical coordinate system and the solution for the three-dimensional acoustic field is approximated by accounting for mode coupling in the radial direction and including horizontal refraction in the azimuthal direction. The atmosphere above the sea surface is modeled as an acoustic half space having the properties of air and sea surface height is allowed to vary arbitrarily as a function of range and azimuth. For the sea surfaces presented in this work, the amplitude spectrum of the surface waves is modeled according to the JONSWAP spectrum and the directionality is included by assuming cosine-squared spreading. The acoustic field is calculated for sea surfaces determined for varying levels of wind intensity and fetch. A modal decomposition of the acoustic field is used to provide insight into the effects of the rough sea surface on the predicted transmission loss. The importance of using three-dimensional versus two-dimensional models for acoustic propagation under rough sea surfaces is investigated. [Work supported by ONR.]

10:20

4aUWb5. Acoustic interface treatment with an adjoint operator for linear range-dependent ocean index of refraction inversions. Edward Richards and Gopu Potty (Univ. of Rhode Island, Narragansett Bay Campus, South Ferry Rd., Narragansett, RI 02882, edwardlrichards@gmail.com)

Hurskey *et al.* [J. Acoust. Soc. Am. **115**(2), 607–619 (2004)] introduced the adjoint method and incorporated local sound-speed measurements into range-dependent ocean sound-speed inversion. They left two practical issues for the implementation of this method unresolved in this paper: the collection of appropriate environmental measurements and the implementation of bottom boundary

conditions. The first of these issues was considered in a simulation study that used an oceanographic glider to collect range-dependent sound-speed measurements [Richards, CMRE memorandum (2012)]. A covariance matrix was constructed from the changes observed in the range-dependent sound-speed field. The adjoint inversion was performed in a reduced element subset of the empirical orthogonal functions (EOF) base of the covariance matrix. The second issue is the focus of this paper, which describes a bottom boundary condition with a defined adjoint operator. A horizontal fluid-bottom interface is implemented using the implicit finite difference form of the parabolic equation introduced by McDaniel and Lee [J. Acoust. Soc. Am. **71**(4), 855–858 (1982)]. Combined with local range-dependent sound-speed statistics gathered with gliders, this development may provide a method of near real-time acoustic measurement of ocean sound-speed variations between an acoustic source and vertical hydrophone array.

10:40

4aUWb6. Applicability of two-dimensional boundary scattering models as a proxy for three-dimensional models. Bryant M. Tran (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bmtran@gmail.com), Sumedh M. Joshi (Ctr. for Appl. Mathematics, Cornell Univ., Ithaca, NY), and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Three-dimensional numerical models offer unique insight into the nature of scattering from rough surfaces. However, use of these models is computationally prohibitive for any application more time-sensitive than basic research. This work seeks to determine a proxy for full three-dimensional rough boundary scatting models using appropriate two-dimensional models. Specifically, a Monte Carlo Kirchhoff approximation model in 2D with a derived proxy relationship applied is compared to a similar model in 3D. The region of validity of the proxy will be explored. The usage of the proxy function when applied to an finite element method model will also be discussed. [Work supported by ONR Ocean Acoustics.]

11:00

4aUWb7. Comparison of two-dimensional axial-symmetric and two-in-one-half dimensional Green's function methods for three-dimensional shallow water acoustic propagation using finite element methods. Benjamin M. Goldsberry and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas at Austin, 7201 Wood Hollow Dr., Apt.# 436, Austin, TX 78731, bmg08c@my.fsu.edu)

In shallow water acoustic propagation, performing a fully three-dimensional finite element model is currently unfeasible due to difficulty in implementation and limits in computational power. Therefore, alternative representations of the 3D acoustic field are sought. Two promising methods to represent the 3D field are a locally invariant, 2.5D Green's Function kernel, and a 2D axial-symmetric reduction of the 3D Helmholtz equation. When a spherical source is used, azimuthal symmetry of the acoustic propagation is assumed, and these two methods can be compared in 2D planes of the 3D field. Because the 2.5D method takes more computational time than the axial-symmetric method, the accuracy of the pressure field are compared to see if the axial-symmetric method can be used in place of the 2.5D method. First, the two methods are compared for a flat ocean surface with stratified media. Then, a wedge-shaped ocean surface is considered, and the two methods are compared with 2D PE solutions. These comparisons will show if the axial-symmetric method produces similar results to the 2.5D method, and if so, under which geometrical and physical situations the axial-symmetric method can be used in place of the 2.5D method. [Work sponsored by the Office of Naval Research, Ocean Acoustics.]

11:20

4aUWb8. The rate of convergence and error distribution of Galerkin approximations to eigenvalues in underwater acoustics. Richard B. Evans (Mathematical Sci., RPI, 99F Hugo Rd., N. Stonington, CT 06359, rbevans@99main.com)

The eigenfunctions of the depth separated wave equation can be expanded in terms of a known finite basis set. The expansion coefficients are found by requiring that the error is orthogonal to each of the M members of the basis. This is the Galerkin approximation, wherein one obtains an MxM matrix eigenvalue problem that can be solved by existing software packages. The convergence of the matrix eigenvalues depends on the suitability of the chosen basis set. Typically, the errors in the matrix eigenvalues are bounded by $1/(M^r)$, for large M, where the exponent $r > 0$ is the rate of convergence: Consider basis sets consisting of trigonometric (Fourier) or orthogonal (Legrendre) polynomials. The density discontinuity at the bottom of the ocean creates a corner in the eigenfunctions that should be built into the basis sets. A corner in the sound speed profile (e.g., at the bottom of the mixed-layer) yields $r = 3/2$, which assures convergence, but is still a practical consideration. The distribution of errors is a determining factor in the choice between Fourier-Galerkin and Legendre-Galerkin. The errors in the first (2/pi)M of the eigenvalues are orders of magnitude smaller with Legendre-Galerkin, in the problem presented.