Three-dimensional point-cloud room model in room acoustics simulations

Markovic, Milos; Olesen, Søren Krarup; Hammershøi, Dorte

Published in:
The Journal of the Acoustical Society of America

DOI (link to publication from Publisher):
10.1121/1.4806371

Publication date:
2013

Document Version
Early version, also known as pre-print

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 $2.4 \pm 0.6\ \Omega\cdot\mathrm{cm}^2$ compared to $3.8 \pm 0.5\ \Omega\cdot\mathrm{cm}^2$ 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

4a THU. AM

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; ±0.1 dB tolerance; 1 Hz high-pass enabled), 1/2-in. Preamplifier/Free-Field Microphone (3.15 Hz– 20 kHz: ±2.0 dB; 5Hz–10 kHz: ±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

4a THU. AM

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

4a THU. AM

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.

Session 4pAAa

Architectural Acoustics: Room Acoustics Computer Simulation II

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

1:00

4pAAa1. The differences and though the equivalence in the detection methods of particle, ray, and beam tracing. Uwe M. Stephenson (HafenCity Univ. Hamburg, Hebebrandstr. 1, Hamburg 22297, Germany, post@umstephenson.de)

Within the numerical methods used in room acoustics, the geometrical and energetic methods of sound particle, ray and beam tracing are often confused. This rather tutorial paper does not treat the tracing algorithms but rather aims to explain the differences in the physical models and the corresponding detection and evaluation methods. While ray tracing needs spherical detectors as receivers to count rays, the particle model is based on a weighting of the energies of the particles with their inner crossing distances to compute the local sound energy densities. For beam tracing, receiver points are sufficient. In its core, this paper shows the convergence of the evaluated intensities computed from immitted sound particle energies to those predicted by the well-known 1/R<+>2<+>-distance law for the free field—as applied with the mirror image source method and beam tracing as its efficient implementation. Finally, the geometrical methods are classified depending on their efficiency with higher orders of reflection and their extensibility by scattering and diffraction.

1:20

4pAAa2. Modeling (non-)uniform scattering distributions in geometrical acoustics. Dirk Schröder (LCAV, EPFL, Station 14, Lausanne 1015, Switzerland, dirk.schroeder@epfl.ch) and Alexander Pohl (HCU Hamburg, Hamburg, Germany)

In most cases, a surface is not ideally smooth. It rather contains regular and irregular dents, bumps, and other textures that influence the reflection of the incident wave. A reflection on such a corrugated surface causes a frequency-dependent redirection of the incident sound energy outside the specular direction, called scattering. While the computation of the specular part is well elaborated today, a model that thoroughly captures the wave phenomenon of scattering is still under discussion. Here, the most common assumption is that scattered energy follows a uniform Lambert distribution, which has proven to be a good approximation, especially in room acoustical applications. In this contribution, we will discuss Lambert-based scattering models (specular/diffuse sound field decomposition and vector mixing) and their implementations in methods of Geometrical Acoustics. We will analyze benefits and flaws of the respective models and investigate possibilities to introduce angle-dependent scattering for use cases where the uniform Lambertian distribution becomes invalid.

1:40

4pAAa3. A hybrid acoustic model for room impulse response synthesis. Alexander Southern and Samuel Siltanen (Dept. of Media Technol., Aalto Univ. School of Sci., Otaniemi, Finland, samuel.siltanen@aalto.fi)

The prediction and synthesis of room impulse responses (RIR) has wide application from computer gaming to architectural acoustics. When a level of physical accuracy is important, a single acoustic modeling technique is usually limited by its computational load. Hybrid acoustic models target different time/frequency regions of the RIR with different modeling techniques. This paper introduces a hybrid acoustic model consisting of a physical FDTD model for low-mid frequencies, beam-tracing, and the acoustic radiance transfer method in the early part and late parts at high frequencies respectively. In this work, attention is given to establishing the equivalence of the boundary characteristics in each modeling domain. Good agreement is demonstrated indicating that mixing the separate model responses leads to an energetically consistent RIR.

2:00

4pAAa4. Comparison of sound field measurements and predictions in coupled volumes between numerical methods and scale model measurements. Paul Luizard (LIMSI-CNRS, BP 133, Université Paris Sud, Orsay 91403, France, paul.luizard@limsi.fr), Makoto Otani (Faculty of Eng., Shinshu Univ., Nagano, Japan), Jonathan Botts, Lauri Savioja (Dept. of Media Technol., Aalto Univ. School of Sci., Aalto, Finland), and Brian F. Katz (LIMSI-CNRS, Orsay, France)

Prediction of sound fields in closed spaces can be achieved by various methods, either physical or numerical, based on different theoretical features. While the benefits and limitations of many methods have been examined for single volume spaces, there has been little effort in examining these effects for coupled volume situations. The present study presents a case study comparing theoretical, experimentally physical measurements on a scale model, and various numerical methods, namely boundary element method (BEM), finite-difference time-domain (FDTD), and ray-tracing through the commercial software CATT-ACOUSTIC and ODEON. Although these numerical methods all use 3D numerical models of the architecture, each is different. Ray-tracing is more suitable to geometries with larger planes; BEM requires a more regular finer surface mesh; and FDTD requires a volumetric mesh of the propagation medium. A simple common geometry based on the scale model is used as a basis to compare these different approaches. Application to coupled spaces raises issues linked to later parts in the decay due to multi-slope decay rates, as well as diffraction phenomenon due to acoustic energy traveling between coupling surfaces from one volume to another. The ability of these numerical methods to adequately model these effects is the question under study.

2:20

4pAAa5. Inversion of a room acoustics model for the determination of acoustical surface properties in enclosed spaces. Soenke Pelzer and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr. 50, Aachen 52066, Germany, spe@akustik.rwth-aachen.de)

Acoustic consultants are often in charge of treating spaces to fix problems or improve their room acoustics. To assess the situation and to find a solution, it is common practice to perform computer simulations. This technique is well established, cheap and effective. But it requires a CAD model of the room as well as properties of its boundaries, such as absorption and scattering coefficients. The CAD model is usually easy to obtain by asking the architect or measuring yourself, but quantifying the absorption and scattering coefficients of every single wall is a challenging task. This contribution presents a method that automatically matches absorption coefficients for every single wall by applying an inverse room acoustics model which bases on geometrical acoustics. The inversion is done numerically using a non-linear least-squares optimization process in MATLAB. The independent variables are all absorption coefficients and the goal is to minimize the error between measured and simulated impulse responses at all measured positions in the room. In addition to the acquisition of absorption and scattering coefficients, the goal after the optimization process is to perform interactive binaural auralizations that have a high perceptual congruence with the existing space.

2:40

4pAAa6. Construction and optimization techniques for high order schemes for the two-dimensional wave equation. Stefan Bilbao and Brian Hamilton (Music, Univ. of Edinburgh, Rm. 7306B, JCMB, Kings Bldgs., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, sbilbao@staffmail.ed.ac.uk)

With the advent of high performance parallel computing, audio rate room auralization using finite difference time domain (FDTD) methods is becoming possible in a reasonable computation time. Yet, there are still deficiencies in the methods, which are used for this purpose, particularly with regard to minimizing numerical dispersion over the full range of audible frequencies. This paper is concerned with construction techniques for families of methods for the test case of the 2D wave equation. Such methods are explicit, can be of very high accuracy, and operate over a small local stencil. Such schemes can be attractive in a parallel computation environment. As such methods will depend, invariably, on a set of free parameters, including the Courant number, a major concern is optimization. The remainder of this paper approaches the problem of setting up such an optimization problem in terms of various constraints and a suitable cost function. Some of the constraints follow from consistency, stability, isotropy, and accuracy of the resulting scheme, and others from perceptual considerations peculiar to audio. Simulation results will be presented.

3:00–3:20 Break

Contributed Papers

3:20

4pAAa7. Speech intelligibility prediction in very large sacral venues. Wolfgang Ahnert and Tobias Behrens (Ahnert Feistel Media Group, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu)

In very large sacral venues like cathedrals or mosques the intelligibility of spoken words is very important especially during praying. For such venues with volumes of up to more than one million m3 special routines are needed for simulation to obtain predicted STI values by using of up to more than 1000 sound sources. Special cloud computing has been developed which allow to do the calculation by providing the needed memory size and by cutting the calculation time from days or weeks to hours. Here also modern binaural or ambisonic B-format impulse responses are derived. Additionally the absorption behavior of typical floor materials in such venues has to be known like worshipers in church pews or sitting or kneeing on carpets in mosques. This absorption is often the only one in sacral venues to reduce the reverberation time. For mosque projects the measurement of absorption coefficients of persons in typical postures and arrangements has been done according to the reverberation room method. Persons have been tested on a carpet as a 10 m<+>2<+> sample within a surrounding reflective barrier while standing, kneeing on carpet or being in Muslim specific praying posture.

3:40

4pAAa8. The effect of edge caused diffusion on the reverberation time - A semi analytical approach. Stefan Drechsler and Uwe Stephenson (HafenCity Univ., Hebebrandstr. 1, Hamburg 22297, Germany, drechsler@anklick-bar.de)

The basic numerical model here is the Anisotropic Reverberation Model (ARM). This geometrical/energetic model assumes a homogeneous but anisotropic sound field in room acoustics. Its system of linear differential equations describes the redistribution of sound energy to different directional ranges by wall reflections, which may be specular or diffuse, where the diffuse reflections are caused also by the edges (edge effect). The reverberation times result from eigenvalues and eigenvectors of the differential equation system. Recently, an analytical formula has been found, that calculates the diffracted, angle dependent sound field, averaged over the octave band even for arbitrarily shaped polygons. The reverberation times calculated with the ARM extended by that edge diffraction are presented. So, first time, not only for the

4p THU. PM

typical shoe-box room with an absorbing floor and reflecting walls realistic reverberation times can be calculated, taking the edge effect into account.

4:00

4pAAa9. Numerical models for predicting absorption/insulation performance of acoustic elements. Naohisa Inoue and Tetsuya Sakuma (The Univ. of Tokyo, 5-1-5 Kashiwanoha, Kashiwa 2778563, Japan, naohisa.inoue7@gmail.com)

With a great improvement of computer resource availability, numerical analysis is widely used to investigate acoustic characteristics of various materials. A further expectation will be to predict absorption/insulation performance of acoustic elements used for buildings, automobile and so on, which can be the alternative to the actual measurements. This paper presents general numerical models for predicting the absorption coefficient and the transmission loss of acoustic elements with arbitrary shape and material composition. The features of the models are: (1) A test sample is mounted in the cavity or aperture on a thick rigid baffle: (2) FEM is employed for the materials and the air in the cavity, and coupled with sound fields out of the baffle by BEM: (3) Acoustical indices are calculated from the incidence power and the absorption/transmission power on the interfaces. Numerical simulation demonstrates the oblique incidence absorption coefficients and transmission losses of single- and multi-layered materials, and the influence of the sample's area is discussed in comparison with theoretical values for the infinite area. Additionally, the influence of the thickness of the cavity/aperture is examined, which is so-called "niche-effect."

4:20

4pAAa10. Hexagonal vs. rectilinear grids for explicit finite difference schemes for the two-dimensional wave equation. Brian Hamilton and Stefan Bilbao (Acoust. & Fluid Dynam. Group, Univ. of Edinburgh, Rm. 4350, JCMB, Kings Bldgs., Mayfield Rd., Edinburhg EH9 3JZ, United Kingdom, b.hamilton-2@sms.ed.ac.uk)

Finite difference schemes for the 2-D wave equation operating on hexagonal grids and the accompanying numerical dispersion properties have received little attention in comparison to schemes operating on rectilinear grids. This paper considers the hexagonal tiling of the wavenumber plane in order to show that the hexagonal grid is a more natural choice to emulate the isotropy of the Laplacian operator and the wave equation. Performance of the 7-pt scheme on a hexagonal grid is better than previously reported as long as the correct stability limit and tiling of the wavenumber plane are taken into account. Numerical dispersion is analyzed as a function of temporal frequency to demonstrate directional cutoff frequencies. A comparison to 9-pt compact explicit schemes on rectilinear grids is presented using metrics relevant to acoustical simulation. It is shown that the 7-pt hexagonal scheme has better computational efficiency than parameterized 9-pt compact explicit rectilinear schemes and that the error remains isotropic to fourth-order. Simulation results are presented.

4:40

4pAAa11. Acoustic propagation modeled by the curvilinear Fourier pseudospectral time-domain method. Maarten Hornikx and Daan Steeghs (Eindhoven Univ. of Technol., P.O. Box 513, Eindhoven 5600 MB, Netherlands, m.c.j.hornikx@tue.nl)

The Fourier pseudospectral time-domain method is an efficient domain-discretization wave-based method to model sound propagation in inhomogeneous bounded media. The method was successfully applied to model atmospheric sound propagation and acoustics in urban environments. One of the limitations of the method is its restriction to a Cartesian grid, confining it to staircase-like geometries. When applying a transform from the Cartesian coordinate system to the curvilinear coordinate system, more arbitrary geometries may be solved by the method. In free field, the frequency dependent accuracy of the curvilinear Fourier pseudospectral time-domain method is investigated as a function of the deformation angle of the grid. Further, the performance of the pseudospectral method with a curvilinear grid as well as a Cartesian grid for scattering of elementary objects as an inclined plate and a cylinder is studied. Finally, sound propagation in a room with non-parallel boundaries and over a building with gabled roof is computed with the pseudospectral method with a curvilinear grid and compared with results obtained from the boundary element method. All computed results are in 2D.

5:00

4pAAa12. Three-dimensional point-cloud room model in room acoustics simulations. Milos Markovic, Søren K. Olesen, and Dorte Hammershøi (Section of Acoust., Dept. of Electron. Systems, Aalborg Univ., Fredrik Bajers Vej 7 B4-209, Aalborg Ø 9220, Denmark, mio@es.aau.dk)

Telepresence applications require communication with the feeling of being together and sharing the same environment. One important task in these applications is to render the acoustics of the distant room for the telepresence system user. This paper presents a fast method for the room geometry acquisition and its representation with a 3D point-cloud model, as well as utilization of such a model for the room acoustics simulations. A room is scanned with a commercially available input device (Kinect for Xbox360) in two different ways; the first one involves the device placed in the middle of the room and rotated around the vertical axis while for the second one the device is moved within the room. Benefits of both approaches were analyzed. The device's depth sensor provides a set of points in a three-dimensional coordinate system which represents scanned surfaces of the room interior. These data are used to build a 3D point-cloud model of the room. Several models are created to meet requirements of different room acoustics simulation algorithms: plane fitting and uniform voxel grid for geometric methods and triangulation mesh for the numerical methods. Advantages of the proposed method over the traditional approaches are discussed.

Session 4pAAb

Architectural Acoustics and Noise: Control of Impact and Airborne Noise in Buildings

Jean-Philippe Migneron, Chair
Ecole d'architecture, Universite Laval, 1, cote de la Fabrique, Quebec City, QC G1K 7P4, Canada

Invited Papers

1:00

4pAAb1. Global understanding of important parameters for improvement of impact insulation. Jean-Philippe Migneron and Jean-Gabriel Migneron (Ecole d'architecture, Universite Laval, 1, cote de la Fabrique, Quebec City, QC G1K 7P4, Canada, jean-philippe.migneron.1@ulaval.ca)

Floors impact insulation performances can be very different from one assembly to another. Many years of research and development have been done in this area. Now, it seems that a new solution or another product is commercialized every week. From buyers' point of view, there is a need to decide which topping and underlay will suit some noise requirement to the lowest cost. However, acousticians and specialists in noise control might consider a more complex problem, especially in multi-family dwellings. In lightweight construction, the relation between the floor and the ceiling underneath also affect the overall performance in terms of IIC, or even in risk of complaints. Knowing that it is often difficult to compare a real situation to a datasheet from a manufacturer or to building codes, few key ideas should be remembered. This paper aims to briefly review some conclusions of previous works done in impact sound insulation and to analyze how fundamental parameters can be applied to real installations. An example of variable modification on a topping sample also tries to demonstrate the influence of basic aspects without according most attention to single number ratings.

1:20

4pAAb2. Effects of flooring, topping, and underlayment on impact sound insulation of wood-joisted floor-ceiling assemblies. Lin Hu, Anes Omeranovic (Bldg. Systems, FPInnovations, 319 rue Franquet, Quebec, QC G1P 4R4, Canada, lin.hu@fpinnovations.ca), and Richard Dufour (Res. and Development, Feutre National Felt Inc., St-Narcisse, QC, Canada)

Footstep impact noise transmission through floor-ceiling assemblies is a major source of complaints in wood-framed multi-family buildings. Experience shows that adding a finishing-topping-underlayment sandwich on a base floor-ceiling assembly significantly affects the noise transmission. So far, no reliable tool has been developed for designing the proper sandwich and most proposed solutions rely basically on trial and error. FPInnovations has launched a major research project to develop such tools. First phase of the project is focused on understanding the effects of flooring, topping and underlayment on the impact sound insulation. A mock-up assembly simulating a pair of typical stacked rooms was built. Standard field impact sound transmission tests were conducted on floors topped with a 1.2 m by 1.2 m four-layer sandwich patch. By varying the combination of flooring, topping and underlayment, over 50 patches were tested to better understand the effects of the type of materials in each layer on the assembly's overall impact sound insulation. Based on a large number of tests conducted so far, it is evident that proper combination of flooring, topping and underlayment produces satisfactory impact sound insulation on wood-joisted floor-ceiling assembly. Verification testing on the assembly fully covered by the 4-layer samples is under way.

1:40

4pAAb3. On the relevance of impact source impedance at low frequencies. Berndt Zeitler, Stefan Schoenwald, and Brad Gover (Construction, NRC Canada, 1200 Montreal Rd., Bldg. M-27, Ottawa, ON K1A 0R6, Canada, berndt.zeitler@nrc.ca)

Many researchers have posed the question of whether the standard tapping machine simulates the impedance of real sources well enough to properly judge the impact sound insulation performance of a floor. Proposed solutions such as the modified tapping machine, the bang machine, and ball were results of these investigations. Recent data collected on bare (wood and concrete) floors, suggest that in the low frequency range, the impedance of the source has no influence on the power injected into the floor. This is presumably due to the fact that the bare floors have much higher impedances than most common sources, meaning only the blocked force of the source influences the injected power. This furthermore suggests that modifying or redeveloping the source is not necessary, and that through use of an appropriate weighting curve a single number rating that correlates well with subjective measurements can be defined. Supporting objective and subjective results will be presented.

4p THU. PM

Contributed Papers

2:00

4pAAb4. Impact noise isolation provided by a bare concrete slab evaluated according to European and American regulations and two prediction software compared to field measurements. Nicolas Lévêque (MJM Acoust.al Consultant, 6555 Côte des Neiges, Bureau 440, Montréal, QC H3S 2A6, Canada, nicolas.leveque00@gmail.com)

European regulation EN12354-2:2000 outlines a procedure to predict impact noise isolation provided by a floor covering installed on a concrete slab. In Annex B of this regulation, one can found a formula to calculate the Normalized Impact Sound Pressure Levels (NISPL) for a bare concrete slab. In North America, ASTM 2179-03 standard describes a procedure to measure impact noise insertion loss provided by a floor covering in laboratory conditions, using NISPL of a reference bare concrete slab. INSUL software uses Cremer's point force excitation theory to evaluate NISPL of concrete slab whereas BASTIAN software uses the procedure described in EN12354-2:2000 European regulation. This paper presents a comparison between NISPL calculated for 8 to 10 inches thick bare concrete slab according to procedures and software listed above and field NISPL measurements of thirty-five bare concrete slabs varying in thickness from 8 to 10 inches

presented in a article published in the Journal of Canadian Acoustical Association [Morin (2009)]. This comparison suggests that a statistical approach is required to evaluate accurately NISPL provided by a floor covering installed on a concrete slab according to the procedures and software listed above.

2:20

4pAAb5. Partition intersections and their effect on transmission loss and apparent sound transmission class. Jean-François Latour (Acoust. and Vib., SNC-Lavalin Inc., 2271 Fernand-Lafontaine, Longueuil, QC J5G 2R7, Canada, jean-francois.latour@snclavalin.com)

It is widely recognized and accepted that poorly designed intersections between partitions can significantly reduce the sound isolation that is achieved. However, the effect of specific intersection details are not widely reported and available. This presentation will focus on a case study in which different intersection details have been tested in the same conditions using the same source and receiving rooms. In terms of ASTC and transmission loss, *in situ* results (i.e., ASTM E 336) will be presented and compared with expected performance based upon laboratory test results (i.e., ASTM E 90) for the same partition type. Observed differences between laboratory and field performance will also be compared with results from similar previous studies.

2:40

4pAAb6. A study of a real world transmission loss chamber and the Kinetics UniBrace-L technology. Scott Hulteen, Eric McGowan, and Dominique J. Chéenne (Columbia College Chicago, 4118 N Ashland Ave., Chicago, IL 60613, scott.hulteen@loop.colum.edu)

This study performed at Columbia College Chicago (CCC) had two purposes: The first was to test the functionality of the recently developed real-world transmission loss (RWTL) chamber, while the second was to evaluate the performance of the Kinetics UniBrace-L product on a double wall construction assembly. CCC's RWTL chamber is designed to illustrate issues that have influence when testing partitions in the field. It is smaller in volume than a certified STC chamber, which results in modal effects on both sides of the chamber. Numerous microphone positions are available and are used to display the modal effects of the rooms and to determine an average sound pressure level in both spaces during testing. Absorption values are also substantially different between the sending and the receiving spaces (each side can be switched as either sending or receiving room) and diffuse-field conditions are not achieved in either side of the chamber. As such, the RWTL chamber will typically yield a lower STC value than a certified STC Chamber and will also yield lower values than what may be experienced when performing a test that would follow the standard ASTM E-336 or associated procedures.

THURSDAY AFTERNOON, 6 JUNE 2013 — 510B, 12:55 P.M. TO 4:40 P.M.

Session 4pAB

Animal Bioacoustics: Animal Vocal Modification in Noise

Susan Parks, Chair
Syracuse Univ., 114 Life Sci. Complex, Syracuse, NY 13244

Chair's Introduction—12:55

Invited Papers

1:00

4pAB1. Calling in gray treefrog choruses: Modifications and mysteries. Mark A. Bee (Ecology, Evolution and Behavior, Univ. of Minnesota, St. Paul, MN) and Joshua J. Schwartz (Biology, Pace Univ., Pleasantville, NY, NY 10570, JSCHWARTZ2@PACE.EDU)

Frogs are well known model systems in the study of communication for investigating the influences of noise on both signaling behavior and auditory processing. The best-studied frogs in this regard are two sister-species in the *Hyla versicolor* species complex (*H. versicolor* and *H. chrysoscelis*). Males of both species produce loud, pulsatile advertisement calls that function to attract females. In the competitive social environment of a breeding chorus, males commonly shift to producing longer calls (with more pulses) at slower rates when the level of competition increases. These behavioral modifications can be evoked in controlled laboratory experiments using playbacks of calls and chorus-shaped noise. In contrast to birds and mammals, however, there is no evidence that males increase the amplitude of their vocalizations (the Lombard Effect) in response to increasing noise levels. In addition, current evidence suggests that males do not necessarily profit significantly from producing longer calls at slower rates in terms of increasing their overall attractiveness to females, overcoming interference by overlapping calls, or increasing the detectability of their calls in noise. Despite the robust and directional nature of call modifications in noise, the evolutionary function of these modifications remains obscure.

1:20

4pAB2. Anthropogenic noise constrains acoustic communication in urban-dwelling frogs. Kirsten M. Parris (School of Botany, The Univ. of Melbourne, Parkville, VIC 3010, Australia, k.parris@unimelb.edu.au)

Urban noise may hinder acoustic communication in a diversity of animal groups by reducing the distance over which vocal signals can be detected. Given the importance of such signals for mate attraction and territory defence, this acoustic interference may have wide-ranging consequences for individual fitness. I will present a mathematical model of the active space of frog calls in urban noise as a function of body size. Despite having lower auditory thresholds, larger species with lower-frequency calls are predicted to suffer the greatest reduction in communication distance in noisy urban environments. During a field study in Melbourne, Australia, my colleagues and I found that the southern brown tree frog *Litoria ewingii* called at a higher frequency in traffic noise. However, modeling indicates that the observed frequency shift would confer only a modest increase in active space. Furthermore, as females of certain frog species

appear to prefer lower-frequency advertisement calls, this strategy may improve the audibility of calls but reduce attractiveness to potential mates. Calling more loudly would result in a larger increase in active space, but the high metabolic cost of this strategy could limit chorus tenure and ultimately reduce breeding success.

1:40

4pAB3. Acoustic invasion: How invasive species can impact native species acoustic niche? Camila Both (Programa de Pós-graduação em Ecologia e Evolução, Universidade Federal de Goiás, Campus Samambaia, Cx. 131, Goiânia, Goiás 74001970, Brazil, camilaboth@gmail.com) and Taran Grant (Instituto de Biociências, Universidade de São Paulo, São Paulo, Brazil)

The effects of invasive species on native taxa due to direct predation, food, and space competition, and disease transmission are well documented. However, the effects of acoustic invaders on animal communication have not been explored. We simulated an invasion of the acoustic niche by exposing calling native male white-banded tree frogs (*Hypsiboas albomarginatus*, harmonics at 60–1430 Hz and 2720–2780 Hz or 2280–2850 Hz) to recorded calls of the invasive American bullfrog (*Lithobates catesbeianus*, frequencies from 90 to >4000 Hz) at a non-invaded site in the Brazilian Atlantic Forest. In response, tree frogs immediately shifted calls to significantly higher frequencies. In the post-stimulus period, they continued to use higher frequencies and also decreased signal duration. Tree frogs did not change calling rate or inter-call interval. Acoustic signals are the primary basis of mate selection in many anurans, and such changes could negatively affect the reproductive success of native species. The effects of bullfrog vocalizations on acoustic communities are expected to be especially severe due to their broad frequency band, which masks the calls of multiple species simultaneously. These results show that invasive species could affect native species by interfering in their acoustic niche.

Contributed Paper

2:00

4pAB4. Impacts of acoustic competition between invasive Cuban treefrogs and native treefrogs in southern Florida. Jennifer B. Tennessen (Dept. of Biology, Penn State Univ., 208 Mueller Lab., University Park, PA 16802, jbt148@psu.edu), Susan E. Parks (Dept. of Biology, Syracuse Univ., Syracuse, NY), Ray W. Snow (U.S. Dept. of the Interior, National Park Service, Everglades National Park, Homestead, FL), and Tracy L. Langkilde (Dept. of Biology, Penn State Univ., University Park, PA)

The natural acoustic environment has undergone substantial changes over the past century due to human activities, creating novel soundscapes. Much research has focused on the impacts of anthropogenic noise on acoustic communication, including noise from transportation, construction, energy development, and defense. The impact of acoustic invasive species has been largely overlooked in bioacoustic studies on the behavioral and ecological consequences of noise. We conducted a passive monitoring experiment and a playback experiment to quantify the impact of invasive Cuban treefrog (*Osteopilus septentrionalis*) acoustic signals on the acoustic environment and on native treefrog acoustic behavior. Our results show that Cuban treefrog chorus altered the soundscape in Everglades National Park and affected the acoustic behavior of native treefrogs. Collectively, these results suggest that acoustic invasive species are important yet rarely considered sources of noise that can have ecological consequences at scales ranging from the individual to the ecosystem.

Invited Papers

2:20

4pAB5. Modification of humpback whale social sound repertoire and vocal source levels with increased noise. Rebecca Dunlop, Michael Noad (School of Veterinary Sci., Univ. of Queensland, Cetacean Ecology and Acoustics Lab., Gatton, QLD QLD 4343, Australia, r.dunlop@uq.edu.au), and Douglas Cato (Inst. of Marine Sci., Univ. of Sydney, Sydney, NSW, Australia)

In acoustic communication, high background noise is an important obstacle in successful receiver signal detection and perception of an intended acoustic signal. To overcome this problem, many animals modify acoustic signals by increasing the repetition rate, duration, amplitude, or frequency range of the signal. Humpback whales are the most vocal of the baleen species in that they use a wide and varied catalog of social sounds. More than 36 different sound types (vocal sounds and surface-generated sounds from energetic surface behaviors) were found during a three year study on migrating humpback whales. During periods of high wind noise (where there were no audible boats or singing whales in the area), humpback whales modify both their acoustic repertoire as well as vocal signal properties. We found that humpback whale groups gradually switched from primarily vocal to primarily surface-generated communication in increasing wind speeds and background noise levels, but kept both signal types in their repertoire. We also found evidence of the Lombard effect, in that in increased wind-dominated background noise levels, humpback whale groups tended to increase the amplitude of their vocalizations. Determining how whales modify their vocal behavior in increasing levels of background noise will give us an important insight into how they might cope with increasing levels of anthropogenic noise.

2:40–3:00 Break

3:00

4pAB6. Variation in the vocal behavior of southern right whales (*Eubalaena australis*) in coastal Brazilian waters. Susan Parks (Biology Dept., Syracuse Univ., 114 Life Science Complex, Syracuse, NY 13244, sparks@syr.edu), Karina Groch (Projeto Baleia Franca, Instituto Australis, Florianópolis, SC, Brazil), Paulo A. C. Flores (Centro Mamíferos Aquáticos – CMA, Centro Nacional de Pesquisa e Conservação de Mamíferos Aquáticos, ICMBio, MMA – CMA SC, Florianópolis, SC, Brazil), Renata S. Sousa-Lima (LaB - Laboratório de Bioacústica, Departamento de Fisiologia, Centro de Biociências, Universidade Federal do Rio Grande do Norte, Natal, RN, Brazil), and Ildar R. Urazghildiiev (Bioacoustics Res. Program, Cornell Univ., Ithaca, NY)

Currently, there are three recognized species of right whales. The largest population is the southern right whale (*Eubalaena australis*), with circumpolar distribution in the southern hemisphere. One calving area for this population is in Brazilian waters, where increasing numbers of right whales have been sighted over the past decade along with an increase in anthropogenic activities including shipping traffic and fishing. The goals of this study were to describe the vocal behavior of southern right whales in Brazilian waters, assess the difference in vocalizations between areas with low and high human activity, and compare these results to studies conducted with North Atlantic right whales (*Eubalaena glacialis*) in the Western North Atlantic. Bottom-mounted archival acoustic recorders were deployed in October and November 2011 in two coastal locations in central Santa Catarina State, southern Brazil. One recorder was placed off Gamboa (27056'S and 48039'W, low traffic) and a second off Ribanceira (28011'S and 48037'W, high traffic). Automated detectors and noise statistic analysis tools developed for North Atlantic right whale upcalls were utilized to analyze the dataset. Calls produced by Brazilian right whales were significantly lower in fundamental frequency than North Atlantic right whale calls and the implications for these results will be discussed.

3:20

4pAB7. Are there metabolic costs of vocal responses to noise in marine mammals? Marla M. Holt (Conservation Biology Div., NOAA NMFS Northwest Fisheries Sci. Ctr., Seattle, WA), Robin C. Dunkin (Long Marine Lab., Dept. of Ecology and Evolutionary Biology, Univ. of California, 100 Shaffer Rd., Santa Cruz, CA 95060, dunkin@biology.ucsc.edu), Dawn P. Noren (Conservation Biology Div., NOAA NMFS Northwest Fisheries Sci. Ctr., Seattle, WA, United Kingdom), and Terrie M. Williams (Dept. of Ecology and Evolutionary Biology, Univ. of California, Santa Cruz, Santa Cruz, CA)

Many species respond to increases in environmental noise by increasing the amplitude, duration, and/or repetition rate of their vocalizations. Potential costs of noise-induced vocal modifications include increased energetic costs but no empirical data in marine mammals exist. This study's objective was to compare the metabolic costs of communicative sounds produced by captive bottlenose dolphins ($N = 2$) under two conditions (low- and high-amplitude vocalization trials) to assess energetic costs of vocal responses to noise. An experimental trial consisted of a 10-min rest period to determine resting metabolic rate, followed by a two-minute vocalization period, and concluded with another 10-min rest period to measure recovery. Open-flow respiratory was used to measure oxygen consumption during each trial component. Vocalizations were recorded using a calibrated hydrophone for analysis. Both dolphins tended to produce longer vocalizations during high-amplitude trials. Thus, metabolic rates were related to total sound energy of all vocalizations produced during the vocal period for each trial. Metabolic costs tended to be higher during high sound energy trials, but only verged on statistical significance when vocal-performance differences were at least 10 dB (in cumulative sound exposure level). This study provides key data to assess biological consequences of anthropogenic noise exposure in marine mammals.

3:40

4pAB8. Vocal modifications in primates: Effects of noise and behavioral context on vocalization structure. Cara F. Hotchkin (NAVFAC Atlantic, 6506 Hampton Blvd, Norfolk, VA 23508, hotchkin.cara@gmail.com), Susan E. Parks (Dept. of Biology, Syracuse Univ., Syracuse, NY), and Daniel J. Weiss (Dept. of Psych., The Penn State Univ., University Park, PA)

During increased noise, modifications of the acoustic structure of vocalizations (amplitude, temporal, and spectral parameters) may allow release from masking, potentially conferring fitness benefits to vocally flexible signalers. Among primates, humans have demonstrated extreme vocal flexibility during noise, with modifications to all three speech parameters affected by both noise type and motivational state of the signaler. While non-human primates have also demonstrated changes to call amplitude and temporal characteristics, to the best of our knowledge spectral modifications have not been observed and the influence of behavioral context remains unknown. This experiment used playbacks of broad (10 kHz) and narrowband (5 kHz) white noise to investigate the effects of noise level and bandwidth on chirps and combination long calls (CLCs) produced by cotton-top tamarins. Noise amplitude and frequency content both influenced the structure of vocalizations; modifications included increased call amplitude (the Lombard effect), changes to call durations, and previously undocumented spectral shifts. Behavioral context was also relevant; modifications to CLCs were different from those observed in chirps. These results provide the first evidence of noise-induced spectral shifts in non-human primates, and emphasize the importance of behavioral context in vocal noise compensation.

Contributed Papers

4:00

4pAB9. Active ultrasonic vocal communication channel found in Mongolian gerbils through the cochlear microphonics with noise exposure. Hiroshi Riquimaroux, Keizo Fukushima, and Kohta I. Kobayasi (Life and Med. Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan, hrikimar@mail.doshisha.ac.jp)

Mongolian gerbils (*Meriones unguiculatus*) use ultrasonic vocal communication in frequency range of 22–45 kHz. However, hearing threshold of this frequency range reported has been very high, which is not suited for vocal communication. We examined possible active amplification created by the outer hair cells for frequency range of 22–45 kHz. In this study, we evaluated the amount of active amplification by the cochlear microphonics (CM) combined with temporary damage created by noise exposure. Adult gerbils received surgical implantation of a silver wire electrode on the round window of their cochlea through the middle ear to record CM. They were exposed to broadband noise (0.5 to 45 kHz) at 90 dB SPL for 5 min. CMs were recorded for tone bursts of 1 to 45 kHz. The following results were obtained. First, we observed the largest CM reduction just after the noise exposure. Second, decrements in CM amplitude depended on frequency. Low sensitivity frequency range above 22 kHz produced large reduction in CM amplitude. Third, decrease in CM amplitude was greater for lower stimulus intensities. Fourth, for testing frequencies, which produced large CM decrements, it took a longer period to recover back to pre-noise exposure amplitude levels. These findings indicate that reduction in CM amplitude appeared to be related to the cochlear nonlinearity generated by the outer hair cells.

4:20

4pAB10. Benign exclusion of birds using acoustic parametric arrays. Eric A. Dieckman, Elizabeth Skinner (Dept of Appl. Sci., College of William and Mary, P.O. Box 8795, Williamsburg, VA 23187, eric.dieckman@gmail.com), Ghazi Mahjoub, John Swaddle (Dept. of Biology, College of William and Mary, Williamsburg, VA), and Mark Hinders (Dept. of Appl. Sci., College of William and Mary, Williamsburg, VA)

Excluding birds from areas can be important in aviation safety, agriculture, and facilities maintenance. Presenting audible stimuli or predator vocalizations in the affected area often has initial success, but has a limited effect over the long-term, even if the signals are varied to reduce the chances of the birds habituating to the sounds and objects. Many birds are highly vocal and rely on auditory communication in almost every aspect of their life history. By creating noise specifically targeted to be within the vocal range of the

nuisance species, we hypothesize that the birds will be less able to communicate and will move to more acoustically suitable environments. To avoid introducing noise pollution to the surrounding environment we create spatially well-controlled "sonic nets" using a mix of speakers and acoustic parametric arrays. To better understand the interaction of the sound field and the environment we combine finite difference solutions of the KZK equation with 3D acoustic finite integration simulation. These simulations allow us to propagate a nonlinear acoustic beam to a real-world target and then study the scattering from the target. We discuss initial experiments with a parametric array in an aviary on the exclusion of starlings from a food source.

THURSDAY AFTERNOON, 6 JUNE 2013 — 510D, 1:20 P.M. TO 4:20 P.M.

Session 4pAO

Acoustical Oceanography and Underwater Acoustics: Biologic and Non-Biologic Scatterers

James Lynch, Cochair
WHOI, Bigelow 203, WHOI, Woods Hole, MA 02543

Ikuo Matsuo, Cochair
Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan

Contributed Papers

1:20

4pAO1. Acoustic scattering from a water-filled cylindrical shell: Mode identification and interpretation via finite element and analytical models. Aubrey L. Espana, Kevin L. Williams (Acoust. Dept., Appl. Phys. Lab. - Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105, aespana@apl.washington.edu), Daniel S. Plotnick, and Philip L. Marston (Dept. of Phys. and Astronomy, Washington State Univ., Pullman, WA)

Understanding the physics governing the interaction of sound with targets in an underwater environment is essential to improving upon existing target detection and classification algorithms. Simple models are viable tools for meaningful interpretation of scattering results. To illustrate this, two modeling techniques are employed to study the acoustic scattering from a water-filled cylindrical shell. The first model is a hybrid 2-D/3-D finite element (FE) model, whereby the scattering in close proximity to the target is handled via a 2-D axisymmetric FE model, and the subsequent 3-D propagation to the far-field is determined via a Helmholtz integral. This model is characterized by the decomposition of the fluid pressure and its derivative in a series of azimuthal Fourier modes, a technique that has previously facilitated mode identification [Espana *et al*., J. Acoust. Soc. Am. **130**, 2332 (2011)]. The second is an analytical solution for an infinitely long cylindrical shell, coupled with a simple approximation that converts the results to an analogous finite length form function. These two model results, when examined together on a mode-by-mode basis, offer visualization of the mode dynamics and the ability to distinguish the different physics driving the target response (i.e., structural modes versus water-waveguide modes). [Work supported by ONR.]

1:40

4pAO2. Testing of an extended target for use in high frequency sonar calibration. John L. Heaton (Mech. Eng., Univ. of New Hampshire, Ctr. for Coastal and Ocean Mapping/Joint Hydrographic Ctr., 24 Colovos Rd., Durham, NH 03824, jheaton@ccom.unh.edu), Thomas C. Weber (Ocean/Mechanical Eng., Univ. of New Hampshire, Ctr. for Coastal and Ocean Mapping/Joint Hydrographic Ctr., Durham, NH), Glen Rice (NOAA Office of Coast Survey, Univ. of New Hampshire, Ctr. for Coastal and Ocean Mapping/Joint Hydrographic Ctr., Durham, NH), and Xavier Lurton (IMN/NSE/AS, IFREMER, Plouzane, Bretagne, France)

Acoustic backscatter tests were performed in a tank with a 200-kHz, 7°, SIMRAD EK60 Split-Beam Echo-Sounder, and a 256-beam RESON SeaBat 7125 Multi-Beam Echo-sounder. Tests were done in order to investigate the angular and range dependency of the scattering strength of a new test target in order to validate its use in sonar testing. This target was constructed of small chain links arranged in a "curtain" simulating an extended scattering surface, such as the seafloor. Target strength for individual links was collected as the links were rotated 360°. The links are combined into an extended surface target, spacing between scatterers being approximately 1 cm. The scattering network irregularity is enough to ensure random phase at the wavelength considered. The target scattering strength was measured as a function of grazing angle and range, hence varying the number of scatterers within the beam footprint. These tests suggest that the amplitude envelope of the scattered signals is Rayleigh distributed and that the backscatter strength depends linearly on the number of active scatterers, all desirable features for calibrating sonars used to make measurements of similarly random surfaces such as the seafloor. Results show a promising in-tank calibration technique when extended surface targets are desirable.

2:00

4pAO3. Observing natural methane seep variability in the northern Gulf of Mexico with an 18-kilohertz split-beam scientific echosounder. Kevin Jerram, Thomas C. Weber, and Jonathan Beaudoin (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, kjerram@ccom.unh.edu)

Underwater methane seeps support diverse biological communities on the seafloor and, in cases of bubble survival to the surface, contribute to the quantity of atmospheric methane. The National Oceanic and Atmospheric Administration (NOAA) ship *Okeanos Explorer* completed two research cruises for seep mapping and characterization in the northern Gulf of Mexico during August and September of 2011 and April of 2012. Seeps originating at depths of approximately 1500 m were observed during multiple transects with a 30-kHz Kongsberg EM 302 multibeam echosounder (MBES) and an 18-kHz Simrad EK60 split-beam scientific echosounder calibrated for backscatter. A methodology for determining vessel offsets for the EK60 using MBES seep observations as benchmarks is discussed as part of a larger framework for transformation of seep targets from the split-beam echosounder reference frame to the geographical reference frame. Utilizing sound speed and attitude data collected for the MBES, several EK60 observations of strong individual seeps are scrutinized for variability of seep position and target strength between 2011 and 2012.

2:20

4pAO4. *In situ* measurement of the individual target strength of crustacean zooplankton with concurrent optical identification. Christian Briseño-Avena, Jules S. Jaffe, Paul L. Roberts, and Peter J. Franks (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0208, cbriseno@ucsd.edu)

Acoustic methods are common tools in the study of zooplankton distributions and behaviors. However, relating acoustic scattering to zooplankton abundance is difficult because zooplankton acoustic properties have been

challenging to obtain *in situ*. Most of the acoustic measurements of zooplankton come from either preserved or recently dead organisms, or are derived from computer models. Other attempts to measure *in situ* acoustic target strength (TS) of zooplankton have targeted larger taxa such as euphausiids and amphipods. *In situ* measurements of copepods and other small crustaceans are extremely desirable as these taxa can be dominant in marine ecosystems. Here we present *in situ* measurement of TS using frequencies of 1.5–2.5 MHz with concurrent stereoscopic imaging. The concurrent calibration of the optics and acoustics permits the quantification of individual acoustic TS associated with the individual organism that gave rise to the echo. Furthermore, new technological advances have allowed us to measure organisms with TS as small as −125 dB. The results of this work will permit improvements in extant acoustic models, and enhance our interpretation of acoustic data collected in the field.

2:40

4pAO5. Complimentary ultrasound methods for the estimation of sound speed in macroalgae. Jo Randall (Environ. Hydroacoustics Lab., Universite Libre de Bruxelles, Bruxelles, Belgium), Jean-Pierre Hermand (Environ. Hydroacoustics Lab., Universite Libre de Bruxelles, Ave. Franklin Roosevelt, Bruxelles B-1050, Belgium, jhermand@ulb.ac.be), Marie-Elise Arnould (Laborelec, Linkebeek, Belgium), Jeff Ross, and Craig Johnson (Inst. for Marine and Antarctic Studies, Univ. of Tasmania, Sandy Bay, TAS, Australia)

Temperate kelp forests are among the most productive ecosystems in the world. However, there is mounting evidence that these habitats are in decline, both in range and productivity. Acoustic propagation modeling has been used to identify primary productivity in seagrass beds, and work is ongoing in development as a method of providing large scale measurements of productivity in macroalgae forests. Acoustic predictive models require knowledge of the material properties of interest, yet little is known about the acoustic properties of seaweed species. As a preliminary step towards acoustic modeling of seaweed systems, this study investigates the acoustic properties of *Ecklonia radiata*, a key species in temperate Australian marine systems. Measuring sound speed in macroalgae, as with other biological material, provides unique challenges due to their intrinsic morphological and anatomical characteristics. Using a range of frequencies between 2 and 10 MHz different methods are proposed to measure sound speed both directly and indirectly. The measurements show a consistent result, with variation according to tissue type. This research provides an important first step toward the development of acoustic propagation models in kelp forest ecosystems.

3:00–3:20 Break

3:20

4pAO6. Modeling the acoustic scattering from large fish schools using the Bloch-Floquet theorem. Jason A. Kulpe, Michael J. Leamy, and Karim G. Sabra (Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, jkulpe@gatech.edu)

Scattering from large fish schools can significantly contribute to volume reverberation in the open ocean measured by mid-frequency long-range SONAR systems (1–10 kHz). This can potentially cause large false-alarms, especially if the resonance frequencies of the fish' air-filled swim bladder is excited. Hence, to ultimately improve the detection performance of long-range SONAR systems, we seek an efficient modeling technique for the acoustic scattering created by large school of fish, which readily accounts for the fish bio-acoustic properties, school's spatial configuration and multiple scattering effects. We exploit here a key observation to simplify our problem: fish in larger schools tend to swim in a periodic arrangement whereby we approximate the large school as an infinite system with a periodic collection of fish' air filled swimbladders. Thus, the Bloch-Floquet theorem, governing waves in periodic media, allows predictions of the acoustic field in an infinite media by simply modeling the dynamic response of a single unit cell only (containing one fish). This approach allows one to rapidly predict the frequency dependent reflection and transmission coefficient on a semi-infinite fish school for various incident waves. Good agreement was found with the results obtained from finite element modeling of realistic, finite sized fish schools.

3:40

4pAO7. Clustering of acoustic fish features tracked by broadband split-beam echo sounder. Masanori Ito, Ikuo Matsuo (Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, ito@cs.tohoku-gakuin.ac.jp), Tomohito Imaizumi, Tomonari Akamatsu (National Res. Inst. Fisheries Eng., Fisheries Res. Agency, Kamisu, Japan), Yong Wang, and Yasushi Nishimori (Furuno Electric Co., Nishinomiya, Japan)

Monitoring fish species and amount with acoustical instruments is a challenging problem to survey fish resources in the ocean. Broadband split-beam echo sounder was useful to observe individual fish behavior in schools. Echoes from fish schools were measured from an anchored vessel for several hours. Echo signals were gathered into one block within a certain depth and a period of time and analyzed to track individual fish in the schools. Target strength was calculated from the individual fish echo and associated with tilt angle which could be estimated by using the tracking result. Feature in each block was statistically calculated by averaging target strengths according to the tilt angles. The blocks of the signals were clustered by K-means method with the features and divided into some clusters. The distribution of the clusters in time and depth was investigated. It appeared that the distributions of the clusters were dependent on both time and depth. Clustering of the features would be effective to monitor diversity of fish in the ocean. [Research supported by JST, CREST.]

4:00

4pAO8. Scattering and reverberation from fish schools in the 500–1500 Hertz band. Arthur Newhall, James Lynch, Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr., Bigelow 213, WHOI, Woods Hole, MA 02543, anewhall@whoi.edu), Thomas Grothues (Inst. of Marine and Coastal Sci., Rutgers Univ., Nw Brunswick, NJ), and Glen Gawarkiewicz (Physical Oceanogr., Woods Hole Oceanogr., Woods Hole, MA)

We report here on the preliminary results from an experiment off Cape Hatteras, North Carolina, to look at acoustic scattering and reverberation from fish schools in the 500–1500 Hz band. The experiment, which was performed during the period May 12–29, 2012, was a joint acoustics, biology and physical oceanography effort, with distinct, but coordinated, goals in each area. Acoustically, our goal was to examine the scattering of sound from fish schools over a full range of azimuthal angles. To do this, we employed a source mounted on an autonomous vehicle and a moored, four element hydrophone array receiver. The source traveled around the fish school and the receiver, giving the desired angular diversity. Biologically, we were interested in mapping and imaging/classifying the fish (both individually and as schools) with the sidescan sonars on the vehicles, and contrasting/verifying this information with video images from high definition cameras attached to the vehicles. Oceanographically, the correlation between the ocean temperature field and the fish species encountered was of first order interest. Results from all three areas will be presented, including some interesting video images, and directions for analysis and further research will be discussed. [Work sponsored by the Office of Naval Research.]

Session 4pBA

Biomedical Acoustics: High-Frequency Ultrasound (20–80 MHz)

Michael Oelze, Chair
UIUC, 405 N Mathews, Urbana, IL 61801

Chair's Introduction—12:55

Invited Papers

1:00

4pBA1. Acoustic and photoacoustic imaging of spheroids. Michael C. Kolios, Elizabeth S. Berndl, Lauren C. Wirtzeld, Eric M. Strohm (Phys., Ryerson Univ., 350 Victoria St., Ontario, Toronto, ON M5B2K3, Canada, mkolios@ryerson.ca), and Gregory J. Czarnota (Med. Biophys., Univ. of Toronto, Toronto, ON, Canada)

Acoustic and photoacoustic high frequency imaging (50–100 MHz) can be used to generate images of cell constructs and spheroids with good spatial resolution and contrast. Here we demonstrate how co-registered acoustic and photoacoustic imaging can be used for imaging spheroids. Spheroids are widely used in cancer research and biology since they emulate a three-dimensional environment such as that experienced in tumors. Spheroids were made by the hanging-drop method using the MCF-7 cancer cell line. To generate photoacoustic contrast, MCF-7 cells were incubated with optical absorbing nanoparticles (e.g., gold nanorods, 780 nm absorption) for 24 h and mixed with native MCF-7 cells prior to spheroid formation. The spheroids were between 0.5 mm and 1mm in diameter. Imaging was performed with the VisualSonics VEVO 770 (25–55 MHz) and a high-resolution SASAM acoustic/photoacoustic microscope for frequencies over 80 MHz (Kibero GmbH, Germany). The spheroid was imaged first using pulse echo ultrasound, then with photoacoustics immediately after. The necrotic core of the spheroid had a 20 dB increase in ultrasound backscatter compared the viable cells surrounding the core, and the ultrasound/photoacoustic images of the spheroid were co-registered showing the distribution of the optical absorbing agents.

1:20

4pBA2. High-frame-rate retrospective imaging of mouse-embryo cardiac function using annular array and Doppler-derived gating. Jeffrey A. Ketterling, Erwan Filoux (Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., New York, NY 10038, jketterling@riversideresearch.org), Orlando Aristizabal (Skirball Inst. of Biomolecular Medicine, NYU School of Medicine, New York, NY), and Jonathan Mamou (Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY)

A high-frequency (HF) imaging system based on a custom 5-element, 40 MHz annular array has been used to study the cardiovascular development of mouse embryos. High-frame-rate imaging of the heart dynamics was achieved using a retrospective reconstruction method based on Doppler-derived electrocardiogram (ECG) waveforms and respiratory gating was used to suppress motion artifacts. The ECG signals were obtained by measuring blood-flow velocities in major arteries of *in-vivo* mouse embryos using a custom HF Doppler apparatus made from two 20 MHz, single-element, PZT transducers with a Doppler sample volume of 15 mm. Co-registered M-mode data were acquired from the annular array excited with a 5-channel pulser/receiver. A synthetic-focusing (SF) algorithm was used to improve spatial resolution (<100 μm), depth-of-field (>10 mm) and signal-to-noise ratio (>45 dB). This technique was used on embryos aged from 11.5 to 14.5 days and provided high-resolution, morphologically correct B-mode cine-loops of the heart chamber dynamics at frame rates of 1 kHz. The ultra-fine temporal resolution (1 ms) allowed for precise quantification of the mean cardiac cycle length and detailed visualization of fast events such as opening and closing of the mitral valve. The speckle characteristics of the high-resolution images could be used to assess blood flow and to quantify myocardial strain at each developmental stage of the embryonic heart.

1:40

4pBA3. Quantitative ultrasound evaluation of tumor cell death response in locally advanced breast cancer patients to chemotherapy treatment administration. Gregory Czarnota, Ali Sadeghi-Naini, Naum Papanicolau, Omar Falou (Imaging Res. and Radiation Oncology, Sunnybrook Health Sci. Ctr., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, gregory.czarnota@sunnybrook.ca), Rebecca Dent, Sunil Verma, Maureen Trudeau (Medical Oncology, Sunnybrook Health Sci. Ctr. and the Univ. of Toronto, Toronto, ON, Canada), Jean-Francois Boileau (Surgical Oncology, Sunnybrook Health Sci. Ctr. and the Univ. of Toronto, Toronto, ON, Canada), Jacqueline Spayne (Radiation Oncology and Medical Biophys., Univ. of Toronto, Toronto, ON, Canada), Sara Iradji, Ervis Sofroni, Justin Lee (Imaging Res. and Radiation Oncology, Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), Sharon Lemon-Wong (Nursing, Odette Cancer Ctr. and Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), Martin Yaffe (Imaging Res. and Radiation Oncology, Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), and Michael Kolios (Phys., Ryerson Univ., Toronto, ON, Canada)

A clinical study was undertaken investigating the efficacy of ultrasound to quantify cell death in tumor responses with cancer treatment. Patients ($n = 25$) with locally advanced breast cancer received anthracyline and taxane-based chemotherapy treatments over four to six months. The majority of patients went on to have a modified radical mastectomy and correlative whole mount histopathology. Data collection was carried out using an Ultrasonix-RP and an L15-5 6 cm transducer pulsed at 10 MHz with RF data collected five times during neoadjuvant chemotherapy. Data indicated increases of approximately 9 dBr (+/−1.67) maximally in ultrasound backscatter in patients who clinically responded to treatment. Patients assessed as responding poorly demonstrated significantly lower increases

(2.3 +/− 1.7 dBr). Increases in 0-MHz intercept followed similar trends while increases in spectral slope were observed locally from tumor regions demonstrating increases in tissue echogenicity. This study demonstrates the potential of ultrasound to quantify changes in tumors in response to cancer treatment administration in a clinical setting. The results indicate that such responses can be detected early during a course of chemotherapy and should permit ineffective treatments to be changed to more efficacious ones potentially leading to improved treatment outcomes.

2:00

4pBA4. High-frequency quantitative ultrasound approaches for cancer detection in freshly-excised lymph nodes. Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@riversideresearch.org), Alain Coron (Laboratoire d'Imagerie Paramétrique, CNRS and UMPC, Univ Paris 06, Paris, France), Emi Saegusa-Beecroft (Dept. of Surgery, Kuakini Med. Ctr. and Univ. of Hawaii, Honolulu, HI), Masaki Hata (Dept. of Surgery, Juntendo Med. Ctr., Tokyo, Japan), Michael L. Oelze (Bioacoustics Res. Lab., Univ. of Illinois, Urbana-Champaign, IL), Eugene Yanagihara (Dept. of Surgery, Kuakini Med. Ctr. and Univ. of Hawaii, Honolulu, HI), Tadashi Yamaguchi (Res. Ctr. for Frontier Med. Eng., Chiba Univ., Chiba, Japan), Pascal Laugier (Laboratoire d'Imagerie Paramétrique, CNRS and UMPC, Univ Paris 06, Paris, France), Junji Machi (Dept. of Surgery, Kuakini Med. Ctr. and Univ. of Hawaii, Honolulu, HI), and Ernest J. Feleppa (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY)

Histology performed to assess lymph nodes excised during node-dissection surgeries from cancer patients suffers an unsatisfactory rate of false-negative determinations due to labor and time constraints. In this study, more than 300 lymph nodes were scanned in 3D using a 26-MHz high-frequency ultrasound transducer. Following scanning, individual nodes underwent a special histology procedure that involved step-sectioning each node at 50-μm intervals to guarantee that no significant cancer foci were missed. The 3D radio-frequency ultrasound dataset was analyzed using overlapping 3D regions-of-interests that were individually processed to yield 13 quantitative ultrasound (QUS) estimates associated with tissue microstructure and were hypothesized to show contrast between normal and cancerous regions in lymph nodes. Step-wise linear discriminant analyses were performed to yield an optimal QUS-based classifier. ROC curves and areas under the ROC curves (AUCs) were obtained to assess cancer-detection performance. The AUC for the linear combination of four QUS estimates was 0.83 for a dataset of 110 axillary nodes of breast-cancer patients. Similarly, using five QUS estimates, an AUC of 0.97 was obtained for a dataset of 180 nodes of gastrointestinal-cancer patients. These studies demonstrate that QUS methods may provide an effective tool to guide pathologist towards suspicious regions in lymph nodes.

2:20

4pBA5. Radial shear strain elastography imaging of carotid atherosclerotic plaques in a porcine model. Guy Cloutier, Younes Majdouline, Damien Garcia, Louise Allard, Sophie Lerouge (Lab. of Biorheology and Med. Ultrason., Univ. of Montreal Hospital, 2099 Alexandre de Sève, Montreal, QC H2L 2W5, Canada, guy.cloutier@umontreal.ca), Jacques Ohayon (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital, Grenoble, France), and Gilles Soulez (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital, Montreal, QC, Canada)

The objective is to show the feasibility of shear strain elastography (SSE) *in vivo* with intravascular ultrasound (IVUS) radio-frequency data acquired at 20 MHz in carotid arteries of pigs. We previously proposed the Lagrangian Speckle Model Estimator (LSME) to estimate the strain tensor including the shear strain that could be involved in atherosclerotic plaque hemorrhage, inflammation, and rupture mechanisms. However, the LSME performance to compute SSE had never been validated. Atherosclerotic pigs with significant plaques on carotids were studied. To induce atherosclerosis, pigs were put on an atherogenic diet and partial ligations of common carotid arteries were performed. Diabetes was induced by selective intra-arterial injection of streptozotocin. IVUS acquisitions were performed before sacrifice and histologic analysis were realized on fixed dissected carotids. SSE maps at end diastole were estimated with a new implementation of the LSME and matching with histology sections was realized. SSE clearly identified all plaques; reported figures show SSE mapping characterized by cohabitation of high positive and high negative shear values in the specific region of the plaque. This study demonstrates the performance of the LSME implementation to estimate accurately the shear strain distribution, and the feasibility of SSE to highlight atherosclerotic plaque vulnerability characteristics.

2:40

4pBA6. A method to validate quantitative high-frequency power Doppler ultrasound with fluorescence *in vivo* video microscopy. James C. Lacefield (Dept. of Med. Biophys., Western Univ., Thompson Engineering Bldg., Rm. 279, London, ON N6A 5B9, Canada, jlacefie@uwo.ca), Stephen Z. Pinter (Biomedical Eng. Graduate Program, Western Univ., London, ON, Canada), Dae-Ro Kim (Dept. of Med. Biophys., Western Univ., London, ON, Canada), M. Nicole Hague (London Regional Cancer Program, London, ON, Canada), Ian C. MacDonald (Dept. of Med. Biophys., Western Univ., London, ON, Canada), and Ann F. Chambers (London Regional Cancer Program, London, ON, Canada)

Flow quantification with high-frequency power Doppler ultrasound can be performed using the wall-filter selection curve (WFSC) method [Elfarnawany *et al.*, Ultrasound Med. Biol. **38**, 1429–1439 (2012)]. The WFSC method plots color pixel density (CPD) as a function of wall filter cut-off velocity as a means of objectively selecting an operating point cut-off velocity. In this study, an *in vivo* video microscopy (IVVM) system was used to measure the size of small (140–400 μm diameter) mouse testicular vessels immediately after the vessels were imaged with 30 MHz power Doppler. The mouse remained on the same platform throughout ultrasound and IVVM imaging. Measurements in four image planes from three mice demonstrated that, similar to previously reported flow-phantom data, *in vivo* WFSCs exhibit distinct, sloped "characteristic intervals" at cut-off velocities where the CPD approaches the gold-standard IVVM estimate of vascular volume fraction. A wide range of operating point cut-off velocities (4.5 to 12 mm/s) was obtained, which indicates that use of a predetermined cut-off can produce substantial errors in cross-sectional studies that employ power Doppler to quantify vascularity. The WFSC method is a promising strategy for adapting the cut-off velocity to intersubject and longitudinal variations in blood flow during microvascular imaging experiments.

3:00–3:20 Break

3:20

4pBA7. Determining breast pathology in surgical margins with high-frequency ultrasound: phantom and numerical simulations. Timothy E. Doyle, Monica Cervantes (Physics, Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999, Timothy.Doyle@uvu.edu), Laurel A. Thompson (Chemistry, Utah Valley Univ., Orem, UT), Joseph E. Roring, Matthew A. Grover (Physics, Utah Valley Univ., Orem, UT), J. Andrew Chappell, Bradley J. Curtis, Janeese E. Stiles, and Brett D. Borget (Biology, Utah Valley Univ., Orem, UT)

Two parameters in high-frequency ultrasound (20–80 MHz) have been found to be sensitive to a range of pathologies in resected margins from breast conservation surgery: The number of peaks (the peak density) in the waveform spectrum and the slope of the Fourier transform of the waveform spectrum. Previous studies have indicated that peak density and slope may correlate to microscopic heterogeneity in tissue structure, which is modified by atypical and malignant processes. To test this hypothesis, through-transmission and pulse-echo measurements were acquired from gelatin-based phantoms containing polyethylene microspheres and nylon fibers (2.5–10% volume concentration). Multipole methods were also used to model through-transmission measurements of tumor progression in lobular carcinoma *in situ*. The simulated breast tissue contained 1000–2000 nucleated cells with random lobular cavities. The peak densities of the heterogeneous phantoms were significantly greater than those of the homogeneous control samples, whereas the slopes were less. Similarly, the models produced spectra with peak densities that increased with malignant cell proliferation. The results are consistent with breast tissue data, and provide a physical mechanism for the use of peak density and slope in the imaging of breast tissues with atypical and malignant pathologies. [Work supported by Utah Valley University.]

3:40

4pBA8. Molecular profiling of breast cancer using high-frequency ultrasound. Timothy E. Doyle (Physics, Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999, Timothy.Doyle@uvu.edu), Janice E. Sugiyama, Bradley J. Curtis, Mandy H. Marvel, Marcus J. Payne, and Janeese E. Stiles (Biology, Utah Valley Univ., Orem, UT)

In addition to the traditional classifications based on histopathology, breast cancer can also be classified by gene expression profiling into five molecular subtypes that have been found to be more predictive of patient prognosis and treatment response. The purpose of this study was to determine if high-frequency (HF) ultrasound (20–80 MHz) is sensitive to the molecular subtypes of breast cancer, and can differentiate between the more aggressive subtypes such as triple-negative and Her2+ from the less aggressive, more treatable subtypes. Recently, mutations associated with triple-negative and Her2+ have been discovered that are associated with the actin cytoskeleton, extracellular matrix (ECM), and integrin signaling. These mutations may alter the biomechanical and thus ultrasonic properties of tumor cells. This hypothesis was tested using both numerical and cell culture studies. Multipole expansions were used to simulate micro-level ultrasonic scattering from malignant cells with a range of cytoskeletal and ECM properties. Modest property changes produced large variations in simulated spectra. Cell lines of different molecular subtypes were also cultured as monolayers and tested with HF ultrasound. Results from the cell culture tests and their correlation to the models will be discussed. [Work supported by a Utah Valley University Presidential Fellowship Award.]

4:00

4pBA9. Hepatitis fibrosis characterization by a multiparametric study. Mahmoud Meziri, Razika Bouzitoune (Laboratoire LM2S, Université, BP, 790, RP, Annaba 23000, Algeria, mahmoud.meziri@gmail.com), Naamane Remita (Physique, Université, Skikda, Algeria), Christiano Machado, Wagner.C. A Pereira (COPPE/UFRJ, Université, Rio de Janeiro, Brazil), and Frédéric Padilla (Inserm, U1032, LabTau, Université, Lyon, France)

The ultrasonic tissue characterization (UTC) is primarily based on radio-frequency (RF) signals' analysis. The processing of these signals allowed the estimation of different quantitative ultrasonic parameters (backscattered coefficients—ICB-, velocity—SoS, etc). It is known that the RF contains information that can be used to noninvasively characterize the structural and mechanical properties of tissue. Scatterer diameter (or size) from ICB measurements have been already used to discriminate fibrosis from normal tissue. From these findings, our goal was to evaluate the scatterer diameter to test its potential in the discrimination of fibrosis groups (F0, F1, F2, F3, and F4, METAVIR scale) from 20 *in-vitro* human liver samples, explored at 20 MHz. The mean scatterer diameters (μm) measured were: 42.14 □ 4.90 (F0), 40.18 □ 10.51 (F1), 38.82 □ 6.05 (F3), and 40.30 □ 2.22 (F4). The Kolmogorov-Smirnov test has shown a non-significant level ($p > 0.05$) indicating that the scatterer size estimation alone cannot differentiate between all fibrosis groups; an obvious overlap between groups appears. However, for the two different combinations (ICB, Size) and (ICB, Size, SoS), the discriminant analysis has correctly classified 75% and 85%, respectively, of liver samples at a significant level ($p < 0.00005$). The multiparametric study could play an important role to aid in the diagnostic of liver fibrosis.

4:20

4pBA10. Probability distribution variation in high-frequency ultrasound blood echogenicity under *in-vitro* and *in-vivo* blood flow. Tae-Hoon Bok, Kweon-Ho Nam, Dong-Guk Paeng, and Juho Kim (Ocean System Eng., Jeju National Univ., 102 Jejudaehakno, Jeju 690-756, South Korea, bth012@jejunu.ac.kr)

The dynamic phenomena of erythrocyte aggregation (EA) need to be analyzed statistically since EA varies spatially and temporally. In the present study, the cross-sectional B-mode images were acquired from a mock circulatory system with varying blood flow velocity under steady flow, and the human radial artery using an ultrasound biomicroscopy system at 20 MHz. The kurtosis (K) and skewness (S) coefficients, and the Nakagami parameter (m) were computed for each image. For the *in-vitro* experiment, both K and S increased about 0.87 ± 0.18 and 0.63 ± 0.09, respectively; while m decreased about 0.90 ± 0.20 with increasing blood velocity from 12 to 44 cm/s. *In-vivo* experimental results also showed that K, S, and m varied during a cycle. When the blood velocity varied from 5 to 15 cm/s during a cardiac cycle, K and S increased about 1.42 ± 0.64 and 0.44 ± 0.11, respectively; while m decreased about 0.97 ± 0.26. The *in-vivo* results seemed to be consistent with the *in-vitro* results in the sense that K and S increased with blood velocity while m decreased with velocity. This study suggests that the statistical analysis of blood echogenicity can be useful for *in-vivo* hemorheology and blood characterization. [Work supported by NRF-2012-0005005 and NIPA-2012-H0401-12-2006.]

4:40

4pBA11. Improvement of an intravascular ultrasound elasticity modulus imaging approach for detecting vulnerable atherosclerotic plaques. Zahra Keshavarz-Motamed (Lab. of Biorheology and Med. Ultrason., Univ. of Montreal HospitalRes. Ctr., Montreal, QC, Canada, zahra.keshavarz@crchum.qc.ca), Simon Le Floc'h, Jacques Ohayon (Lab. TIMC-IMAG-UJF-CNRS UMR 5525, Univ. Joseph-Fourier, Grenoble, France), and Guy Cloutier (Lab. of Biorheology and Med. Ultrason., Univ. of Montreal Hospital, Montreal, QC, Canada)

Atherosclerotic plaque rupture is the major cause of acute coronary syndrome, myocardial infarction, and stroke in the western world. Stress concentration is recognized to be a good indicator of vulnerable plaques (VP). The Lagrangian speckle model estimator (LSME) for vascular ultrasound elastography, developed by our group, provides the strain field within the plaque. However, evaluation of the stress field relies on a precise identification of the mechanical properties of plaque components. As a response to this need, our group recently developed an approach called imaging modulography (iMOD). iMOD uses a continuum-mechanics-based segmentation method and the inverse finite-element method to reconstruct elasticity maps (or modulograms) of atheroma plaques based on the radial strain field calculated by the LSME. The present theoretical study was designed to further develop segmentation and optimization procedures of iMOD to incorporate both radial and shear components of the strain tensor. Simulated IVUS images of coronary lesions with

known material properties and known stress fields were used to validate the new iMOD algorithm and assess its robustness and performance in detection and quantification of VPs. The results demonstrate promising benefits of the new optimized iMOD-LSME clinical imaging method for VP detection.

5:00

4pBA12. Acoustical imaging of internal spheroid structures at a variety of frequencies. Elizabeth S. Berndl and Michael C. Kolios (Physics, Ryerson Univ., 350 Victoria St., Toronto, ON M5N 2K3, Canada, eberndl@ryerson.ca)

We have previously shown that ultrasound is capable of identifying apoptosis in individual cells and cell pellets due to changes in the ultrasound attenuation, speed of sound, and backscatter. Spheroids can be used to more accurately model non-vascularized tumors due to their three-dimensional growth pattern, cell-cell interaction, disorganized growth, and development of a necrotic core when grown to sufficiently large sizes. To examine cell death in spheroids due to necrosis or chemotherapy, quantitative ultrasound methods were used on the ultrasound backscatter power spectrum throughout the spheroid. MCF7 spheroids ranging in size from 100 to 1000 μm were probed at 25 and 55 MHz using a VEVO770 high frequency ultrasound machine, and at 80 and 200 MHz using an acoustic microscope. Changes in spheroid structure as the necrotic core develops, and after it is exposed to chemotherapeutic agents were recorded and analyzed. An increase in the ultrasound backscatter amplitude from necrotic cells within the core of the spheroid versus the viable cells around the core was observed. Changes in the ultrasound backscatter were also observed for spheroids treated with chemotherapeutics to induce apoptosis. This work furthers our understanding of non-invasively identifying the viability of cancerous tumors, and the efficacy of chemotherapeutic treatments.

THURSDAY AFTERNOON, 6 JUNE 2013 — 512AE, 1:00 P.M. TO 5:20 P.M.

Session 4pEAa

Engineering Acoustics: Sound Field Control in the Ear Canal

Pablo Hoffmann, Cochair
Aalborg Univ., Fredrik Bajers Vej 7B5, Aalborg 9220, Denmark

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

Invited Papers

1:00

4pEAa1. Individual *in-situ* calibration of insert headphones. Marko Hiipakka and Ville Pulkki (Aalto Univ., Otakaari 5 A, Espoo 02150, Finland, Marko.Hiipakka@aalto.fi)

An important procedure in binaural reproduction is the calibration of headphones, which is commonly achieved by first measuring the headphone transfer functions (HpTFs). The commonly used methods of measuring the HpTF are not applicable for insert headphones, since the inserts block the ear canal entrance and since the transducer ports of the inserts are inside the ear canals. Recently, an alternative technique of obtaining HpTFs of inserts using measurements with in-ear microphones, computational modeling, and electro-acoustic Norton-type source models of the inserts has been proposed. In this study, the technique is evaluated using measurements at the eardrums of eight human subjects and computational modeling with normal human ear canal parameters. In addition, different methods of obtaining the electro-acoustic source model parameters are compared. It is shown that the most reliable method of obtaining the Norton source parameters of insert headphones is through measurements with a miniature-sized particle velocity sensor and several tubes with different cross-sectional diameters as acoustic loads. The evaluations show that the proposed technique of obtaining the HpTFs of insert headphones is accurate and reliable at least up to 8 kHz, which bolsters the applicability of the technique for individual *in-situ* calibration in binaural reproduction.

1:20

4pEAa2. Sound transmission in a simple model of the ear canal and tympanic membrane. Antonio Gonzalez-Herrera (Civil Eng., Univ. of Malaga, Calle Diego de Siloe, 29013 Malaga, Malaga, Spain, agh.uma.es@gmail.com), Kapil Wattamwar (Biomedical Eng., Columbia Univ., New York, NY), Christopher Bergevin (Phys. and Astronomy, York Univ., Toronto, ON, Canada), and Elizabeth S. Olson (OTO/HNS & Biomed. Eng., Columbia Univ., New York, NY)

The classic picture of middle ear (ME) transmission has the tympanic membrane (TM) as a piston and the ME space as a vacuum. However, the TM moves in a complex wavy pattern and modern theories link this behavior to the ME's broadband transmission [e.g., Fay *et al*, Proc. Natl. Acad. Sci. U.S.A. (2006), Parent and Allen, J. Acoust. Soc. Am. (2007)]. Furthermore, Rabbitt [J. Acoust. Soc. Am. (1990)] predicted that the TM radiated considerable sound pressure into the ME space that could in return affect TM motion. This study explores these ideas with a simple model, using a tube terminated with a plastic membrane to model the EC and TM. We measured membrane motion via laser interferometry, and pressure (on both sides) via micro-sensors that allowed positioning very close to the membrane (10 micrometers) without disturbance. We made a finite element model of the system to complement the experimental results. The theoretical results show the interaction of acoustic and mechanical resonances, and both theoretical and experimental results show strong transmission of sound through the membrane at some of its resonant frequencies.

1:40

4pEAa3. Pole-zero modeling of the middle ear based on acoustic reflectance measurements. Sarah Robinson and Jont Allen (Elec. Eng., Univ. of Illinois at Urbana-Champaign, 209 N Coler Ave, APT 2, Urbana, IL 61801, srrobin2@illinois.edu)

Fitting poles and zeros to complex acoustic reflectance (CAR) data using a "rational approximation method" [Gustavsen and Semlyen (1999)] allows for a precise parameterization of complex real-ear measurements. CAR is measured using a foam-tipped probe sealed in the ear canal, containing a microphone and receiver (i.e., MEPA3 system, Mimosa Acoustics). From the complex pressure response to a broadband stimulus, the acoustic impedance and reflectance of the middle ear can be calculated as functions of frequency. The goal of this work is to establish a quantitative connection between the fitted pole-zero locations and underlying physical properties of the CAR and impedance of the middle ear. It was found that (1) the contribution of the ear canal may be approximated as the lossless all-pass component of the factored reflectance fit, (2) individual CAR magnitude variations for normal middle ears in the 1 to 4 kHz range give rise to closely placed pole-zero pairs, and (3) the locations of the poles and zeros in the s-plane may differ between normal and pathological middle ears. Pole-zero fitting allows for concise characterization of individual CAR measurements, providing a foundation for modeling individual and pathological variations of middle ears.

2:00

4pEAa4. Mitigation of excessive acoustic compliance and trapped volume insertion gain in ear-sealing listening devices: Toward a safe, full-frequency-response hearing aid. Stephen D. Ambrose and Samuel P. Gido (Asius Technologies LLC, 1257 Whitehall Dr., Longmont, CO 80504, stephen.ambrose@asiustechnologies.com)

When a sound producing device is sealed in the ear canal, acoustical compliances resulting from pressurization of the trapped volume lead to dramatic boosts in SPL, up to 60 dB, especially at low frequencies. This has been found to result in listener fatigue, and to trigger the acoustic (stapedius) reflex, as well as producing temporary threshold shift. Repeated exposure can cause temporary threshold shift to become permanent. Hearing aids avoid this problem by suppressing frequencies below about 300 Hz, where the effect is most pronounced. Other devices, such as ear buds and professional in-ear monitors, offer wider frequency response and thus expose listeners to potentially dangerous sound pressures. The acoustical compliance and trapped volume insertion gain is measured for ear buds and hearing aids by comparing SPL, measured in the ear canal, for sealed and unsealed conditions. New ear sealing technology is demonstrated that allows release of the excess acoustical compliance and thus mitigates the trapped volume insertion gain: (1) a vent covered with a flexible membrane, and (2) an inflatable bubble seal. This novel technology has allowed the creation of a hybrid device with hearing aid functionality that also has the broad frequency response of professional in-ear monitors.

2:20

4pEAa5. A comparison of methods for measuring the acoustic input impedance of ear canals for hearing aid applications. Tobias Sankowsky-Rothe, Simon Köhler, Matthias Blau (Institut für Hörtechnik und Audiologie, Jade Hochschule WOE, Ofener Straße 16-19, Oldenburg, Niedersachsen 26121, Germany, Tobias.Sankowsky@jade-hs.de), and Alfred Stirnemann (Adv. Products, Phonak AG, Stäfa, Switzerland)

In hearing aid fitting the sound pressure at the ear drum is a reference quantity, since all *real ear* characteristic values refer to it. Typically, the sound pressure at the ear drum is estimated by a model of an average ear canal (e.g., a coupler). Such a model cannot account for inter-individual differences. Alternatively, there are methods to predict the acoustics of the individual ear canal. Some of these methods make use of the acoustic input impedance of the ear canal. In general, the accuracy of the measured impedance depends on the effort that will be made. Therefore, different methods of impedance measurements were investigated concerning accuracy and effort. The methods differ in the number of calibration measurements (and calibration parameters). They were compared on the basis of impedance measurements on different model ear canals. Measurements were done with an impedance probe consisting of a typical hearing aid receiver and a hearing aid microphone. The measurements were compared to measurements with a reference impedance probe and method. As a result, it was observed that with a single calibration measurement the maximum absolute error of the transfer impedance was smaller than 3 dB up to 8 kHz.

Contributed Paper

2:40

4pEAa6. A comparison of methods for estimating individual real-ear-to-coupler-differences in hearing aid fitting. Simon Köhler, Tobias Sankowsky-Rothe, Matthias Blau (Institut für Hörtechnik und Audiologie, Jade Hochschule WOE, Ofener Str. 16/19, Oldenburg 26121, Germany, simon.koehler@jade-hs.de), and Alfred Stirnemann (Adv. Products, Phonak AG, Stäfa, Switzerland)

The sound pressure at the ear drum is *the* reference quantity for almost all applications of sound delivery to the ear, especially in hearing aid fitting. Since hearing aids are typically calibrated using the so called 2cc-coupler, the link to the individual sound pressure at the ear drum is given by the *real-ear-to-coupler-difference* (RECD). Nowadays, averaged RECDs are used for hearing aid fitting, which do not account for inter-individual differences in ear canal acoustics. As a consequence, resulting coupling errors may reach 15 dB for frequencies up to 10 kHz. Alternatively, there are methods for estimating individual RECDs, based on acoustic impedance measurements at the inner face of the ear mold. These methods differ in effort (e.g., the complexity of the ear canal model and fitting algorithm) and accuracy. By using an integrated ear canal microphone, individual RECD estimation could be feasible in future hearing aid fitting. In this research, six different methods to predict individual RECDs were compared using simulations as well as real ear measurements with open and closed ear molds. As a result, it appeared that relatively simple cylindrical and conical ear canal models give the best compromise between effort and accuracy.

Invited Paper

3:00

4pEAa7. Using inter-individual standard deviation of hearing thresholds as a criterion to compare methods aimed at quantifying the acoustic input to the human auditory system in occluded ear scenarios. Matthias Blau, Tobias Sankowsky-Rothe, Simon Köhler (Institut für Hörtechnik und Audiologie, Jade Hochschule Wilhelmshaven/Oldenburg/Elsfleth, Ofener Str. 16/19, Oldenburg D-26121, Germany, matthias.blau@jade-hs.de), and Jan-Henning Schmidt (Physikalisch-Technische Bundesanstalt, Braunschweig, Germany)

Occluded ear scenarios are found in many applications, e.g., hearing aids or insert ear phones. Unfortunately, the correct quantification of the acoustic input delivered to the auditory system in such a scenario is complicated by the individual character of our outer ear anatomy. For instance, one can easily observe inter-individual differences in ear drum pressure level of up to 30 dB at 10 kHz with one and the same sound source. We may thus ask: (1) Is the sensitivity of our auditory system at threshold adapted to our outer ear anatomy? and (2) what is the best method to quantify the acoustic input? We propose to use the inter-individual standard deviation of hearing thresholds as a means to answer these questions: The quantity that is best suited to describe the input to the auditory system should result in the lowest inter-individual standard deviation of thresholds. Preliminary results based on tests with custom ear shells and with foam ear plugs show that up to 6 kHz, there are no significant differences between the methods tested, whereas in the 6–9 kHz frequency range, individual estimates of the sound pressure at the ear drum yield a significantly lower inter-individual standard deviation than, e.g., the ear simulator pressure.

Contributed Papers

3:20

4pEAa8. Acoustics of enclosed spaces: The differences when the dimensions are millimeters vs. tens of meters. Martin Kuster (Sci. & Technol., Phonak AG, Laubisrütistr. 28, Stäfa 8712, Switzerland, kuster_martin@hotmail.com)

The dimensions of the ear canal are at least 3 orders of magnitude smaller than those typically encountered in room acoustics but at the same time the range of wavelengths for audio applications is identical. This results in a disparity not only in length scale but also a disparity in time scale. The influence of these disparities on well-known room acoustics parameters or features such as the reflection density, the direct-to-reverberant ratio, the critical distance, or transfer function nulls is reviewed and highlighted. The nature of the two substantially different sound fields is also important for active sound control. Consequently, the respective relevance of total absorption as well as values of source and sink impedance are also compared.

3:40

4pEAa9. Effect of the middle ear cavity on the response of the human auditory system. Antonio Garcia-Gonzalez and Antonio Gonzalez-Herrera (Civil Eng., Univ. of Malaga, AVDA. SALVADOR ALLENDE, 322, MALAGA, MALAGA 29017, Spain, AGH@UMA.ES)

The effect of the acoustic cavities on the response of the auditory system has been usually focused on the influence of the external ear canal (EEC). The presence of the middle ear cavity (MEC) has been ignored. Experimental difficulties to obtain information inside this cavity without altering the whole system make difficult its study. In order to explore the influence of this cavity, a numerical study is made. This is made by means of a complete finite element (FE) model including the tympanic membrane, ossicular chain, and acoustic cavities. Different FE models are used to analyze the influence of each component. By means of different calculations removing these components from the model, their relative effects can be distinguished. At low frequencies (below 2 kHz) the influence of the MEC is negligible. Piston-like motion is dominant. Nevertheless, at higher frequencies a new resonant peak appears at a frequency of 4 kHz. This is due to the presence of the MEC. It combine with the pressure gain due to the ear canal (at a frequency of 3 kHz) increasing the response of the system in terms of Umbo velocity. This effect is observed in different published experimental results.

4:00

4pEAa10. Estimation of ideal open-cavity middle-ear responses from responses with partial cavity opening. Nima Maftoon (BioMedical Eng., McGill Univ., 3775, rue University, Montréal, QC H3A 2B4, Canada, nima.maftoon@mail.mcgill.ca), W. Robert J. Funnell (BioMedical Eng. and Otolaryngol. – Head & Neck Surgery, McGill Univ., Montréal, QC, Canada), and Sam J. Daniel (Paediatric Surgery and Otolaryngol. – Head & Neck Surgery, McGill Univ., Montréal, QC, Canada)

An important step in developing mathematical models of the middle ear is the validation of simplified models without the middle-ear air cavity. However, open-cavity experimental results are often collected with only a partial opening of the middle-ear cavity, due to experimental limitations. The partial opening introduces an anti-resonance that obscures features of the frequency response in its neighborhood. In this study, we suggest a numerical method for estimating ideal open-cavity responses from experimental results with partial openings. We fit rational-fraction polynomials to portions of the response in order to parametrically identify the transfer function associated with the anti-resonance. The ideal open-cavity response is then estimated by dividing the experimentally measured frequency response by the identified anti-resonance transfer function. The method has been validated against synthesized transfer functions with features similar to those caused by partial opening of the cavity and against responses calculated using models of the middle ear with a partially open cavity.

4:20

4pEAa11. Finite-element modeling of the newborn ear canal and middle ear. Hamid Motallebzadeh, Brian Gariepy, Nima Maftoon (BioMedical Eng. Dept., McGill Univ., 3775, rue University, Rm. 303, Montreal, QC H3A 2B4, Canada, hamid.motallebzadeh@mail.mcgill.ca), W. Robert J. Funnell (BioMedical Eng. and Otolaryngol. – Head & Neck Surgery, McGill Univ., Montreal, QC, Canada), and Sam J. Daniel (Paediatric Surgery and Otolaryngol. – Head & Neck Surgery, McGill Univ., Montreal, QC, Canada)

Available hearing-screening procedures cannot distinguish clearly between conductive and sensorineural hearing loss in newborns, and the results of available diagnostic tests in very young infants are difficult to interpret. Admittance measurements can help to detect conductive losses but do not provide reliable results for newborns, where the ear is anatomically different from the adult ear. Finite-element models of the newborn ear canal and middle ear were developed and their responses were studied for frequencies up to 2000 Hz. Material properties were taken from previous measurements and estimates, and the sensitivities of the models to these different parameters were examined. The simulation results were validated through comparison with previous experimental measurements. Simulations indicate that at frequencies up to 250 Hz the admittance of the canal wall is comparable to that of the middle ear in the newborn. Above 250 Hz, the canal-wall admittance remains almost constant but for the middle ear there is a clearly defined resonance peak, which produces an admittance much larger than that of the canal wall. These results suggest that admittance measurements in the vicinity of the middle-ear resonance frequency can provide clinically useful information about the newborn middle ear.

4:40

4pEAa12. Harmonic hydromechanical movement. Santos Tieso, Francisco Messina, Lucas V. Fantini, Nicolás Casco Richiedei, Nahuel Cacavelos, Ignacio Talento, Nicolás Vallese, Rodrigo Fernández, and Adrián Saavedra (Ciencia y Tecnología, Universidad Nacional de Tres de Febrero, zapiola, Mendoza, Capital federal 1428, Argentina, francisco.messina@gmail.com)

The current theoretical physics tools used to describe the acoustic phenomena that occur outside the ear, such as the specific impedance, the acoustic impedance and the mechanical impedance are not applicable to describe the cochlear mechanics. For this reason, this study uses the hydro-mechanical impedance concept. The latter is only applicable to a harmonichydro-mechanical systems, which consist of a rigid recipient, filled with liquid and two elastic windows that relate

the system with a sound environment, considering that the distance between them should be much smaller than the sound wavelength. This system could be considered as the most primitive model inner ear to build. The movement of the contained fluid in this system has particular characteristics that differentiate it from the wave motion and from a simple mass-spring-dumping vibration system. In order to demonstrate the existence of the harmonic hydro-mechanical movement, was modeled and built an equivalent harmonic electrical system, which results corresponded with the ones from the theoretical mathematical model.

5:00

4pEAa13. Secondary path variation in human listeners and its effect on an active noise cancellation system. Jinjun Xiao, Buye Xu, and Tao Zhang (Starkey Hearing Technologies, 6600 Washington Ave. S., Eden Prairie, MN 55344, jinjun_xiao@starkey.com)

Active noise cancellation (ANC) has been applied to cancel the penetrated ambient noise in the ear canal for hearing impaired listeners [Zhang *et al.* (2012)]. The performance of the proposed ANC system depends on the characteristics of the secondary path (SP). In this study, we developed an in-ear ANC system where the error microphone and the miniature loudspeaker were both placed in the ear canal. In such a case, the SP response depends on the error microphone response, the loudspeaker response, and how the microphone, the loudspeaker, and the ear canal are acoustically coupled. A robust method was proposed to measure the SP response of the proposed ANC system under various coupling conditions using human listeners. Variations of the measured SP responses were analyzed both within each individual and across different individuals. The effect of the SP variations on the performance of the proposed ANC system was evaluated. The implications for improving the proposed ANC system will be discussed.

THURSDAY AFTERNOON, 6 JUNE 2013 — 512BF, 1:00 P.M. TO 5:20 P.M.

Session 4pEAb

Engineering Acoustics: Fields and Devices

Daniel Armstrong, Chair

Mech. Eng., McGill Univ., MacDonald Eng. Bldg., Rm. 364, 817 Sherbrooke St. West, Montreal, QC H3A 0C3, Canada

Contributed Papers

1:00

4pEAb1. Collocation analysis of junction conditions for waveguides at high-frequencies. Jerry H. Ginsberg (School of Mech. Eng., Georgia Inst. of Technol., 5661 Woodsong Dr., Dunwoody, GA 30338-2854, jerry.ginsberg@me.gatech.edu)

Analyses of high-frequency transmission and scattering at junctions of waveguide segments having different cross-sectional area must account for the generation, reflection, and transmission of higher order modes. The usual analysis relies on modal orthogonality properties, which are awkward to implement if a junction does not have a regular geometrical configuration. An alternative formulation uses a collocation procedure, in which continuity and boundary conditions at the interface are enforced exactly at a set of discrete points. The viability of that approach was examined recently [J. Acoust. Soc. Am. **132**, 1955 (2012)], but time restrictions only permitted assessment of the technique for a waveguide whose end is excited by a discontinuous velocity distribution. It was found that the convergence and computational requirements of the collocation method are essentially the same as those of an analysis that uses modal orthogonality. The present work extends the formulation to address situations where waveguide segments having different cross-sections are joined. A least squares solution of the junction equations for the modal coefficients is described and illustrated by an example.

1:20

4pEAb2. Inspectability of interfaces between composite and metallic layers using ultrasonic interface waves. Michael D. Gardner (Grad. Program in Acoust., The Penn State Univ., 225 Strouse Ave., State College, PA 16803, gardnerfrance@gmail.com), Joseph L. Rose (Dept. of Eng. Sci. and Mech., The Penn State Univ., University Park, PA), Kevin L. Koudela, and Clark A. Moose (Appl. Res. Lab., The Penn State Univ., University Park, PA)

The interface between an anisotropic composite material and a metallic material is inspected non-destructively for disbonded regions using ultrasonic guided waves. The material properties of the composite and metal have been tailored to demonstrate their effect on inspectability. The material properties have been designed to be either favorable or unfavorable to the existence of propagating Stoneley waves. Stoneley waves can exist because the layer thicknesses are large enough compared to the wavelength to be considered half-spaces. The existence of Stoneley waves between generally anisotropic materials depends on the elastic constants and densities in a complicated way. The range of material properties that allow Stoneley waves is small; however, when the vertically polarized shear wave speeds are similar in the two materials, the existence of Stoneley waves is generally possible. If the conditions do not strictly allow Stoneley waves, other interface waves can still exist such as leaky waves. Disbonds are inserted into the materials before bonding and are inspected using interface waves. Sensitivity to disbonds is determined and thus inspectability is demonstrated for cases that are favorable and unfavorable to Stoneley waves. Both numerical and physical experimental results are shown.

1:40

4pEAb3. Case studies of casing inspection with multi-functional ultrasonic imaging logging tool. Zhifeng Sun, Honghai Chen, and Xien Liu (WellTech, China Oilfield Services Ltd., G.P.O. Box 232, Beijing 101149, China, sunzf@cosl.com.cn)

The multi-functional ultrasonic imaging logging tool can provide casing inspection and cement bonding evaluation by using ultrasonic pulse echo technique. The casing's internal conditions can be inspected from echo amplitude and transit time curves. The casing thickness and the cement impedance can be calculated from resonance frequency and resonance decay. The paper describes three well cases histories of casing inspection in detail with this instrument. The first case reports the evaluation of perforation interval in the cased hole; hole enlargement of perforation interval and perforations can be seen from echo amplitude and casing internal radius imaging curves. In the second case, we cannot distinguish corrosion or deposits in the casing surface from the echo amplitude and transit time curves, but casing thickness imaging curve characterize some slight corrosion. Finally, the last case history describes evaluation of casing mechanical wear in a deviated well; casing thickness imaging curve shows mechanical wear is caused by logging

instrument in low side well. All cases shows that the multi-functional ultrasonic imaging logging tool can provide both quantitative and qualitative evaluation and diagnosis of casing problems.

2:00

4pEAb4. Enhancing noise control in a cavity using Helmholtz resonators. Ch. Surya Narayana V. Reddi and Chandramouli Padmanabhan (Mech. Eng., Indian Inst. of Technol. Madras, 406 Machine Design Section, Chennai, Tamilnadu 600036, India, mouli@iitm.ac.in)

In this paper, it is shown that there is a decrease in sound levels, not only when a resonator is tuned exactly to a cavity mode, but also when it is tuned to a frequency slightly different from the natural frequency of the chosen cavity mode. The finite element (FE) method is used for numerical modeling of the coupled Helmholtz resonator-cavity system. To validate the FE model prediction, a boundary element (BE) analysis is performed by specifying an impedance boundary condition on the element where the resonator is mounted. This impedance has been calculated, from a BE model of the resonator alone, using a plane wave excitation, over a range of frequencies around the cavity mode of interest. Numerical experiments have been performed in a cavity with two close modes (155 and 173 Hz) and it is observed that when a resonator is tuned to a frequency slightly lower than the first cavity mode, the performance of the resonator is much better than when it is tuned exactly to the first cavity mode or tuned to a slightly higher frequency. A detailed parametric study has been carried out and guidelines for tuning resonators for superior noise control is proposed.

2:20

4pEAb5. Acoustical impedance characterization of liners using a Bayesian approach. Yorick Buot de l'Epine, Jean-Daniel Chazot, and Jean-Michel Ville (CNRS UMR6253 Roberval, Université Technologique de Compiègne, Centre de Recherche Royalllieu, BP20529, Compiègne 60205, France, ybuotdel@utc.fr)

Acoustic liners composed of perforated plates and honeycomb layers are used in several applications. These liners are used for example in aerospace as acoustic treatments for aircraft nacelles. To describe their behavior, empirical models or standing wave tube experiments can be employed. However, the resulting impedance is not always accurate to describe the real behavior of the liner when submitted to several planes waves at various angles (higher order modes in the duct), or when submitted to a grazing flow. In this paper, an inverse method based on a Bayesian approach is presented in order to characterize acoustic liners in real conditions. An analytical solution and experimental data are used to calculate the likelihood function of the estimated impedance. The posterior probability density function can then be obtained by adding the prior information. Finally, an evolutionary Markov Chain Monte Carlo method (eMCMC) is implemented to explore the probability density space. This inverse method is first validated on simulated data. Then experimental data are used.

2:40

4pEAb6. Cylindrical cyclic acoustic imaging with a Bayesian approach for cyclostationary sources reconstruction. Sebastien Personne (Laboratoire Roberval CNRS, UTC, BP 20529 cedex, Compiegne 60205, France, sebastien.personne@hotmail.fr), Jerome Antoni (LVA, INSA, Lyon, France), and Jean Daniel Chazot (Laboratoire Roberval CNRS, UTC, Compiegne, France)

Standard acoustic imaging techniques, such as beamforming or near acoustical holography, are now widely used in engineering contexts. However, large arrays of microphones are sometimes required to have a good resolution. Besides new challenges arise, particularly in the field of non stationary sources, which need to be identified and solved. Cyclostationary sound sources, a specific kind of non stationary signals, are characterized by statistical properties evolving periodically in time. In practice, the first-order statistical properties contain some periodic components while the second orders may be random with a periodic flow of energy. The present work tackles the acoustic imaging of cyclostationary sources with a scanning microphone, i.e., without any array. Cylindrical surfaces, adapted to standard rotating machines, are considered. The reconstruction difficulty of acoustic sources from discrete measurements is addressed here thanks to the cyclostationary properties. A cyclic sound field is hence extracted from the discrete measurements. Finally, a Bayesian formulation, gathering both physical and probabilistic information on this inverse problem, is used to back propagate the sound over the radiating surface.

3:00

4pEAb7. Pressure mapping system based on guided waves reflection. Nicolas Quaegebeur, Patrice Masson (GAUS - Dept. Mech. Eng., Université de Sherbrooke, 2500 Blvd Universite, Sherbrooke, QC J1K2R1, Canada, nicolas.quaegebeur@usherbrooke.ca), Nicolas Beaudet, and Philippe Sarret (Dept of Physiol. and Biophys., Université de Sherbrooke, Sherbrooke, QC, Canada)

In this paper, guided wave interaction is used to develop a pressure mapping system for medical and touch-screen applications. The principle is based on interaction of guided waves in the presence of an added local mass and in the presence of a local pressure. For this purpose, piezoceramics are used for injecting guided waves into a thin structure and to measure the reflected waves due to the presence of an added mass or pressure. SHM imaging algorithms, based on time-of-flight (EUSR) or correlation (Excitelet), are implemented in order to obtain cartography of the reflections and deduce the presence, localization, and intensity of local contact spots. Analytical and numerical models are first derived to assess the critical parameters in order to maximize the reflection of guided waves (first order modes A0 and S0). It is shown that the sensitivity of the guided waves with respect to an added mass and pressure is highly related to the Young's modulus of the host structure. Validation on a 0.5 mm thick plane aluminum plate prototype is addressed using 4 sensor/actuator pairs. It is observed that imaging of single pressure spot and multiple or extended pressure spots can be achieved using S0 mode around 300 kHz with a resolution of 0.5 mm.

3:20

4pEAb8. Microbubble histogram reconstruction by nonlinear frequency mixing. Matthieu Cavaro (CEA, DEN, Lab. of Instrumentation and Technolog. Test, Cadarache, Bâtiment 202, Saint Paul lez Durance 13108, France, matthieu.cavaro@cea.fr) and Cédric Payan (Aix-Marseille Université, Aix en Provence, France)

In the 4th generation sodium fast nuclear reactors (SFR), different phenomena can lead to gaseous microbubbles presence in the primary liquid sodium pool. This paper investigates the ability of nonlinear acoustics techniques to characterize these microbubbles presence. The goal is here to determine the void fraction (volume fraction of free gas) and the histogram of bubbles radius. Different acoustic techniques are currently developed at CEA. Among others, the nonlinear mixing of two frequencies [V. L. Newhouse and P. M. Shankar, J. Acoust. Soc. Am. **75**(5), 1473–1477 (1984)] is under study. Based on the nonlinear behavior of bubble resonance, this technique allows determining the radius histogram of a bubble cloud. Two different mixing techniques are here presented: the mixing of two high frequencies and the mixing of a high and a low frequency. The first step is an air-water experimental set-up. Microbubbles clouds are generated with a like dissolved air flotation process and an optical device gives us reference measures. Generated bubbles have radii in the range of several microns to several tens of microns. The developed experimental procedure allows us to determine the bubble size's histograms with accuracy never reported yet.

3:40

4pEAb9. Effects of a boundary layer trip in the prediction of noise from flow past tandem cylinders. Daniel Armstrong and Luc Mongeau (Mech. Eng., McGill Univ., MacDonald Eng. Bldg., Rm. 364, 817 Sherbrooke St. West, Montreal, QC H3A 0C3, Canada, daniel.armstrong2@mail.mcgill.ca)

The use of boundary layer trips in wind tunnel experiments forces transition to fully turbulent flow, which affects the resulting acoustic signature. Boundary layer trips are often too small to be modeled geometrically in computational simulations, making it difficult to consider their influence. The goal of this study was to determine the best method for representing the effects of a boundary layer trip in the benchmark aeroacoustic problem of flow over tandem cylinders in a wind tunnel. This case is pertinent to the study of aircraft landing gear noise because periodic vortex shedding results in pronounced acoustic tones in addition to broadband noise from turbulent

wake interactions. A hybrid lattice-Boltzmann Method/Ffowcs-Williams Hawkings technique was used to predict the transient flow field and the resulting noise. This study compared four methods for simulating the trip's effects: (1) applying a surface roughness on the upstream cylinder; (2) reducing the viscosity of the fluid; (3) forcing velocity perturbations in the incoming free stream; and (4) applying a ridge to the upstream cylinder that is one volume element thick. Preliminary results have shown that reducing the fluid's viscosity is an effective way to reproduce the highly turbulent flow patterns and the acoustic signature.

4:00

4pEAb10. Characteristics of ultrasonic complex vibration for hole machining in brittle materials: Comparison of longitudinal and complex vibration sources. Takuya Asami and Hikaru Miura (Nihon Univ., 1-8-14 Kanda-Surugadai, Chiyoda-ku, 325 Rm., Tokyo 101-8308, Japan, asami.takuya@gmail.com)

Ceramic materials have the advantage of abrasion resistance, heat resistance, and corrosion resistance compared with metal materials. The combination of ultrasonic vibration and polishing slurry has been shown to be an effective method for machining holes in brittle materials. However, conventional ultrasonic methods use only longitudinal vibration. Complex vibration sources with diagonal slits have been applied to ultrasonic motors and ultrasonic welding; in contrast, few studies have been conducted on ultrasonic machining using complex vibration and polishing slurry. Removal rates and machining accuracy are expected to be improved by using ultrasonic complex (longitudinal-torsional) vibration. Therefore, we have developed a new method using polishing slurry together with ultrasonic longitudinal and torsional vibration sources with diagonal slits for hole machining of brittle materials. Torsional vibration is considered to improve the processing of the hole side of ceramic materials such that the polishing slurry can circulate more easily. We assume improvement of removal rate and machining accuracy for that reason. In experiments, soda-lime glass is used as the processing material in ultrasonic complex vibration or ultrasonic longitudinal vibration, and machining time is measured to assess the hole machining characteristics.

4:20

4pEAb11. Sound generation using photoacoustic effect. Kaoru Yamabe, Yasuhiro Oikawa, and Yoshio Yamasaki (Intermedia Art and Sci., Waseda Univ., 59-407-2, 3-4-1 Okubo, Shinjuku-ku, Tokyo 169-8555, Japan, shikisokuzeku@fuji.waseda.jp)

It is highly important to generate sound sources in mid air for several applications such as virtual realty and rigorous acoustic measurement. One possible solution for generating sound sources in mid air is photoacoustic effect that generates sounds from the alternate-current component of air expansion due to the heat generated by light absorption of materials when the materials are radiated light modulated with acoustic signal. Our previous research confirmed that audible sound could be generated by radiating light modulated with acoustic signal to charcoal, which has high absorptive power. Therefore, it is possible to generate point sound source in mid air by applying this method to molecule of gases in mid air. However, gases are difficult to absorb light since gases have low density of molecule. Thus in this paper, as the preliminary step applying photoacoustic effect to gas molecule, it is discussed to generate audible sound by radiating light modulated with acoustic signal to liquid phase of H_2O and solid phase of CO_2. These molecules of greenhouse gases can absorb infrared light that is safer than ultraviolet light that is absorbed by monatomic molecules such as N_2 and O_2.

4:40

4pEAb12. Acoustic compressor coupled with fluidic diodes. Sonu K. Thomas and T. M. Muruganadam (Dept. of Aerosp. Eng., Indian Inst. of Technol., Madras, Rarefied Gas-dynamics Lab, IIT, Madras, Chennai, Tamil Nadu 600036, India, thomas.sonu91@gmail.com)

Performance of an acoustic compressor coupled with a fluidic diode is studied. The acoustic compressor works on the idea of Resonant Macrosonic Synthesis (RMS) technology demonstrated by Lawrenson *et al*. The main idea is to replace non-return valves by no moving part fluidic device. Fluidic diode rectifies an oscillatory flow analogous to rectifying an electric AC to obtain DC output. Synthetic jets analogous to AC current falls in the category of jet-driven acoustic streaming, which has a zero mean mass flux. The synthetic jet can be combined with no-moving part fluidic device to generate Hybrid Synthetic Jets with non zero mean mass flux. Non-zero mass flow rate was achieved by coupling fluidic diode and the resonator. In the present study, better rectification is achieved by having number of fluidic diodes in series. Experiments were done for the case of 3 and 4 diodes in series. In the case of three diodes in series the mass flow was 4.6 l/min and in case of four diodes it was 4.9 l/min. Full paper will present the pressure and mass flow measurements for 1 to 5 diodes in series. Thus, RMS cavities with fluid diodes can work as a pump.

5:00

4pEAb13. Plane wave echo particle image velocimetry. Samuel Rodriguez, Xavier Jacob, and Vincent Gibiat (Université Paul Sabatier Toulouse III, PHASE Lab., Bât. 3R1-B2, 118, route de Narbonne, Toulouse 31062, France, vincent.gibiat@univ-tlse3.fr)

This paper deals with the application of topological imaging to ultrasonic echo-particle image velocimetry (Echo-PIV). Echo-PIV is a recent alternative to optical PIV for measuring the instantaneous velocity field of a fluid flow previously seeded with small particles. It consists in imaging the flow with a ultrasonic array at a high frame rate. Topological imaging is a method that benefits from the refocusing properties of the time-reversal principle in a systematic way, so that a single plane wave illumination of the medium leads to a fine resolution. Multiple insonifications are then possible at very high speed allowing not only static images of the medium but successive images of a moving medium. Experimental results are presented for a fluid seeded with stone powder. Two cases are studied: a vortex flow and the propagation of water surface waves.

Session 4pMU

Musical Acoustics: Measurements, Modeling, and Simulations of Brass Instruments

James W. Beauchamp, Cochair
School of Music and Elec. & Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824

Wilfried Kausel, Cochair
Inst. of Music Acoust., Univ. of Music and Performing Arts, Anton-von-Webern-Platz 1, Vienna 1030, Austria

Thomas Moore, Cochair
Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789

Invited Papers

1:20

4pMU1. Do trumpeters tune resonances of their vocal tract? Jer-Ming Chen, John Smith, and Joe Wolfe (The Univ. of New South Wales, School of Phys. UNSW, Sydney, NSW 2052, Australia, jerming@unsw.edu.au)

In most wind instruments, the acoustic output is generated by airflow through a non-linear valve, whose sounding frequency is largely determined by resonances in the bore of the instrument (an acoustic duct downstream of the valve) and mechanical properties of the non-linear valve that converts DC to AC power. The player's vocal tract (a second duct, upstream) also has acoustic resonances, which—in particular cases—play a significant role in performance technique. For example, when executing advanced techniques (e.g., pitch-bending, altissimo playing) on the clarinet and saxophone, we showed that expert control of vocal tract resonances is essential for performance [Chen *et al*., *Science*, **319**, 726 (2008)]. To understand how such a tract-valve-bore system might interact during trumpet performance, we measured the acoustic impedance spectrum in seven trumpeters' mouths as they played normal notes, high-register notes and while pitch-bending below and above the normal note. Unlike the behavior seen in saxophonists and clarinetists, none of the trumpeters studied showed any systematic adjustment of their vocal tract resonances to the notes played. The much greater control that trumpeters have over the natural frequency of the vibrating valve may explain the difference with clarinetists and saxophonists.

1:40

4pMU2. How can we deduce playing frequencies from measured resonance frequencies for trumpets? René E. Caussé, Pauline Eveno (Instrumental Acoust., IRCAM, 1 place Igor Stravinsky, Paris 75004, France, Rene.Causse@ircam.fr), Joël Gilbert (LAUM, Universite du Maine, Le Mans, France), and Jean-François Petiot (IRCCYN, Ecole Centrale de Nantes, Nantes, France)

Lip-type valve ("striking outwards" type) is responsible for sound production for brass instruments. The operation of the valve is controlled by feedback from a passive resonator. The purpose of this study is to compare experimentally how far the resonance frequencies of instrument, taken from their input impedance (which does not involve the intervention of the player's lips) are able to give informations about the playing frequencies. A family of three trumpets made from a basic instrument for which the lead pipe will be slightly modified for each model, were considered for the experiment. Four expert musicians were asked to play the first five playable notes, for four different fingerings and for three nuances. This exercise was repeated three times. All these notes allow to make a quantitative assessment of the relation between the resonance frequencies and the playing frequencies, using in particular statistical methods. Several results will be presented: the influence of the player on the overall intonation, the effect of nuances on the pitch and the relation between small changes of geometry and playing frequencies. Functions made from the input impedance, such as the « sum function » proposed by Wogram, do not bring more relevant informations than the input impedance itself.

2:00

4pMU3. A trombone model emphasizing acoustic accuracy and playability. Tamara Smyth (Music, Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, trsmyth@ucsd.edu) and Frederick Scott (Computing Sci., Simon Fraser Univ., Vancouver, BC, Canada)

This work contributes a physical synthesis model of the trombone, a virtual musical instrument emphasizing quality sound production, and interactivity. The focus is on modeling and coupling four parts of the trombone: the instrument bore, the bell, the vibrating lips, and the mouthpiece. The model of the instrument is made parametrically flexible by using a combination of filter elements modeled either using known theory or, for elements not well described theoretically, from acoustic measurement. In particular, acoustic accuracy of the bell reflection and transmission is explored by comparing results obtained from measurement, to those obtained from a piecewise conical model. In addition, the playability and performance characteristics of the complete sounding model—when coupled to a mouthpiece and a configurable generalized reed model—is discussed with reference to expected acoustic behavior.

2:20

4pMU4. Influence of the bell profile of the trombone on sound reflection and radiation. D. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, d.m.campbell@ed.ac.uk), Arnold Myers (Edinburgh College of Art, Univ. of Edinburgh, Edinburgh, United Kingdom), and John Chick (School of Eng., Univ. of Edinburgh, Edinburgh, United Kingdom)

One of the most striking external features of a modern trombone is its wide and rapidly flaring bell. The bore profile of this final section of the instrument influences its musical behavior in a number of different ways, since it determines both the strength of the acoustical feedback from the instrument to the lips of the player and the nature of the radiated sound field. These effects have been explored in an experimental study in which a number of trombones have been progressively modified by the removal of annular sections of the bells. Measurements of input impedance, transfer function, and directivity of radiated sound are presented, and the implications for the timbre and playability of the instruments are discussed.

2:40

4pMU5. Brass instrument power efficiency and the relationship between input impedance and transfer function. Wilfred Kausel (Inst. of Musical Acoust., Univ. of Music ad Performing Arts, Institut f. Wiener Klangstil, Vienna, Austria, Kausel@mdw.ac.at), James W. Beauchamp (School of Music and Elec. & Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Sandra Carral (Inst. of Musical Acoust., Univ. of Music ad Performing Arts, Vienna, Austria)

From observation of graphs of brass input impedance magnitude and transfer function vs. frequency, it is obvious that there is a strong relationship between the two. Both exhibit a series of strong resonances extending from a low frequency limit to a cutoff frequency, which is inversely proportional to the instrument's bell radius ($f_c = c/(\pi a)$. However, the maxima of the impedance function correspond to the minima of the transfer function. As previously shown [Elliott *et al*., J. Acoust. Soc. Am. (1982)], the relationship can be seen through a formula for *efficiency* given by $power_{out}/power_{in}$. This formula leads to the squared transfer function being proportional to the efficiency times the real part of the reciprocal of the input impedance, divided by the real part of the radiation admittance. Curves for input impedance, transfer function, and efficiency have been measured, simulated, and compared for several brass instruments. For frequencies below cutoff, the efficiency has an approximate monotonically increasing relationship with frequency, where the log-log slope is dependent on internal losses.

3:00–3:20 Break

3:20

4pMU6. Comparing steady-state and transient phenomena in brass instruments. Peter L. Hoekje (Phys. and Astronomy, Baldwin Wallace Univ., 275 Eastland Rd., Berea, OH 44017, phoekje@bw.edu)

Input impedance and radiation patterns are well-known examples of important brass instrument characteristics measured under steady state conditions. Transient phenomena are less studied, but potentially as important to the player and listener. For example, the heights and harmonicity of the peaks of the instrument's input impedance affect its steady-state playing response, which the player might describe on a range from "stuffy" to "open" or "free-blowing." However, the player also wants an instrument that will facilitate clean and reliable attacks. The development of the instrument's pressure spectrum during the onset transient can serve as an additional diagnostic tool to reveal information about the instrument response under playing conditions. The contribution of the instrument body vibrations to the radiated sound field is small and generally imperceptible under steady-state conditions. However, the bandwidths of the body vibration resonances are generally much narrower than those of the air column resonances and accordingly their transient responses are much longer. This leads to a time signature that enhances their detection by the listener. A complete picture requires consideration of both the steady-state and transient phenomena.

3:40

4pMU7. Nonlinear wall vibration and wave steepening contributing to tonal metallicness and brassiness in a horn. Takayasu Ebihara (YAMAHA Corp., 203 Matsunokijima, Iwata, Shizuoka 438-0192, Japan, prawn_taka@yahoo.co.jp) and Shigeru Yoshikawa (Grad. School of Design, Kyushu Univ., Fukuoka, Japan)

It is well known that wave steepening and shock-wave formation due to nonlinear propagation through the bore are responsible for tonal *brassiness* of brass instruments. On the other hand, penetrating metallic tones are produced by hand-stopping the French horn. The present study demonstrates that the mechanism account for tonal *metallicness* of the French horn is nonlinear wall vibration of the bell. The measured waveforms of radiated pressure of the stopped tones indicate rapidly corrugating changes, which are not observed in brassy tones. Also, their spectra show much larger amplitudes of higher harmonics than those in normal mezzo-forte playing. The measurement of the wall vibration at the bell in hand stopping demonstrates similar characteristics. These results suggest that the bell wall vibration is responsible for the radiated tone color. Excitation experiments on the bell are carried out to elucidate the mechanism how the higher harmonic vibration is generated in hand stopping. They indicate that wall vibrations over 3 kHz are excited by the superharmonic generation derived from the geometrical nonlinearity of the bell. Moreover, for a direct support to our inference above, sound pressure of the stopped tone radiated when the horn bell is heavily damped will be examined.

4:00

4pMU8. Analysis of vibroacoustics of trombone bells thanks to an adaptation of the Miller experiment. Francois Gautier, Mathieu Secail, and Joel Gilbert (Laboratoire d'Acoustique de l'Université du Maine, Université du Maine, Avenue O. Messiaen, Le Mans 72000, France, francois.gautier@univ-lemans.fr)

The influence of wall vibrations on the sound produced by a wind instrument is an open question. If it is clear that the vibrations of bells vibrations can be felt and measured, the influence of these vibrations on the radiated sound is more difficult to bring to light, because the fluid-structure couplings involved are particularly weak except when coincidence effects occur. We propose to study the case of a trombone bell, which is large and thin, favoring the vibrations of large amplitudes and thus the vibroacoustic coupling between

the wall and the air column. For studying the light fluid-structure interaction in organ pipes, Miller [Science **29**(735), 161–171 (1909)] developed one century ago an experiment consisting in blowing an organ pipe surrounded by water. A water tank can be filled progressively in order to modify the mechanical modes of the system in a continuous manner. We propose an adaptation of this experiment to the case of a trombone bell. The acoustic impedance, the acoustical and mechanical responses of a trombone bell excited by a loudspeaker or a shaker are measured for different levels of water allowing an analysis of the vibroacoustic couplings.

Contributed Papers

4:20

4pMU9. Axial vibrations of brass wind instruments. Thomas Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu), Wilfried Kausel, Vasileios Chatziioannou (Inst. of Music Acoust., Univ. of Music and Performing Arts, Vienna, Austria), Nikki Etchenique, and Britta Gorman (Dept. of Phys., Rollins College, Winter Park, FL)

It has been proposed that axial vibrations of the bells of brass wind instruments can lead to audible effects in the sound [Kausel *et al*. (2010)]. Using both laser Doppler vibrometry and a novel implementation of electronic speckle pattern interferometry, we have demonstrated that these vibrations exist, and that the magnitude is of the order predicted. [Work supported in part by a grant from the National Science Foundation.]

4:40

4pMU10. The soft-source impedance of the lip-reed: Experimental measurements and computational simulation. Michael Newton, Reginald Harrison Harrison (Reid School of Music, Univ. of Edinburgh, Alison House, Nicolson Square, Edinburgh EH8 9DF, United Kingdom, michael.newton@ed.ac.uk), and Jonathan Kemp (Dept. Music, Univ. of St Andrews, St Andrews, United Kingdom)

Most theoretical descriptions of the brass instrument lip-reed consider the acoustical condition at the lips to be a closed, rigid termination, corresponding to a unitary reflectance. This assumption is carried through to many computational models as well. In reality, the protrusion of the player's lips into the mouthpiece causes a periodic shortening/extension of the acoustical tube downstream, an effect sometimes but not always incorporated into such models. Of interest here is the absorption properties of the lip termination, the so-called "soft source impedance." This provides a further modification to the boundary condition at the lips, since the soft, deformable nature of the lips are likely to cause some extra damping of the acoustic standing wave. Measurements are presented to demonstrate this damping effect using an artificial mouth. This is achieved through measurements of the lip reflectance from downstream of the lips, from where it is shown that the reflectance shows a dip at the peak absorbance frequency of the lips. The frequency of the absorbance is shown to vary as the lip parameters are changed. A simple computational model is described to account for the effect.

Session 4pNSa

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

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 D.C. degli Abruzzi, 24, Turin 10124, Italy

Chair's Introduction—12:55

Invited Papers

1:00

4pNSa1. Effects of background noise alternating between two levels at varying time intervals on human perception and performance. Andrew Hathaway and Lily M. Wang (Durham School of Architectural Eng. and Constr., Univ. of Nebraska – Lincoln, Peter Kiewit Inst., 4014 Burt St., Omaha, NE 68131, ahathaway@unomaha.edu)

Heating, ventilation, and air-conditioning (HVAC) systems commonly produce noise in the built environment, and the increased noise levels that these systems can produce have been shown to impact occupant comfort. However, relatively little is understood about how the fluctuations in HVAC noise over time can impact human perception and performance. This research aims to measure human responses under HVAC-like background noise that is alternating between two different levels (one low and one high) at varying time intervals. Twenty-seven participants were tested over four 30 min sessions during which they were subjected to broadband noise at room criteria ratings of RC-29(H) and RC-47(RV) that alternated at certain time intervals. The time intervals of variation tested were 2, 5, 8, and 10 min, and would remain the same during one 30 min test session. The results of an arithmetic test dealing with short-term memory and a subjective questionnaire are presented to show whether or not shorter time intervals of variation have different effects than longer ones. [Work supported by a NASA Nebraska Space Grant.]

1:20

4pNSa2. Background noise in Chinese schools – Student and teacher perceptions. Kenneth P. Roy (R&D, Armstrong World Industries, Innovation Ctr., 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrong.com) and Jerry Li (Regional Res. Ctr., Armstrong W.I., China Ltd., Shanghai, China)

A significant research study was conducted in 2011/2012 to evaluate the Acoustic Comfort in 10 primary/middle schools throughout China as a joint effort by the Green Campus workgroup (Tongji University, Shanghai), and Armstrong World Industries (Research and Development). In the fall of 2012, an entire grade school in Nanjing, China, was involved in a combination of renovated and new construction with a focus on the requirements of Chinese Standard GB50118 for acoustic design/performance of classrooms. These research programs all included both objective measurements of acoustic performance, and subjective perception by students and teachers of both speech clarity (architecture) and distractions and comprehension (noise). The noise aspects of these measurements and surveys will be addressed in this paper. The primary noise sources in these schools are mainly from outside the classroom itself and involve student activities in corridors and other classrooms, and adjacent transportation noise. Effectiveness of in-room acoustic treatments on these noise perceptions is also reviewed.

1:40

4pNSa3. Influence of classroom acoustics on the vocal behavior of teachers. Arianna Astolfi (Dept. of Energy, Politecnico di Torino, Corso Duca degli Abruzzi, 24, Turin 10124, Italy, arianna.astolfi@polito.it), Alessio Carullo, Alberto Vallan (Dept. of Electron. and Telecommunications, Politecnico di Torino, Turin, Italy), and Lorenzo Pavese (Dept. of Energy, Politecnico di Torino, Turin, Italy)

Erroneous vocal behavior of teachers and their changes in the voice production due to poor acoustics in classrooms, can be investigated through recently developed voice-monitoring devices. These devices are portable analyzers that use a miniature contact microphone glued to the jugular notch in order to sense the skin acceleration level due to the vibration of the vocal folds. They estimate the Sound Pressure Level (SPL) at a certain distance from the speaker's mouth, provided that a suitable calibration procedure is performed, the fundamental frequency and the time dose. Two different devices are compared in this work: the former is a commercial device, whose phonation sensor is a small accelerometer; the latter, recently developed by the authors, uses an electret condenser microphone to sense the skin acceleration level. SPL and fundamental frequency are estimated over 30 ms- and 50 ms-length frame and the results that refer to a sample of 40 primary school teachers and some university professors are analyzed. The length of the voice and pause frames is analyzed in order to detect the maximum of occurrence and accumulation in different conditions of noise and reverberation. A method for the detection and analysis of the emphatic speaking is also proposed.

2:00

4pNSa4. Effects of reverberation and noise on speech comprehension by native and non-native English-speaking listeners. Zhao Peng, Lily M. Wang, Siu-Kit Lau, and Adam M. Steinbach (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182, zpeng@unomaha.edu)

Previous studies have demonstrated the negative impact of adverse signal-to-noise-ratios on non-native English-speaking listeners' performance on speech recognition using recall tasks, as well as implied that comprehension skills were more impaired than recognition skills under reverberation and noise. The authors have themselves previously conducted a pilot study on three native and three non-native English-speaking listeners to examine the effects of reverberation and noise using speech comprehension tasks. Those results suggested that speech comprehension performance is worse under longer reverberation times (RT), and that a longer RT is more detrimental to speech comprehension by non-native listeners than native listeners. This paper reports on the refined full study, in which a larger number (up to 30) of each group was tested. Each participant was exposed to 15 acoustic conditions, created from combinations of five RTs (0.4 to 1.2 s) and three background noise levels (RC-30, 40, and 50). Speech comprehension performance under each condition was recorded. Confounders related to general speech comprehension abilities were screened for, including listening span, oral comprehension abilities, and English verbal skills. Results are presented and compared between native and non-native listeners. [Work supported by a UNL Durham School Seed Grant and the Paul S. Veneklasen Research Foundation.]

2:20

4pNSa5. Open plan office: The appropriate privacy and material metrics. Kenneth W. Good (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

Recent metrics for speech privacy may be misapplied to the open plan environment resulting in false and misleading conclusions. This paper will review the state of privacy and material metrics, discuss the application and perspective of "privacy" in the open plan. And present the real effects of these metrics in the open plan environment.

Contributed Papers

2:40

4pNSa6. The teachers perspective on noise in the classroom. Ana M. Jaramillo, Michael Ermann, and Patrick Miller (School of Architecture + Design, Virginia Tech, 424 C Harding Ave., Blacksburg, VA 24060, anaja@vt.edu)

A survey was sent to third grade teachers in Orange County, FL, to find out about their noise awareness and coping strategies. Results of the survey were also correlated to mechanical system type and achievement data. Preliminary analyses show very little awareness on mechanical noise by teachers but a good range of coping strategies when noise sources are present (mostly activity noise). The survey also helped to better understand the classroom environment. For example, most classrooms have a frequent use of computers or projectors and a few schools are still open-plan. These facts create new questions about noise in the classroom that need to be addressed in further studies.

3:00–3:20 Break

3:20

4pNSa7. Perception and evaluation of noise sources in open plan office. Marjorie Pierrette, Etienne Parizet (Laboratoire Vibrations Acoustique, INSA, 25 bis, av. J. Capelle, Villeurbanne F-69621, France, marjorie.pierrette@insa-lyon.fr), and Patrick Chevret (Laboratoire "Réduction du bruit au travail", INRS Centre de Lorraine, Vandoeuvre Les Nancy, France)

Open plan offices are now the most common form of workspaces organization. They can improve communication between workers while saving space. Their main drawbacks are the lack of intimacy for occupants and the increase of noise level. Noise is one of the most important annoyance factor as described by workers (see SBISB study, 2010). This paper describes a study aiming at a better knowledge of most annoying noise sources in an open plan offices. It consisted in interviews and questionnaires conducted in offices together with physical measurement. This provided some information about sources and tasks for which workers are mainly disturbed. The analysis of recorded answers allowed to evaluate the influence of this annoyance on job stress and health and emphasize the influence of environmental and individual factors in the assessment of noise annoyance.

3:40

4pNSa8. Work performance and mental workload in multiple talker environments. Ange Ebissou, Patrick Chevret (Institut National de Recherche et de Sécurité (INRS), Rue du Morvan, CS 60027, Vandoeuvre-les-Nancy 54519, France, ange.ebissou@inrs.fr), and Etienne Parizet (Laboratoire Vibration et Acoustique, INSA-Lyon, Villeurbanne Cédex, France)

The impairment of cognitive performance resulting from the presence of speech sounds is known to increase as the intelligibility of the speech signals is improved. For that reason, speech intelligibility measures are used to quantify the nuisance potential of an unattended voice. However, most of these indexes struggle with situations in which the level of the masking sound is fluctuating. This is the case in open-plan offices, where competing voices are involved. This paper relates a set of experiments in which subjects had to carry out a basic memory task in various noise settings. In addition to a target speech, the masking sounds were made up of speech and differed in temporal variability. The signal-to-noise ratios and the overall long-term spectra were kept constant. Disturbance was assessed both through objective measurements of performance and subjective reports of workload. The results highlight the importance of taking into account the temporal fluctuations of the overall ambient sound when trying to ascertain the influence of speech intelligibility on observed and perceived disturbance during the performing of a mental activity. Insights are provided, which could lead to the use of a speech intelligibility measure better equipped to deal with multi-sources environments.

4:00

4pNSa9. Planned versus achieved acoustical performance and user satisfaction for an office fit-out. Ryan Bessey (Acoust., Noise and Vib. Group, Golder Assoc. Ltd., 141 Adelaide St. West, Ste. 1220, Toronto, ON M5H 3L5, Canada, ryan_bessey@golder.com)

The purpose of the project was to move 500 office workers separated between two aging buildings into one new larger building. To this end, a team was formed to create an office fit-out design including several open plan offices, private offices, meeting rooms, boardrooms, a conference center, and a cafeteria to connect with the base-building infrastructure. Our role, as part of the design team, was to provide guidance such that the acoustical performance for new spaces was at least equivalent but preferably better than

their existing counterparts. Factors that were considered included acoustic comfort, background sound level, speech privacy, and speech intelligibility. This paper summarizes our approach to the project, some difficulties that were encountered during construction, measured acoustical performance in both the newly constructed and existing buildings, and the statistical change in user satisfaction before and after the move determined via surveys.

4:20

4pNSa10. Noise stress for patients in hospitals – A literature survey. Gert Notbohm and Silvester Siegmann (Inst. of Occupational Medicine, Univ. of Duesseldorf, Universitaetsstr. 1, Duesseldorf 40225, Germany, notbohm@uni-duesseldorf.de)

The growing number of publications on noise in hospitals reflects not only a rising interest in this theme during the last decades, but also an increasing noise exposure of the patients: the average SPL reported in literature between 1960 and 2005 has risen from 57 to 72 dBA in daytime and from 42 to 60 dBA at night. The hospitals in question differ substantially with regard to type of construction, technical equipment, and organizational issues. But especially for intensive care units (ICUs), the main sources of noise described in international research are similar: sounds from technical appliance such as alarms, noise caused by the staff talking or handling material, and communication systems such as overhead paging. With regard to patients in ICUs, sleep disturbances in terms of falling asleep and sleeping through are the greatest problem as assessed by questionnaires or by physiological measurements. They might have harmful effects on the outcome of the medical treatment influencing the duration of recovery and the need for sedative medication. Several intervention programmes for noise reduction are reported in literature combining a variety of methods such as acoustical insulation, sound level reduction with regard to equipment, and especially behavior modification of the staff.

4:40

4pNSa11. Vocal strain in UK teachers: An investigation into the acoustic causes and cures. Nick Durup, Bridget Shield, Stephen Dance (Dept. of Urban Eng., London South Bank Univ., 14 Lynton Dr., Ely cb6 1dq, United Kingdom, nicksenate@hotmail.com), and Rory Sullivan (Sharps Redmore Partnership Ltd., Ipswich, United Kingdom)

Recent surveys indicate that approximately 60% of UK teachers experience voice problems during their career. This costs £15M annually in teacher absence and can have a significant human cost for those involved. This study investigated the impact of classroom acoustics on teachers' voice levels to determine if acoustic modifications of classrooms could reduce the vocal load placed on teachers. Measurements of teachers' voice levels were made using an ambulatory phonation monitor (APM), which measures voice parameters directly from skin vibrations on the neck. Simultaneous sound level meter measurements of various parameters were also carried out in the classrooms. The room acoustic parameters of the classrooms were measured separately to the APM measurements. Measurements have been taken in a range of classrooms as part of a pilot study. Results will be reported as to the effects of different acoustic environments in the classroom on the teachers' voice levels.

5:00

4pNSa12. Acoustic quality on board ships. Robin D. Seiler and Gerd Holbach (Naval Architecture Ocean Eng., Berlin Inst. of Technol., Salzufer 17-19, Bldg. SG1, Secr. SG 6, Berlin 10587, Germany, r.seiler@tu-berlin.de)

Approaches to determine acoustic quality on board ships are usually based on a three- or five-stage classification system using critical A-weighted sound pressure levels. At times other criteria such as the sound insulation between cabins, impact noise from upper decks, speech interference levels or general noise-rating curves are also taken into account. With regard to increasing requirements of passenger comfort and crew accommodation, a more detailed evaluation of auditory perception on board would be worthwhile. Furthermore, other industrial sectors have stated that psychoacoustic or room acoustic models are useful tools to analyze and guarantee product-sound quality. In order to find better indicators for the quantification of acoustic performance of (luxury) vessels, audio material was acquired at a sea trial and evaluated with the help of a paired-comparison listening test in the laboratory by 30 test-persons. Also, the possibility to comment the judgments was given to the subjects. The results were then analyzed by using the statistic model of the "Law of Comparative Judgment" and compared by correlation analysis with physical, psychoacoustic, and room acoustic parameters. Highly correlating parameters could be identified.

THURSDAY AFTERNOON, 6 JUNE 2013 — 511CF, 1:00 P.M. TO 3:20 P.M.

Session 4pNSb

Noise and Architectural Acoustics: Noise Control

Fabian Probst, Chair
Res. & Development, DataKustik GmbH, Gewerbering 5, Greifenberg 86926, Germany

Contributed Papers

1:00

4pNSb1. Airborne sound insulation as a measure for noise annoyance. Reinhard O. Neubauer and Jian Kang (School of Architecture, Univ. of Sheffield, Theresienstr. 28, Ingolstadt, Bavaria 85049, Germany, r.neubauer@sheffield.ac.uk)

There is currently a lack of measure to describe airborne sound insulation in terms of subjective evaluation of noise annoyance. With a given sound insulation value, different kinds of sound signals could produce rather different hearing sensation levels. Physical noise measurements to describe airborne sound insulation often cannot solve problems in terms of noise annoyance, and psychoacoustic metrics are increasingly used. Recently, new results of evaluating sound insulation spectra by single-numbers have been adapted for practical applications such as in ISO 16717-1. In this paper, comparisons are carried out to demonstrate how single-number ratings are affected by non-steady-state sounds. The effect of a sound insulation having a frequency dip of 6 dB has also been examined. It is well known that noises with tonal components could be rather annoying, so that it would be of significance to examine if a frequency depending sound insulation can act as a filter for tonal components. In this paper, it will be shown that psychoacoustic magnitudes like loudness, sharpness, and fluctuation strength can largely account for different aspects, especially if airborne sound insulation is supposed to describe hearing sensation.

1:20

4pNSb2. Noise reduction from large machineries by using sound enclosures. Hyun-Sil Kim, Jae-Seung Kim, Seong-Hyun Lee, and Yun-Ho Seo (Acoust. and Noise Res. Team, Korea Inst. of Machinery and Mater., Yusung-Gu Jangdong 171, Daejeon 305-343, South Korea, hskim@kimm.re.kr)

A sound enclosure is an effective measure to reduce the noise emitting from the large noise sources such as diesel engines and gas turbines. In this study, insertion loss prediction of the large enclosure is presented. Inside the enclosure, diffuse sound field is assumed, and there exist no air leakages. Insertion loss is predicted by using statistical energy analysis (SEA). From the energy equilibrium equations, sound pressure inside the enclosure is derived in terms of the acoustic power from the machinery. Insertion loss is defined as the ratio between acoustic power inside and transmitted power outside the enclosure. It is shown that sound radiation from the panel vibration can be neglected compared to that transmitted through panel. Insertion loss predictions are compared to the measurements. The enclosure size is 6.4 m x 2.65 m x 4.8 m (L x W x H) and 4.5 m x 2.5 m x 2.0 m, where panel consists of 1.5 mm steel plate and 70 mm mineral wool. The comparisons show good agreements. It is concluded that to increase the insertion loss, panel must have a large sound transmission loss and sound absorption coefficient inside the enclosure must be high.

1:40

4pNSb3. Noise reduction in working areas by the application of absorbing baffle-systems. Fabian Probst (Res. & Development, DataKustik GmbH, Gewerbering 5, Greifenberg 86926, Germany, info@datakustik.com)

Baffle systems are arrangements of absorbing panels that allow free flow of air and therefore do not disturb the acoustic climate. This is one of the reasons why they are often used in industrial environments because there is no need to take into account aspects of thermal isolation that may be a problem with closed suspended ceilings. A method to determine the absorption coefficient of baffle systems was derived and published in 2008 [Probst W.: "Sound absorption of baffle systems", Lärmbekämpfung Nr.2 (2008)]. A method is now presented how such systems can be taken into account if the acoustic behavior of even complex rooms is determined by computer modeling. For simple cases, the mentioned analytical method can be applied, and it is shown that the results are in good agreement with the detailed simulation. But this detailed simulation allows to determine the acoustic influence of partially covered areas with different heights and otherwise complex layouts with absorbing appliances.

2:00

4pNSb4. Experimental study on sound absorbing performance of rubber crumb. Davide Borelli, Corrado Schenone, and Pittaluga Ilaria (DIME - Sez. TEC, Università Degli Studi di Genova, Via all'Opera Pia 15/A, Genova, GE 16145, Italy, davide.borelli@unige.it)

The present paper describes an experimental campaign aimed at the determination of acoustical properties of vulcanized rubber crumbs obtained by the shredding of used tires. In particular, their performance as sound absorbing material in lined ducts has been investigated. The most innovative aspect that is addressed in the study is the use of a waste material such as rubber tires reduced into small grains as a sound absorbing material: tires are in fact usually used at the end of their life cycle as fuel and burned in cement kilns in order to take advantage of their high heating value, with all the problems of pollution that this solution produces. Two kinds of rubber crumbs have been investigated in terms of characteristic dimension of the grains, porosity, and sound absorbing coefficient, while their "in situ" performance when used inside lined and parallel-baffle rectangular ducts has been evaluated measuring their insertion loss. The results of this research show that the acoustical behavior of the tested rubber crumbs is the typical behavior of the granular materials, showing a noteworthy performance of the tested material in the low frequency range, opening a scenery of possible applications where noise has relevant tonal components below 315 Hz.

2:20

4pNSb5. Shape optimization of reactive mufflers using threshold acceptance and finite element method methods. Abdelkader Khamchane, Youcef Khelfaoui, and B. Hamtache (Material Technol. and Eng. Process Lab., Univ. of Abderahman Mira of Bejaia, Route de Targa Ouzemour, Béjaia 06000, Algeria, abdelkader.khamchane@yahoo.fr)

Recently, research on the acoustic performance of reactive mufflers under space constraint becomes important. In this paper, the attenuation performance of single and double expansion-chambers under space constraint is presented. To assess the reactive mufflers, a shape optimization analysis is performed using a novel scheme called threshold acceptance (TA), the best design obtained by the shape optimization method are analyzed by Finite Element Method (FEM). The numerical assessment is based on the maximization of the sound transmission loss (STL) using the Transfer Matrix Method (TMM), a modeling method based on the plane wave propagation model. The FEM solution used to analyze the STL of the shape optimized mufflers is based on the Acoustic Power method, a standard computational code COMSOL Multiphysics is used to analyse in 3D the sound attenuation of the mufflers by the FE method. The acoustical ability of the mufflers obtained is than assessed by comparing the FEM solution with the analytical method. Results show that the maximal STL is precisely located at the desired targeted tone. In addition, the acoustical performance of mufflers with double expansion-chamber is found to be superiors to the other one. Consequently, this approach provides a quick and novel scheme for the shape optimization of reactive mufflers.

2:40

4pNSb6. Numerical mode-matching approach for acoustic attenuation prediction of expansion chambers with single inlet and double outlets. Zhenlin Ji and Zhi Fang (School of Power and Energy Eng., Harbin Eng. Univ., No. 145 Nantong St., Nangang District, Harbin City, Heilongjiang Province, Harbin, Heilongjiang 150001, China, zhenlinji@yahoo.com)

Numerical mode matching (NMM) method is developed to predict the acoustic attenuation performance of expansion chambers with single-inlet and double-outlets. The two-dimensional finite element method is employed to calculate the transversal eigenvalues and eigenvectors, and the mode matching technique is used to determine the modal amplitudes and transmission loss of expansion chamber silencers by combining the boundary conditions at inlet and outlets. For the purpose of validation, the transmission loss predictions of elliptical expansion chambers with single-inlet and double-outlets from the present NMM method and the three-dimensional finite element method (FEM) are compared, and good agreements between them are observed. Then the NMM method is used to investigate the effects of extended lengths and locations of inlet and outlets on the acoustic attenuation performance of elliptical expansion chambers.

3:00

4pNSb7. The impact of neck material on the sound absorption performance of Helmholz resonator. Dong Yang, Min Zhu (Dept. of Thermal Eng., Tsinghua Univ., Rm. 110, Gas Turbine Inst., Bei Jing 100084, China, yd.tsinghua@gmail.com), and Xiaolin Wang (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Helmholtz resonator is an effective acoustic attenuation device at low frequencies and is generally used as passive damper. In this work, parallel perforated ceramics with different perforation diameters were used to improve acoustic impedance at the entry of the resonator and thus achieve better acoustic absorption coefficient and better absorption bandwidth simultaneously. With experimental measurement, ceramics with different perforation diameters are found to improve sound absorption performance of Helmholtz resonator in different extent. At the same time, a model is developed to calculate the resonator's neck mouth impedance and further to predict sound absorption coefficient. Particularly, resonance resistance is considered based on the nonlinear correction to Darcy's law. The results show that large resonance resistance with large perforation diameter materials are due to non-fully developed factor. The largest velocity oscillation amplitude in the resonator neck will lead the Reynolds number up to more than 3000 near the resonance frequency and thus make the nonlinear Forchheimer revision coefficient decrease as Reynolds number increase. Helmholtz resonator with neck filled with sound absorption materials has improved sound absorption capacity. This prediction agrees well with the experiment results and this model can be used to optimize the sound absorption system with Helmholtz resonators.

Session 4pPAa

Physical Acoustics: Nonlinear Acoustics II

Murray S. Korman, Chair
Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402

Contributed Papers

1:00

4pPAa1. Acoustic instability of vortices. Konstantin A. Naugolnykh (Physics, Univ. of Colorado, 325 Broadway, Boulder, CO 80305, konstantin.naugolnykh@noaa.gov)

Large plane is a powerful source of vortex. Disturbances of an axial vortex in a compressible fluid are unstable with respect to Kelvin wave development and produce acoustic wave generation. On the other hand system of counter-rotating vortices (Lamb dipole), with different intensity of vortices, as a result of instability collapse into the center of rotation of vortices. The characteristic time of collapse is in the reciprocal proportion the vortices intensity difference. The evolution of Lamb dipole, determined by the two competing processes of instability, is considered in the presented paper. The sound radiation of Lamb dipole can be used to estimate the vortex structure produced by large plane. The advent of large aircraft, with their attendant large and strong trailing vortex structures, made this a problem of considerable practical concern.

1:20

4pPAa2. General nonlinear acoustical equation of relaxing media and its stationary solutions. Nonna Molevich, Rinat N. Galimov (Theor. Phys., P. N. Lebedev Physical Inst. of RAS, Samara Branch, NovoSadovaya 221, Samara 443011, Russian Federation, molevich@fian.smr.ru), Vladimir G. Makaryan, Dmitriy I. Zavershinskii, and Igor P. Zavershinskii (Physics, Samara State Aerosp. Univ., Samara, Russian Federation)

During previous years, the conditions for the negative second (bulk) viscosity existence were found in a large number of nonequilibrium media. The media with negative viscosity possess a number of new properties including acoustical activity. In the present paper, we investigate the nonlinear stage of acoustical perturbation evolution in acoustically active nonequilibrium media using three models: the vibrationally excited gas with the exponential model of relaxation, the chemical active two component mixture with a nonequilibrium reaction and media with the general heat-loss function. The general nonlinear acoustical equation describing stationary density profiles behind the shock wave front in these media is obtained and solved. Its low- and high- frequency limits correspond to the Kuramoto-Sivashinsky equation and the Burgers equation with a source, respectively. Stationary structures of general equation, the conditions of their establishment and all their parameters are found analytically and numerically. In acoustically active media, it is predicted the existence of the stationary solitary pulse. Unstable weak shock waves disintegrate into the sequence of solitary pulses. Their amplitude, form, and speed are rigidly defined by the nonequilibrium degree and do not depend on the initial weak perturbation amplitude. For weak nonequilibrium degree, this solitary pulse is described analytically.

1:40

4pPAa3. Perturbation methods for the spectral analysis of a weakly nonlinear acoustic field generated by a transient insonation. Hassina Khelladi (Faculté d'Electronique et Informatique, Département Instrumentation, Université des Sci. et de la Technologie Houari Boumediene, BP32, El Alia, Alger, Bab Ezzouar 16111, Algeria, hassinakhelladi@yahoo.fr) and Fahim Rahmi (Faculté des Sci. de l'Ingénieur, Université M'Hamed Bougara, Boumerdes, Algeria)

In this study an infinite plane piston is considered which oscillates with finite amplitude in unbounded homogeneous fluid. To illustrate the shape of the weakly nonlinear acoustic field generated by a transient insonation, the function defined by Funch/Muller representing a damped sinusoid is used to simulate the temporal waveform of the piston vibration. The acoustic transient wave generates harmonic components as result of nonlinearities in the material properties of the fluid and in the convective terms of the propagation equation. The mathematical approach is based upon the generalized Burgers' equation, which is a good approximation of the exact equation for the nonlinear propagation when diffraction effects are assumed to be negligible. The pressure amplitude of the fundamental is considered large enough to produce the second harmonic wave. Under the quasi-linear approximation, an analytical description of the fundamental and the second harmonic waves is elaborated. To simulate the spectrum of the weakly nonlinear acoustic field, the pressure field is written in a perturbation series where the first term is the linear acoustic field that results from an infinitesimal oscillation of the piston and the second term contains the first nonlinear contribution to the acoustic field due to the finite amplitude effects.

2:00

4pPAa4. Strongly nonlinear waves – A new trend of nonlinear acoustics. Oleg V. Rudenko (Radiophysics, Nizhni Novgorod State Univ., Campus Grasvik, Karlskrona, Blekinge 37179, Sweden, oru@bth.se) and Claes M. Hedberg (School of Eng., Blekinge Inst. of Technol., Karlskrona, Blekinge, Sweden)

Strongly nonlinear waves (SNWs) and extreme states of matter are key physical concepts. A SNW is a wave whose amplitude is on the order of the material's internal strength. High-intensity light is a weak nonlinear wave (WNW) if its electric field is weaker than the intra-atomic: $E \ll 10^{11}$ V/m. A SNW irreversibly modifies a medium, up to its destruction. In vacuum a wave 10^{18} V/m is strong, when it creates electron-positron pairs. In acoustics SNWs must be distinguished from WNWs which also can display strongly nonlinear phenomena. When a shock front appears at a distance of 10^2-10^3 wavelengths in water, nonlinearity is weak but strongly expressed. The acoustic pressure is 10^5-10^6 Pa, much less than the internal pressure 2.2×10^9 Pa. However, impurities decrease the breaking strength, and waves create bubbles at smaller pressures. An explosive wave is also a SNW, breaking solids. Nuclear explosions may even create new chemical elements. For WNWs the equation of state can be expanded in power or functional series. However, these cannot be used in three cases. First, if the equation contains singularities, like for "clapping" and Hertz nonlinearities of heterogeneous solids. Second, if the series is divergent. Third, when the linear term is absent and the higher nonlinearities dominate. Such SNWs appear in mechanics and in quantum field theory. Mathematical models of SNW, solutions, and new phenomena observed experimentally will be presented.

2:20

4pPAa5. Chaos and beyond in a water filled ultrasonic resonance system. Laszlo Adler (Ohio State Univ./Adler Consultiants Inc., 1560 Gulf Blvd #1002, Clearwater, FL 33767, ladler1@aol.com), William T. Yost, and John H. Cantrell (NASA-Langley Res. Ctr., Hampton, VA)

Finite amplitude ultrasonic wave resonances in a one dimensional liquid-filled cavity are reported. The resonances are observed to include not only the expected harmonic and subharmonic signals but chaotic signals as well. The nonlinear features of this system were recently investigated and are the focus of this presentation. An ultrasonic interferometer having optical precision was constructed. The transducers having the frequency range of 1–10 MHz, driven by a high power amplifier. Both an optical diffraction system and a receiving transducer were used to assess the generated resonance response in the cavity. At least five regions of excitation are identified: (1) Linear region: at low intensity of the ultrasonic wave the diffraction pattern of a light beam is symmetric. (2) Nonlinear region: with increased sound amplitude the diffraction pattern becomes asymmetrical. (3) Subharmonic region: further increase of the amplitude above a threshold value leads to the generation of subharmonics. (4) Chaos: increasing the drive amplitude to a second threshold level the diffraction pattern is smeared out indicating a time-chaotic region. (5) Beyond chaos: further increase of the amplitude results again a stable diffraction pattern. A first-principle-based explanation of the experimental findings is presented. [Work supported by the Aircraft Aging Program, at NASA Langley Research Center. Pending approving by NASA.]

2:40

4pPAa6. Compressed parametric difference frequency sound with chirp signal. Hideyuki Nomura, Hideo Adachi, Tomoo Kamakura (The Univ. of Electro-Communications, 1-5-1 Chofugaoka, Chofu-shi 182-8585, Japan, h.nomura@uec.ac.jp), and Gregory T. Clement (Harvard Med. School, Boston, MA)

The directivity of parametric difference frequency sound is narrower than that of linear sound with same frequency radiated from a sound source with same aperture size. In addition, the parametric difference frequency sound can propagate a long distance in a dissipative medium, because sound absorption at low frequency is less than those at usual ultrasound frequency for measurements and medical imaging. However, for applications of parametric difference frequency sound on measurements and imaging, that has the disadvantage of low spatial resolution because. In this study, we proposed the application of pulse compression technique with chirp signal to parametric difference frequency sound for the improvement of spatial resolution. Nonlinear propagation of ultrasound in water was numerically simulated to confirm the realization of compressed parametric difference frequency sound. A sound source at center frequency of 1 MHz was driven by up and down linear chirp signals to generate chirp difference frequency with band width of 100 to 400 kHz, and the autocorrelation function of generated difference frequency sound was calculate to archive pulse compression. The results indicated the realization of pulse compressed parametric difference frequency sound with desired pulse width which is inversely proportional to the band width.

3:00

4pPAa7. Exploration of third-order nonlinear acoustics for projection of narrow-beam lower-frequency underwater beams. Robert M. Koch (Chief Technol. Office, Naval Undersea Warfare Ctr., Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708, Robert.M.Koch@navy.mil), Richard A. Katz (Sensors & Sonar Systems, Naval Undersea Warfare Ctr., Newport, RI), Allan D. Pierce (P.O. Box 339, East Sandwich, MA), and Derke R. Hughes (Sensors & Sonar Systems, Naval Undersea Warfare Ctr., Newport, RI)

Projection systems are considered where two or three frequencies (e.g., f1, f2, and/or f3) are simultaneously projected into water in a parallel fashion. High near-field amplitudes produce beams of frequencies equal to any linear combination of f1, f2, and f3, with integer coefficients n1, n2, and n3 (possibly zero or negative). Interest here is in the case where the magnitudes of the coefficients sum to three, associated with a third-order nonlinearity. The question addressed is that of how large the amplitude of the far-field signal will be. The considered causes of the nonlinearities are (1) the convective derivative term in the total time derivative of the fluid velocity, and (2) the higher coefficients in the expansion of the fluid density in terms of the deviation of the pressure from its ambient value. These coefficients are derived from data reported by Holton *et al*. [J. Acoust. Soc. Am. (1968)] on the sound speed in water. A perturbation technique is explored starting with the basic nonlinear equations of compressible time-dependent fluid dynamics, where at each step one has a simultaneous set of coupled linear and homogeneous equations with the source terms dependent on the solutions of the analogous equations corresponding to the previously considered orders.

THURSDAY AFTERNOON, 6 JUNE 2013 — 519B, 3:20 P.M. TO 5:00 P.M.

Session 4pPAb

Physical Acoustics: Thermoacoustics II

Albert Migliori, Chair
Los Alamos National Laboratory, Los Alamos, NM 87545

Contributed Papers

3:20

4pPAb1. Hysteresis of mode transitions with varying cavity length of bottle-shaped thermoacoustic prime movers. Bonnie Andersen, David Pease, and Jacob Wright (Physics, Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84057, bonniem@uvu.edu)

Transition regions to higher resonant modes of a bottle-shaped thermoacoustic prime mover (neck: 5.39 cm long, 1.91 cm ID; variable cavity with a sliding piston: up to 38 cm long, 4.76 ID) were studied. The neck and cavity regions behave as coupled resonators. A variable cavity with a sliding piston was constructed to study the nature of the device as the cavity length is varied. The dominant mode of operation depends on the length of the cavity, favoring successively higher modes as the cavity length increases, occurring roughly where the higher mode overlaps with the fundamental frequency of the neck region. As the cavity length is increased, the transition of the dominant frequency from the fundamental to the first overtone occurs. However, when the length is then shortened, transition back to the fundamental does occur at the same piston position, revealing hysteresis. Three transitions to higher modes were observed. The hysteresis was studied as a function of input power (12.0–16.5 W) and stack volume filling factor (3.0–4.9%). Preliminary results indicate that the transition region occurs shallower in the cavity and the hysteresis widens as the input power is increased. Decreasing the stack mass causes an increase of the hysteresis width, but has no strong effect on the hysteresis depth.

3:40

4pPAb2. Numerical simulations of a transient behavior in the onset of thermoacoustic marginal oscillations in a looped tube. Dai Shimizu and Nobumasa Sugimoto (Mech. Sci., Grad. School of Eng. Sci., Osaka Univ., 1-3, Machikaneyama, Toyonaka, Osaka 560-8531, Japan, dai_shimizu@me.es.osaka-u.ac.jp)

This paper simulates a transient behavior in the onset of spontaneous, thermoacoustic oscillations of a gas in a looped tube with a so-called stack sandwiched by hot and cold heat exchangers. Numerical

simulations are performed to solve an initial-value problem by employing the linearized boundary-layer theory. Initial conditions are chosen in such a way that the gas under a uniform pressure is given impulses at two locations in the outside of the stack to cancel with each other. Because of installation of the stack and heat exchangers, account is taken of discontinuity in the temperature gradient, the cross-sectional area and the wetted perimeter of the gas passages by imposing continuity of mass and energy fluxes. Except for a special value of the temperature ratio of the two heat exchangers, the pressure fluctuates significantly around the initial value transiently, but it eventually tends to grow indefinitely or decay out. At a marginal case, it is observed that a traveling wave tends to emerge spontaneously. The traveling wave always propagates in the sense from the hot to cold heat exchangers in the tube outside of the stack. It is shown qualitatively that the traveling wave is enhanced as the porosity lowers.

4:00

4pPAb3. Marginal conditions of thermoacoustic oscillations in a looped tube based on thick and thin diffusion-layer theories. Hiroaki Hyodo (Dept. of Mech. Sci., Grad. School of Eng. Sci., Univ. of Osaka, Motoyama-kitamati3-4-25, Kobe 6580003, Japan, h_hyodo@mars.me.es.osaka-u.ac.jp) and Nobumasa Sugimoto (Dept. of Mech. Sci., Grad. School of Eng. Sci., Univ. of Osaka, Toyonaka, Japan)

Marginal conditions for the onset of thermoacoustic oscillations of a gas in a looped tube with a "stack" inserted are examined by using two approximate equations for thick and thin thermoviscous diffusion layers in comparison with a span length of a gas passage. The equations are derived from the general thermoacoustic-wave equation valid for any thickness of the layer in the linear framework. Applying those approximate equations, respectively, to the gas in the stack and that in the outside of the stack, a frequency equation is derived by imposing matching conditions at both ends of the stack. Seeking a real solution for the frequency, the marginal conditions are obtained numerically for the temperature ratio at both ends of the stack. The ratio depends not only on the span length of one passage in the stack but also on its porosity. It is revealed that the temperature ratio decreases with increasing the span length and the porosity as well. This is the case when the thick diffusion layer is assumed in the stack. It is also revealed that a traveling wave tends to emerge in the tube outside of the stack in the sense from the hot end to cold end.

4:20

4pPAb4. Introduction of conical phase adjuster for thermoacoustic system. Shin-ichi Sakamoto (Univ. of Shiga Prefecture, 2500 Hassaka, Hikone 522-8533, Japan, sakamoto.s@usp.ac.jp), Manabu Inoue, Yosuke Nakano (Doshisha Univ., Kyotanabe, Japan), Yuichiro Orino, Yoshitaka Inui, Takumi Ikenoue (Univ. of Shiga Prefecture, Hikone, Japan), and Yoshiaki Wanatane (Doshisha Univ., Kyotanabe, Japan)

We have proposed a phase adjuster for the thermoacoustic system and succeeded in improving the energy conversion efficiency from heat to the sound. A phase adjuster in a cylindrical shape was used in the past experiments. In this report, a conical phase adjuster is introduced. The inside diameter in one side of the phase adjuster is 20.5 mm, and the other is 39.5 mm. The length of phase adjuster is 45 mm.The phase adjuster is placed in two ways; the larger inner diameter of the phase adjuster is placed in the left side and the smaller is in the right (with PA L39.5 R20.5); the smaller inner diameter is in the left and the larger is in the right (with PA L20.5 R39.5).The total length of the loop is 3.3 m and the phase adjuster is placed 1.125 m away from the upper end of the prime mover stack in the clockwise direction. The measurements are conducted in three conditions: without phase adjuster and with each phase adjusters. Both the sound pressure and sound intensity generated in the thermoacoustic system with phase adjuster are greater than those without phase adjuster. The biggest sound intensity is observed with PA L39.5 R20.5.

4:40

4pPAb5. Early onset of sound in Rijke tube with abrupt contraction. Konstantin Matveev and Rafael Hernandez (School of Mech. and Mater. Eng., Washington State Univ., MME School, WSU, Pullman, WA 99164-2920, matveev@wsu.edu)

Rijke tube is a system convenient for studying thermoacoustic instabilities both experimentally and theoretically. Common Rijke setups involve tubes with constant cross-sections. With the goal to reduce supplied heat necessary to excite acoustic modes, a segmented Rijke tube is constructed comprising two pipes of different diameters. Experiments with regulated mean flow and supplied heat demonstrate that thermoacoustic instability in the segmented tube occurs at about half of the heat addition rate required for sound onset in a tube with uniform cross-section. This finding suggests that simpler experimental means can be used for studying thermoacoustic instabilities due to significant reduction of required heat and highest temperatures in the system. Also, some practical devices with complicated resonators may appear to be more prone to instabilities of this sort. Stronger coupling between acoustic modes due to their enhanced non-orthogonality in segmented resonators can result in richer nonlinear effects. A simplified energy-based model is developed that predicts the onset of instability in both straight and segmented Rijke tubes.

Session 4pPP

Psychological and Physiological Acoustics and Speech Communication: Computational Modeling of Sensorineural Hearing Loss: Models and Applications

Michael G. Heinz, Cochair
Speech, Lang., and Hearing Sci. & Biomedical Eng., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907

Torsten Dau, Cochair
Ctr. for Appl. Hearing Res., Technical Univ. of Denmark, Kongens, Lyngby 2800, Denmark

Chair's Introduction—12:55

Invited Papers

1:00

4pPP1. Hearing impaired cochlear response simulation. Marcos F. Simon Galvez and Stephen J. Elliott (SPCG, ISVR, Rm. 3049, Bldg. 13, ISVR, Univ. of Southampton, Southampton, Hampshire SO17 1BJ, United Kingdom, mfsg1e10@soton.ac.uk)

A model is introduced which allows the vibration of the basilar membrane to be estimated for different degrees of hearing loss. The model is based on a discrete lumped parameter model of the human cochlea, which uses a three dimensional description of the fluid coupling. The hearing losses are assumed to be caused by the combined malfunction of the outer hair cells (OHCs), the inner hair cells (IHCs), and the endocochlear potential driving the system. OHC loss and damage to endochoclear potential are modeled by a reduction of the cochlear amplifier gain, which is obtained by reducing the feedback gain of the OHCs. IHC loss is modeled as an overall reduction in basilar membrane response. The distribution of OHC and IHC loss along the cochlea are derived using an iterative method, which matches the output vibration amplitude of the model to that assumed to generate the hearing impaired audiogram.

1:20

4pPP2. Modeling disrupted tonotopicity of temporal coding following sensorineural hearing loss. Michael G. Heinz and Kenneth S. Henry (Speech, Lang., and Hearing Sci. & Biomedical Eng., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, mheinz@purdue.edu)

Perceptual studies suggest that sensorineural hearing loss (SNHL) affects neural coding of temporal fine structure (TFS) more than envelope (ENV). Although the "quantity" of TFS coding is degraded only in background noise, Wiener-kernel analyses suggest SNHL disrupts tonotopicity (i.e., the "quality") of TFS coding for complex sounds more than ENV coding. Specifically, auditory-nerve (AN) fibers in noise-exposed chinchillas can have their dominant TFS component located within their tuning-curve tail (i.e., the wrong place) while their ENV response remains centered at CF. Here, the ability of a AN model [Zilany and Bruce (2007)] to replicate this dissociation between TFS and ENV tonotopicity was evaluated. By varying the degree of outer- and inner-hair-cell damage, hypothesized factors such as hypersensitive tails and tip-to-tail ratio were evaluated. The model predicted the main trends in our physiological data: (1) no loss of tonotopicity for lower CFs without a clear tip/tail distinction, (2) more easily disrupted TFS tonotopicity than ENV (without requiring hypersensitive tails), and (3) disruption of both TFS and ENV tonotopicity for severely degraded tips. This computational approach allows exploration of the interaction between tip-tail ratio and phase-locking roll-off, and whether amplification strategies can restore cochlear tonotopicity. [Work supported by NIH grants R01-DC009838 and F32-DC012236.]

1:40

4pPP3. Physiological prediction of masking release for normal-hearing and hearing-impaired listeners. Ian C. Bruce (Dept. of Elec. & Comput. Eng., McMaster Univ., Rm. ITB-A213, 1280 Main St W, Hamilton, ON L8S 4K1, Canada, ibruce@ieee.org), Agnès C. Léger (Institut d'Etude de la Cognition, École normale supérieure, Paris, France), Brian C. Moore (Dept. of Experimental Psych., Univ. of Cambridge, Cambridge, United Kingdom), and Christian Lorenzi (Institut d'Etude de la Cognition, École normale supérieure, Paris, France)

Léger *et al*. [J. Acoust. Soc. Am. (2012)] measured the intelligibility of speech in steady and spectrally or temporally modulated maskers for stimuli filtered into low- (<1.5 kHz) and mid-frequency (1–3 kHz) regions. Listeners with high-frequency hearing loss but near to clinically normal audiograms in the low- and mid-frequency regions showed poorer performance than a control group with normal hearing, but showed preserved spectral and temporal masking release. Here, we investigated whether a physiologically accurate model of the auditory periphery [Zilany *et al*., J. Acoust. Soc. Am. (2009)] can explain these masking release data. Intelligibility was predicted using the Neurogram SIMilarity (NSIM) metric of Hines and Harte [Speech Commun. (2010) and (2012)]. This metric can make use of either an "all-information" neurogram with small time bins or a "mean-rate" neurogram with large time bins. The average audiograms of the different groups of listeners from the study of Léger *et al*. were simulated in the model by applying different mixes of outer and/or inner hair cell impairment. Very accurate predictions of the human data for both normal-hearing and hearing-impaired groups were obtained from the all-information NSIM metric (i.e., taking into account phase-locking information) with threshold shifts produced predominantly by OHC impairment (and minimal IHC impairment).

2:00

4pPP4. Neural-scaled entropy as a model of information for speech perception. Joshua M. Alexander and Varsha Hariram (Speech, Lang., and Hearing Sci., Purdue Univ., Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, alexan14@purdue.edu)

Neural-Scaled Entropy (NSE) is an objective metric used to quantify "information" available in speech consequent hearing loss, hearing aid signal processing, and distortion from various environmental factors. One pursuit is to use NSE to find optimum hearing aid settings that maximize speech perception. Inspired by the Cochlear-Scaled Entropy model [Stilp *et al*., J. Acoust. Soc. Am. 2112–2126 (2010)], NSE uses the neural spike output at the inner hair cell synapse of an auditory nerve model [Zilany *et al*., J. Acoust. Soc. Am. 126, 2390–2412 (2009)]. Probability of spike output from fibers sampled at equidistant places along the model cochlea is computed for short duration time frames. Potential information is estimated by using the Kullback-Liebler Divergence to describe how the pattern of neural firing at each frame differs from preceding frames in an auto-regressive manner. NSE was tested using nonsense syllables from various perceptual studies that included different signal processing schemes and was compared to performance for different vowel-defining parameters, consonant features, and talker gender. NSE has potential to serve as a model predictor of speech perception, and to capture the effects of sensorineural hearing loss beyond simple filter broadening. [Work supported by NIDCD RC1DC010601.]

2:20

4pPP5. Modeling detection of 500-hertz tones in reproducible noise for listeners with sensorineural hearing loss. Laurel H. Carney (Biomedical Eng. and Neurobiology & Anatomy, Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu), Junwen Mao (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY), Kelly-Jo Koch (Biomedical Eng. and Neurobiology & Anatomy, Univ. of Rochester, Rochester, NY), and Karen A. Doherty (Commun. Sci. and Disord., Syracuse Univ., Syracuse, NY)

Detection of tones in reproducible noises provides detailed patterns of hit and false-alarm rates across sets of masker waveforms. Analysis of these detection patterns can identify the cues or combination of cues listeners use for detection in narrowband and wideband noise. Recent work has shown that diotic detection patterns of listeners with normal hearing (NH) are significantly correlated to energy and envelope cues; fine-structure cues also contribute for wideband maskers. Detection patterns are best predicted by an optimal cue-combination model based on signal-detection theory. In this study, listeners with mild to moderate sensorineural hearing loss (HL) were tested using the same waveforms. Their diotic detection patterns were best predicted by energy or envelope cues, with little contribution of fine-structure timing. Also, unlike NH patterns, predictions of HL patterns were rarely improved by an optimal combination of cues. For dichotic detection, NH patterns were better predicted by the slope of the interaural envelope difference (SIED) than by ITD or ILD cues. For HL patterns, the SIED cue, a nonlinear combination of ITD and ILD cues, generally did not predict detection patterns. These results illustrate differences between NH and HL listeners in the use and combination of cues for detection in noise.

2:40

4pPP6. A perceptual model of auditory deafferentation. Enrique A. Lopez-Poveda and Pablo Barrios (Inst. of Neurosci. of Castilla y Leon, Univ. of Salamanca, Calle Pintor Fernando Gallego 1, Salamanca, Salamanca 37007, Spain, ealopezpoveda@usal.es)

Overexposure to intense sound produces temporary threshold shifts but permanent loss of afferent nerve terminals. Here, we present a vocoder designed to explore the perceptual consequences of this type of damage. The basic idea is that the spike train produced by an individual auditory afferent resembles a stochastically digitized binary version of the stimulus waveform and that the quality of the waveform representation in the whole nerve depends on the number of aggregated spike trains. Sounds were processed by filtering them into ten adjacent frequency bands. For the signal in each band, multiple spike trains were then obtained in an attempt to mimic the different representations of that signal conveyed by different auditory afferents innervating a given cochlear region. The aggregated spike train was multiplied by the original signal to obtain an acoustic version of the simulated nerve waveform. Tone-in-noise and speech-in-noise perception tests were performed by young, normal-hearing listeners using different numbers of afferents per frequency band. Results support that deafferentation impairs perception in noise more than in quiet. The proposed vocoder may be extended to model other types of hearing damage and to guide the design of hearing aids and cochlear implants.

3:00–3:20 Break

3:20

4pPP7. Understanding hearing impairment through model predictions of brainstem responses. Sarah Verhulst, Hari Bharadwaj (Auditory Neurosci. Lab, Boston Univ., 677 Beacon St., Boston, MA 02215, save@bu.edu), Golbarg Mehraei (Boston, Massachusetts), and Barbara Shinn-Cunningham (Auditory Neurosci. Lab, Boston Univ., Boston, MA)

Latencies of auditory brainstem response (ABR) wave-V decrease with increasing stimulus level, an effect often ascribed to broadened auditory filters. Following this hypothesis, hearing-impaired subjects with broad auditory filters should exhibit shorter wave-V latencies than normal-hearing listeners. Hearing anomalies resulting from the preferential degradation of low spontaneous rate (LS) auditory nerve (AN) fibers with intact thresholds have recently received attention. However, their effect on the ABR wave-V latency are yet to be elucidated. Here, a model of ABR investigates the relationships between wave-V latency and various forms of hearing damage. ABR wave-Vs are predicted from a model consisting of a nonlinear cochlear model (Verhulst *et al*., J. Acoust. Soc. Am. (in press)], an AN synapse model [Zilany *et al*., J. Acoust. Soc. Am. **126** (2009)], and a model of the cochlear nucleus (CN) and IC [Nelson and Carney, J. Acoust. Soc. Am. **116** (2004)]. Simulations predict that level changes cause smaller latency shifts in AN than in the IC, likely due to how inhibition/excitation shapes CN and IC responses. Furthermore, the increase in wave-V latency with decreasing click-to-noise ratios is predicted from LS fiber responses at low click-to-noise ratios. Preliminary simulation results suggest that wave-V latencies at different click-to-noise ratios may help diagnose LS damage.

3:40

4pPP8. Computational modeling of tinnitus development. Roland Schaette (UCL Ear Inst., 332 Gray's Inn Rd., London WC1X 8EE, United Kingdom, r.schaette@ucl.ac.uk)

Animal models and human neuroimaging studies have shown that tinnitus is generated through pathologically altered spontaneous activity of neurons in the central auditory system. Sensorineural hearing loss has been identified as an important trigger for the development of these aberrant patterns of neuronal activity, but the functional mechanisms that underlie this process have not yet been pinpointed. Using computational models, we have investigated which neuronal plasticity mechanisms could account for the development of neuronal correlates of tinnitus after hearing loss. We could show that a model based on the principle of activity stabilization through homeostatic plasticity can explain the development of neuronal hyperactivity as observed in animal studies. Moreover, the model's predictions of tinnitus frequencies from the audiograms of patients with noise-induced hearing loss and tonal tinnitus are close to the observed tinnitus pitch. The model thus proposes a specific mechanism for how plasticity in the central auditory system could lead to the development of tinnitus after cochlear damage. The model also predicts that central auditory structures may show increased response gain, which could explain why tinnitus and hyperacusis often occur together. Moreover, the homeostasis model is consistent with recent experimental findings from tinnitus patients with normal audiograms, and it explains why auditory deprivation through an earplug can lead to the occurrence of phantom sounds.

4:00

4pPP9. An auditory model for intelligibility and quality predictions. James Kates (Speech Lang. Hearing Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309, James.Kates@colorado.edu)

The perceptual effects of audio processing in devices such as hearing aids can be predicted by comparing auditory model outputs for the processed signal to the model outputs for a clean reference signal. This paper presents an improved auditory model that can be used for both intelligibility and quality predictions. The model starts with a middle-ear filter, followed by a gammatone auditory filter bank. Two-tone suppression is provided by setting the bandwidth of the control filters wider than that of the associated analysis filters. The analysis filter bandwidths are increased in response to increasing signal intensity, and compensation is provided for the variation in group delay across the auditory filter bank. Temporal alignment is also built into the model to facilitate the comparison of the unprocessed reference with the hearing-aid processed signals. The amplitude of the analysis filter outputs is modified by outer hair-cell dynamic-range compression and inner-hair cell firing-rate adaptation. Hearing loss is incorporated into the model as a shift in auditory threshold, an increase in the analysis filter bandwidths, and a reduction in the dynamic-range compression ratio. The model outputs include both the signal envelope and scaled basilar-membrane vibration in each auditory filter band.

4:20

4pPP10. Modeling loudness for impaired ears and applications to fitting hearing aids. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

Models of loudness for impaired ears are based on the assumption that cochlear hearing loss can be partitioned into a component due to outer hair cell dysfunction, resulting in reduced frequency selectivity and a more rapid growth of neural response with increasing level (reduced compression), and a component due to inner hair cell dysfunction, which reduces the neural response at all levels. In the first two models that were developed in Cambridge, the filtering and compression that take place on the basilar membrane were modeled as sequential processes, which is not physiologically realistic. Nevertheless, the models were able to account for many aspects of loudness, and were used to develop methods of fitting multi-channel compression hearing aids that have proven to be effective. More recently, a model of loudness has been developed in which the filtering and compression are modeled using a physiologically plausible nonlinear filter bank. This has also been applied to the fitting of hearing aids. Factors not included in the models include central plasticity resulting from altered auditory input, possible consequences of the operation of the efferent system, and the influence of cognitive factors such as perceived distance of the sound source and perceived vocal effort.

4:40

4pPP11. Modeling music perception in impaired listeners. Martin McKinney, Kelly Fitz (Starkey Hearing Technologies, 6600 Washington Ave S, Eden Prairie, MN 55344, martin_mckinney@starkey.com), Sridhar Kalluri, and Brent Edwards (Starkey Hearing Res. Ctr., Berkeley, CA)

We employ computational models of loudness and pitch perception to better understand the impact of sensorineural hearing loss on music perception, with the aim of guiding technology development for hearing-impaired listeners. Traditionally, hearing aid development has been geared towards improving speech intelligibility and has largely failed to provide adequate restoration of music to those with hearing loss. One difficulty with trying to improve music perception in impaired listeners is the absence of a good quantitative measure of music reception, analogous to speech reception measures like word-recognition rate. Psychoacoustic models for loudness and pitch allow us to guage quantitative parameters relevant to music perception and make predictions about the type of deficits listeners face. We examine the impact of hearing loss to predicted measures of loudness, specific loudness, pitch, and consonance and make suggestions on possible methods for restoration.

5:00

4pPP12. A model-based hearing aid: Psychoacoustics, models, and algorithms. Stephan D. Ewert, Steffen Kortlang, and Volker Hohmann (Medizinische Physik, Universität Oldenburg, Carl-von-Ossietzky Str. 9-11, Oldenburg 26129, Germany, Stephan.ewert@uni-oldenburg.de)

In hearing aids, amplification and dynamic range compression typically aim at compensating the deficits associated with outer-hair cell (OHC) loss. Nevertheless, success shows large inter-individual variability and hearing-impaired listeners generally still have considerable problems in complex acoustic communication situations including noise and reverberation. These problems could be related to inner-hair cell (IHC) damage and reduced frequency selectivity resulting in a loss of spectro-temporal coding fidelity. Here a model-

based, fast-acting dynamic compression algorithm which aims at approximating the normal-hearing BM input-output function in hearing-impaired listeners is suggested. The algorithm is fitted by estimating low-level gain loss (OHC loss) from adaptive categorical loudness scaling data and audiometric thresholds based on Ewert and Grimm [*Proc. ISAAR* (2012)] and Jürgens *et al*. Hear. Res. **270**, 177 (2011)]. Aided speech intelligibility was measured in stationary and fluctuating noise and related to the estimated OHC loss. To improve diagnostics of OHC and IHC loss, a series of five psychoacoustic measurements was conducted aiming at a direct quantification of IHC damage in a group of six young and elderly normal-hearing and 12 hearing-impaired listeners. A model is suggested to account for the temporal fine-structure detection and discrimination data. [Work funded by BMBF 01EZ0741 and DFG FOR1732.]

THURSDAY AFTERNOON, 6 JUNE 2013 512CG, 1:00 P.M. TO 5:00 P.M.

Session 4pSA

Structural Acoustics and Vibration: Applications in Structural Acoustics and Vibration III

Linda P. Franzoni, Cochair
Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Box 90271, Durham, NC 27708-0271

James E. Phillips, Cochair
Wilson, Ihrig & Assoc., Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608

Contributed Papers

1:00

4pSA1. Dynamics and stability of pneumatically isolated systems. Vyacheslav Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA 92606, vyacheslav.ryaboy@newport.com)

Pneumatic vibration isolation is the most widespread effective method for creating vibration-free environments that are vital for precise experiments and manufacturing operations in optoelectronics, life sciences, microelectronics, nanotechnology, and other areas. The modeling and design principles of a dual-chamber pneumatic vibration isolator continue to attract attention of researchers. On the other hand, behavior of *systems* of such isolators was never explained in the literature in sufficient detail. After a brief summary of the theory and a model of a single standalone isolator, the dynamics of a system of isolators supporting a payload is considered with main attention directed to three aspects of their behavior: first, the static stability of payloads with high positions of the center of gravity; second, role of gravity terms in the vibration transmissibility; third, the dynamic stability of the feedback system formed by mechanical leveling valves. The direct method of calculating the maximum stable position of the center of gravity is presented and illustrated by three-dimensional stability domains. A numerical method for feedback stability analysis of self-leveling valve systems is provided, and the results are compared with the analytical estimates for a single isolator. The relation between the static and dynamic phenomena is discussed.

1:20

4pSA2. Dissipative effects in the response of an elastic medium to a localized force. Douglas Photiadis (NRL, 4555 Overlook Ave. SW, Washington, DC 20375, douglas.photiadis@nrl.navy.mil)

The effect of dissipation on the real part of the admittance of an elastic half-space is typically thought to be unimportant if the loss factor of the elastic medium is small. However, dissipation induces losses in the near field of the source and, provided the size of the source is small enough, this phenomenon can be more important than elastic wave radiation. Such losses give rise to a fundamental limit in the quality factor of an oscillator attached to a substrate. Near field losses associated with strains in the elastic substrate can actually be larger than intrinsic losses in the oscillator itself if the internal friction of the substrate is larger than the internal friction of the oscillator. [Research sponsored by the Office of Naval Research.]

1:40

4pSA3. Modal active control applied to simplified string musical instrument. Simon Benacchio, Adrien Mamou-Mani (Instrumental Acoust., IRCAM, 1 place Stravinsky, Paris, France, simon.benacchio@ircam.fr), Baptiste Chomette, and François Ollivier (Institut d'Alembert, Universite Pierre et Marie Curie, Paris, France)

This study aims to control the vibrational eigenmodes of soundboards in order to modify the timbre of string instruments. These structures are wooden plates of complex shape, excited by a string through a bridge. Their modal parameters are first identified using modal analysis algorithms on experimental measurements. Then a digital controller is designed using these parameters and classic active control methods. The effects of this controller are first studied thanks to time simulation. Prior to applying experimentally this controller, an optimization procedure is carried out to determine the quantity, dimensions and positions of sensors and actuators needed for the control. These best possible specifications are obtained according to the controllability, observability and other optimization criteria. Finally, a real time system using the control procedure is tested on a simplified musical instrument. The experiment is conducted on a rectangular spruce plate, clamped at its boundary and excited by means of a single string. This simple case study is presented here and its results are discussed in terms of eigenmodes modifications.

2:00

4pSA4. Simulation of structural dynamics using a non-polynomial method. Teemu Luostari, Tomi Huttunen (Dept. of Appl. Phys., Univ. of Eastern Finland, P.O. Box 1627, Kuopio 70211, Finland, teemu.luostari@uef.fi), and Peter Monk (Dept. of Mathematical Sci., Univ. of Delaware, Newark, DE)

In structural dynamics, the modeling of steady-state thin plate bending is an important but, especially at high frequencies, computationally challenging problem. When solving the displacement of an elastic thin plate, a fourth order partial differential equation (Kirchhoff's plate equation) needs to be solved. In addition, two boundary conditions are needed in order to uniquely solve the problem. Polynomial methods, such as the finite element method (FEM) and discontinuous Galerkin method (DGM), are generally used to solve the plate dynamics. At higher frequencies the computational burden of a low order FEM becomes rapidly unbearable. Consequently, non-polynomial modeling methods are investigated because of their

capability to solve the problem more efficiently than the standard FEM. The non-polynomial method used in this study is called the ultra weak variational formulation (UWVF). The UWVF uses finite element meshes and it is essentially an upwind DGM with a special choice of basis functions. To date, the UWVF has been successfully used in electromagnetism, acoustics and linear elasticity. We shall show, using a mixture of theory and numerical examples, that the UWVF is feasible for thin plate problems. For these problems the UWVF basis consists of plane wave and evanescent (corner) wave functions.

2:20

4pSA5. Impact of mass ratio and bandwidth on apparent damping of a harmonic oscillator with subordinate oscillator array. Aldo A. Glean, Joseph F. Vignola, John A. Judge, and Teresa J. Ryan (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, 10glean@cardinalmail.cua.edu)

The response of a lightly damped resonator with a set of substantially less massive attached oscillators has been studied. The collection of attached oscillators is known as a subordinate oscillator array (SOA). An SOA can function as an energy sink, extracting vibration energy from the primary mass and thus adding apparent damping to the system. We have shown that the limit of apparent damping achievable for this class of system is the inverse of non-dimensional bandwidth (ratio of the bandwidth to the fundamental frequency of the primary oscillator). In practice, the utility of this result is limited because a great deal of mass (~25% of primary) is required to approach critical damping. The mass of the subordinate set required to achieve the most rapid energy transfer from the primary is proportional to the non-dimensional bandwidth squared. Low apparent Q is achieved by increasing non-dimensional bandwidth. The presentation will describe numerical optimizations that investigate the impact of the SOA bandwidth, the mass ratio (the ratio of the total mass of the SOA to the mass of the primary structure) and the apparent damping of the system.

2:40

4pSA6. Imaging crack orientation using the time reversed elastic nonlinearity diagnostic with three component time reversal. 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 reversed elastic nonlinearity diagnostic (TREND) is a method to allow one to nondestructively evaluate a sample by locating nonlinear scatterers. In the TREND method one creates a localized focus of energy using time reversal at each point of interest. The localized nature of the focus, which is at a higher energy level relative to the wave field nearby thereby amplifying the potential nonlinear signature of the focal location, allows one to image localized nonlinearities. It has also been shown that a focus of energy may be individually created in each of the three independent vector components of vibration using time reversal. Here we show that the use of TREND scans in each of the three vector component directions allows imaging of a crack's orientation. This work is conducted on steel samples, each with cracks at known orientations that were created in a controlled manner. The scaling subtraction method is also used at each scan point to classify the nonlinearity. [Work supported by Institutional Support (LDRD) at the Los Alamos National Laboratory.]

3:00–3:20 Break

3:20

4pSA7. Shock dynamics of random structures. Mauro Caresta, Robin S. Langley, and Jim Woodhouse (Engineering, Univ. of Cambridge, Trumpington St., Cambridge CB21PZ, United Kingdom, maurorestaca@yahoo.it)

Predicting the response of a structure following an impact is of interest in situations where parts of a complex assembly may come into contact. Standard approaches are based on the knowledge of the impulse response function, requiring the knowledge of the modes and the natural frequencies of the structure. In real engineering structures the statistics of higher natural frequencies follows those of the Gaussian Orthogonal Ensemble, this allows the application of random point process theory to get a mean impulse response function by the knowledge of the modal density of the structure. An ensemble averaged time history for both the response and the impact force can be predicted. Once the impact characteristics are known in the time domain, a simple Fourier Transform allows the frequency range of the impact excitation to be calculated. Experimental and numerical results for beams, plates, and cylinders are presented to confirm the validity of the method.

3:40

4pSA8. Dynamic analysis of annular sector plate with general boundary supports. Dongyan Shi, Xianjie Shi (College of Mech. and Elec. Eng., Harbin Eng. Univ., No 145, Nantong St., Harbin, Heilongjiang 150001, China, dongyanshi@gmail.com), Wen L. Li (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI), and Qingshan Wang (College of Mech. and Elec. Eng., Harbin Eng. Univ., Harbin, China)

Dynamic behavior of annular sector plate is an important research topic since they have been extensively used in practical engineering applications. However, the dynamic analysis of annular sector plates with general boundary supports is rarely studied in literature. In this investigation, an analytical method is presented for the vibration analysis of annular sector plates with general elastic boundary supports. Unlike most existing framework, arbitrary elastic boundary supports can be easily realized by setting the stiffness of the two types restraining springs. The displacement field is universally expressed as a new form of trigonometric series expansions with a drastically improved convergence as compared with the conventional Fourier series. Mathematically, such a double Furies series is capable of representing any function (including the exact displacement solution) whose third-order partial derivatives are continuous over the area of the plate. Thus, the double Fourier series solution to the dynamic analysis of the structure is obtained by employing the Raleigh-Ritz method. The accuracy and reliability of the current method are validated by both FEA and reference results under various boundary conditions. The present method can be directly applied to other more complicated boundary conditions and other shape plates.

4:00

4pSA9. Structural element vibration analysis. Kamel Falek, Lila Chalah-Rezgui, Farid Chalah (Faculty of Civil Eng., Usthb, Algiers 16111, Algeria, kfalekgc@yahoo.fr), Abderrahim Bali (Polytechnic National School of Algiers, Algiers, Algeria), and Amar Nechnech (Faculty of Civil Eng., Usthb, Algiers, Algeria)

Various approaches are usually used in the dynamic analysis of beams vibrating transversally. For this, numerical methods allowing the solving of the general eigenvalue problem are utilized. The equilibrium equations, describing the movement, result from the solution of a fourth order differential equation. Our investigation is based on the finite element method. The findings of these investigations are the vibration frequencies, obtained by the Jacobi method. Two types of elementary mass matrix are considered, representing a uniform distribution of the mass along the element and concentrated ones located at fixed points whose number is increased progressively separated by equal distances at each evaluation stage. The studied beams have different boundary constraints representing several classical situations. Comparisons are made for beams where the distributed mass is replaced by n concentrated masses. As expected, the first calculus stage is to obtain the lowest number of the beam parts that gives a frequency comparable to that issued from the Rayleigh formula. The obtained values are then compared to theoretical results based on the assumptions of the Bernoulli-Euler theory. These steps are used after for the second type mass representation in the same manner.

4:20

4pSA10. Study on error of vibration intensity measurement in a flat plate with multiple energy flows entering from outside. Mototaka Hibinaga and Masato Mikami (Grad. School of Sci. and Eng., Yamaguchi Univ., 2-16-1 Tokiwadai, Ube, Yamaguchi, Japan, 1-20-308 Yamakado, Ube, Yamaguchi, Japan, Ube, Yamaguchi 755-0097, Japan, s032ve@yamaguchi-u.ac.jp)

Vibration Intensity (VI) method can show flows of vibration energy in surface of mechanical structures as a vector quantity and is one of the techniques to identify vibration source and vibration transmission path. Some previous studies have showed that if there are multiple vibration sources in the analyzed plate, VI causes error due to superposition of multiple waves. In actual structures, however, the vibration source often exists outside of the analyzed plate which is surrounded by the boundary with reflection, such as curved parts and staged parts. The VI error has not been investigated in such

cases close to actual structures. The purpose of this research is to investigate the influence of superposition of multiple waves which come across the stepped boundary into a flat plate on VI by using finite element method analysis. This model also simulates the case with the superposition of progressive wave entering from the outside of the plate and reflected wave caused at a boundary of the plate. The results of calculation showed that the superposition of waves causes a difference between VI and real vibration energy flow, depending on frequency and plate size.

4:40

4pSA11. Mid-frequency vibrations of a double-leaf plate with random inhomogeneities. Hyuck Chung (SCSM, Auckland Univ. of Technol., PB 92006 Auckland, Auckland 1142, New Zealand, hchung@aut.ac.nz)

Predicting vibrations of composite structures such as double-leaf plates is difficult because of the large number of components and random inhomogeneities in the components. In the low and high frequency ranges, the components may be homogenized, and consequently the model of a structure becomes simple enough to be mathematically and computationally tractable. However the vibrations in the mid-frequency range cannot be predicted using such methods because the wavelengths are comparable to the size of the components and junctions between components. Simply adding more details, e.g., higher resolution in finite element mesh, will not result in more accurate predictions. In this paper a double-leaf plate is modeled using the Kirchhoff plate and Euler beam theories. The elastic moduli and junctions are allowed to be inhomogeneous over the plates and beams. These inhomogeneities are simulated as continuous smooth random functions rather than series of discrete random numbers. The random functions are incorporated into the model using the variational formulation and the Fourier expansion of the vibration field. Various probability density functions are tested for the inhomogeneities. Then the distribution of resonant frequencies and the vibration field are studied and compared with other models.

THURSDAY AFTERNOON, 6 JUNE 2013 — 515ABC, 1:00 P.M. TO 3:20 P.M.

Session 4pSCa

Speech Communication: Auditory Feedback in Speech Production II

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

1:00

4pSCa1. Cortical mechanisms of integrating auditory feedback with vocal pitch control. Jean Mary Zarate (Psychology, New York Univ., 6 Washington Place, Rm. 275-276, New York, NY 10003, jean.m.zarate@nyu.edu)

Precise vocal pitch regulation is crucial for both speech and song. The pitch of a speaker's voice can indicate the intent of a sentence, set the emotional context of a conversation, or distinguish meanings in tonal languages. In singing, accurate vocal pitch is the single most important element needed to properly produce notes and melodies. Vocal pitch regulation requires the integration of auditory feedback processing with the vocal motor system, also known as audio-vocal integration; however, the neural substrates governing this integration have been elusive. Recent functional magnetic resonance imaging (fMRI) studies of singing with pitch-shifted feedback are presented here to outline the neural mechanisms of audio-vocal integration for voluntary vocal pitch regulation, and to discuss the effects of long-term vocal training on vocal performance and neural activity during vocal pitch regulation.

1:20

4pSCa2. Cortical plasticity in the sensorimotor control of voice induced by auditory cognitive training. Hanjun Liu, Weifeng Li, and Zhaocong Chen (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., 58 Zhongshan 2nd Rd., Guangzhou, Guangdong 510080, China, lhanjun@mail.sysu.edu.cn)

Multiple lines of evidence have shown that motor-related brain regions can be activated during passively musical listening or beat perception, indicating a connection between auditory and sensorimotor systems. In the present study, we sought to examine whether the neural processing of auditory-vocal integration can be shaped by short-term cognitive training related to auditory attention and working memory. Auditory cognitive training consisted of a ten-day backward digit span task, in which digits embedded in various noise at different SNR levels were presented and subjects were required to repeat the digits in the reverse order. Before and after the cognitive training, subjects also participated in a vocal motor task, in which they heard their pitch auditory feedback unexpectedly altered upwards (50 and 200 cents) during sustained vocalization and their neurophysiological responses were recorded. The results revealed a significantly improved performance on the backward digit span task after the training. Moreover, cortical responses indexed by P2 amplitude to pitch perturbations in voice auditory feedback were significantly increased after the training compared with those before the training. These findings provide evidence that plastic cortical changes in the sensorimotor control of voice can be caused by auditory cognitive training.

1:40

4pSCa3. Neural evidence for state feedback control of speaking. John F. Houde (Otolaryngol. – Head and Neck Surgery, Univ. of California San Francisco, 513 Parnassus Ave., S362, San Francisco, CA 94143, houde@phy.ucsf.edu) and Srikantan S. Nagarajan (Radiology, Univ. of California San Francisco, San Francisco, CA)

An important recent development in neuroscience has been the use of models based on state feedback control (SFC) to explain the role of the central nervous system in motor control. In SFC, control is based on internal feedback of an estimate of the dynamic state of the thing (e.g., arm) being controlled. Within the internal loop, the state is predicted from outgoing motor commands and corrected by comparing the feedback expected to result from this state with actual incoming sensory feedback. SFC has received scant attention in the speech community, but the indirect role it suggests for feedback can account for much of what is known about the role of feedback in speech motor control. Our lab has been investigating how well SFC also accounts for the neural correlates of auditory feedback processing during speaking. Our principal approach has used magnetoencephalography to record the cortical activity of speakers as they hear themselves speaking, but recently, we have also completed an auditory feedback study based on electrocorticography. Many of the results of these studies have supported the SFC model, but some have posed challenges for it, which will be discussed. [Work supported by NSF grant BCS-0926196 and NIH grant R01-DC010145.]

2:00–2:20 Break

2:20

4pSCa4. Intentionality and categories in speech motor control. Takashi Mitsuya (Psychology, Queen's Univ., 62 Arch St., Humphrey Hall, Kingston, ON K7L3N6, Canada, takashi.mitsuya@queensu.ca) and Kevin G. Munhall (Psych. & Otolaryngol., Queen's Univ., Kingston, ON, Canada)

Actions are organized around goals or intentions. In speech production, there has been no agreement on how best to discuss speech goals. However, the auditory feedback perturbation methodology provides a window into the nature of speech goals. To the extent that subjects are sensitive to variation in an acoustic attribute, this attribute must be part of the controlled intention of articulation. In this presentation, we will review a series of studies that speak to this issue. In one study, we examined how intentionality of speech production influences compensatory formant production by instructing subjects to use a cognitive strategy in order to make the feedback sound consistent with the intended vowel. In other studies, we have explored the specificity of vowel formant compensation by comparing cross-language differences. The results indicate that speech goals are (1) very specific, defined by a phonemic category and its relationship with neighboring categories, and (2) multivariate. We will discuss these results by contrasting compensatory behaviors in reaching and limb movements to those observed in speech studies. The presence of a system of categories in speech may result in differences in the way speech goals are represented.

2:40

4pSCa5. Exploring auditory-motor interactions in normal and disordered speech. Jason A. Tourville, Shanqing Cai, and Frank H. Guenther (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 677 Beacon St, Boston, MA 02215, jtour@bu.edu)

Auditory feedback plays an important role in speech motor learning and in the online correction of speech movements. Speakers can detect and correct auditory feedback errors at the segmental and suprasegmental levels during ongoing speech. The frontal brain regions that contribute to these corrective movements have also been shown to be more active during speech in persons who stutter (PWS) compared to fluent speakers. Further, various types of altered auditory feedback can temporarily improve the fluency of PWS, suggesting that atypical auditory-motor interactions during speech may contribute to stuttering disfluencies. To investigate this possibility, we have developed and improved Audapter, a software that enables configurable dynamic perturbation of the spatial and temporal content of the speech auditory signal in real time. Using Audapter, we have measured the compensatory responses of PWS to static and dynamic perturbations of the formant content of auditory feedback and compared these responses with those from matched fluent controls. Our findings indicate deficient utilization of auditory feedback by PWS for short-latency online control of the spatial and temporal parameters of articulation during vowel production and during running speech. These findings provide further evidence that stuttering is associated with aberrant auditory-motor integration during speech.

3:00

4pSCa6. The role of auditory feedback in speech development: A study of compensation strategies for a lip-tube perturbation. Lucie Menard (Phonet. Lab., Universite du PQ a Montreal, CP 8888, succ. Centre-Ville, Montreal, QC H3C 3P8, Canada, menard.lucie@uqam.ca), Pascal Perrier (Speech and Cognition, GIPSA-lab, Grenoble, France), and Jerome Aubin (Phonet. Lab., Universite du Quebec a Montreal, Montreal, QC, Canada)

The role of auditory feedback in speech development was investigated through a study of compensation strategies for a lip-tube perturbation. Acoustic, articulatory, and perceptual analyses of the vowels /i/, /y/, and /u/ produced by ten 4-year-old French speakers and ten adult French speakers were conducted under two conditions: normal and with a 15-mm-diameter tube (for /y/ and /u/) or a 5-mm-diameter tube (for /i/) inserted between the lips. Ultrasound and acoustic recordings of isolated vowels were made in normal condition before any perturbation (N1), for each of the 20 trials in the perturbed condition (P), and in normal condition after the perturbed trials (N2). Data reveal that adult participants moved their tongue in the P condition more than children subjects, to compensate for F1 and F2 alteration induced by the tube. Except for /y/, the perturbation was generally at least partly compensated during the perturbed trials in adults and children, but children did not show a typical learning effect. Results are analyzed from the perspective of (i) goal specification in speech production in the acoustic and/or somatosensory domain, and (ii) the maturity of representations of the motor apparatus in the brain.

Session 4pSCb

Speech Communication: Production and Perception I: Beyond the Speech Segment (Poster Session)

Sam Tilsen, Chair
Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

4pSCb1. Prosodic effects on speech gestures: A shape analysis based on functional data analysis. Christine Mooshammer (Linguist Dept., USC, 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693, mooshamm@usc.edu), Lasse Bombien (Inst. for Phonet. and Speech Processing, LMU, Munich, Germany), and Jelena Krivokapic (Linguist Dept., Yale, New Haven, CT)

Prosodic phrasing and prominence modulate the production of speech gestures by increasing their duration and often also the amplitude. Here we investigate whether the shape of bilabial and dorsal constriction and release gestures is affected by phrase boundary strength and lexical stress. Articulatory movements of four speakers of German were recorded with 3D EMA. Eight bisyllabic test words starting with the clusters /kn, kl, ps, pl/ and either stressed on the first or on the second syllable were embedded in sentences that elicited phrase boundaries of different strengths. Vertical tongue back movements and lip aperture from the constriction onset to the release offset of the initial consonant were extracted and time and amplitude normalized. In order to investigate shape differences a functional principal component analysis [see Ramsay and Silverman (2002)] was performed. The resulting factor scores quantify gestural shape differences. The results indicate that boundary strength affects the skewness, i.e., targets are achieved later for stronger preceding boundaries. Furthermore, movement curves of initial consonants in unstressed syllables were more peaked than in stressed ones. By applying this method, global shape differences due to prosodic modulation of articulatory gestures can be extracted without recourse to specific landmarks. [Work supported by NIH DC03172.]

4pSCb2. Exploring prosodic boundaries: Gradiency and categoricity of prosodic boundaries in articulation. Jelena Krivokapic (Linguistics, Yale Univ., 370 Temple St., 204, New Haven, CT 06520-8366, jelena.krivokapic@yale.edu), Christine Mooshammer (Linguistics, Univ. of Southern California, Los Angeles, CA), and Mark Tiede (Haskins Labs., New Haven, CT)

The prosodic hierarchy is a core concept of prosodic theory. Despite this, the number of categories in the hierarchy, and the structural relationships between them, is not clear. For English, a major and a minor category above the word level is usually assumed, and some studies have suggested additional categories [cf. Shattuck-Hufnagel and Turk (1996)]. Each of these category levels is assumed to be marked by categorically distinct boundaries, but experimental evidence for this view is sparse. In this work, EMA is used to investigate this notion of the prosodic hierarchy [following a preliminary analysis in Krivokapić and Ananthakrishnan (2007)]. Seven subjects read six repetitions of 48 sentences, each containing one, two or three prosodic boundaries, for a total of 56 boundaries per speaker. The predicted boundary strength varied from a weak clitic boundary to a strong sentential boundary. The produced boundaries are evaluated using a Gaussian mixture model analysis. The results bear on the question whether prosodic boundaries behave categorically (i.e., prosodic boundary values cluster within a small number of categories), or in a gradient manner (i.e., they are more evenly spread), thus supporting an alternative view of the prosodic hierarchy. [Work supported by NIH.]

4pSCb3. Acoustic characteristics of intervocalic stop lenition in American English. Dominique Bouavichith and Lisa Davidson (Linguistics, New York Univ., 10 Washington Place, New York, NY 10003, bouavichith@nyu.edu)

Descriptions of English and other languages have claimed that intervocalic stops are often lenited to fricatives or approximants in connected speech, but few acoustic analyses of factors that affect lenition have been reported for American English [cf. Lavoie (2001)]. In this analysis, intervocalic voiced stops produced in bi- and trisyllabic words during story reading are examined (participants $N = 14$). The first result shows that American English speakers never lenite to fricatives, but rather produce approximants whenever lenition occurs. Second, stress plays an essential role: 51% of stops are lenited when stress is on the first syllable (e.g., "yoga"), but only 7% of stops lenite when stress is on the second syllable (e.g., "begin"). Overall, approximant productions are significantly higher for /d/ (which becomes a flap–63%) and /g/ (70%) as compared to /b/ (43%). For both stress placements, stop productions are longer and lower in intensity than approximant productions. Another significant factor is frequency, as higher frequency words are produced as approximants more often. These acoustic findings are generally consistent with Kingston's (2006) claim that lenition within words and phrases occurs to minimize interruptions to the prosodic unit and indicate that the current constituent is ongoing.

4pSCb4. Variability attenuates sensitivity to acoustic detail in cross-language speech production. Sean Martin, Lisa Davidson (Linguistics, New York Univ., 10 Washington Place, New York, NY 10003, sean.martin@nyu.edu), and Colin Wilson (Cognit. Sci., Johns Hopkins Univ., Baltimore, MD)

In the production of non-native consonant clusters, speakers' systematic errors have been attributed to the influence of native-language phonotactics [Dupoux *et al.* (1999)]. However, recent models of non-native speech production suggest that speakers are also sensitive to acoustic details [Wilson *et al.* (2012)]. We examine whether speakers' sensitivity to phonetic detail is modulated by variability in the speech signal, and whether they abstract away from subphonemic detail given sufficient variability. This was tested by presenting English speakers with ill-formed clusters (e.g., bdafa, tmape, zgade) containing systematically manipulated sub-phonemic acoustic properties: stop burst duration and amplitude for stop-initial clusters, and the presence/absence of pre-obstruent voicing (POV) for voiced clusters. In experiment 1, which presented stimuli produced by a single Russian talker, significant effects were found for the duration manipulations on the rates of epenthesis, the amplitude manipulation on consonant change/deletion errors, and the POV manipulation on the rate of prothesis. In experiment 2, which contained stimuli produced by three talkers, there was a substantial attenuation of the influence of the acoustic manipulations on speakers' productions. These results suggest that an account of non-native speech production that models the relative contribution of phonotactics and phonetic detail must incorporate information about variability in the environment.

4pSCb5. Deriving functional load of phonemes from a prosodically extended neighborhood analysis. Mafuyu Kitahara (School of Law, Waseda Univ., 1-6-1 Nishiwaseda, Shinjuku-Ku, Tokyo 1698050, Japan, kitahara@waseda.jp), Keiichi Tajima (Dept. of Psych., Hosei Univeristy, Tokyo, Japan), and Kiyoko Yoneyama (Dept. of English Lang., Daito Bunka Univeristy, Tokyo, Japan)

The functional load of phonemes is a long-standing, but not a mainstream notion in modern linguistics: that some pairs of phonemes distinguish more words than other pairs is intuitively plausible, but hard to quantify. Meanwhile, neighborhood effects in word recognition and production have been one of the central topics in psycholinguistics, leading to a wide variety of investigations. However, the Greenberg-Jenkins calculation, the most common definition of phonological neighborhood, deals only with deletion, addition, and substitution of phonemes, lacking any consideration of prosody. For example, homophones, which cannot be segmental neighbors and thus excluded in most neighborhood research, can be distinctive if lexical accent is specified. The role of onset/rhyme distinction in neighborhood calculation has been discussed, but morae, another basic unit of prosody, were not mentioned in the literature. We propose a novel method for calculating the functional load based on a prosodically extended neighborhood analysis. It is a frequency-weighted neighborhood density summed across neighbors for a particular phoneme. Accentual distinctions, morae or syllables, and context effects within a word are taken into account. The proposed method gives a better account for the difference in the acquisition order of segments across languages. [Work supported by JSPS.]

4pSCb6. Durational characteristics of English by Chinese learners of English: A case of the northeast dialect speakers of Chinese. Kiyoko Yoneyama (Daito Bunka Univ., 1-9-1 Takashimadaira, Itabashi, Tokyo 175-8571, Japan, yoneyama@ic.daito.ac.jp)

This study examined the durational patterns of English production by Chinese learners of English. The same production experiment of Mochizuki-Sudo and Kiritani (J. Phonet. **19**, 231–248) was conducted to investigate how Chinese speakers of English control the durational properties of inter-stressed intervals (ISIs) and target stressed vowels by compressing the stressed vowels when unstressed syllables are added. Five male and three female English non-proficient Chinese speakers of the northeast dialect participated in the study. They were all recruited from Changchun city in Jilin province of China. They graduated from the same high school, and they did not have any experience of oversea study in any English speaking country. Durations of the ISIs and the target vowel were analyzed. The tentative analysis revealed that the ISI durations produced by the non-proficient Chinese learners of English showed the similar durational patterns like the non-proficient Japanese learners of English did in Mochizuki-Sudo and Kiritani (1991). The analysis of stressed vowel durations showed that when they produced the vowel durations, they didn't show the similar durational patterns like the non-proficient Japanese, but those to the American speakers. The implications from the results will be discussed in the paper. [Work supported by JSPS.]

4pSCb7. Auditory free classification of nonnative speech by nonnative listeners. Eriko Atagi and Tessa Bent (Speech and Hearing Sci., Indiana Univ., 200 S Jordan Ave., Bloomington, IN 47405, eatagi@indiana.edu)

Nonnative listeners are less accurate than native listeners at classifying talkers by regional dialect [Clopper and Bradlow (2009)]. This decrement may be due to less robust knowledge about the underlying sound structure of the target language or less extensive experience with socio-cultural phonetic variation in the target language. To disentangle the contribution of these two factors, this study examined native and nonnative listeners' abilities to classify talkers who varied on another sociophonetic dimension: foreign accent. Unlike regional dialect variation, nonnative listeners typically have more experience with nonnative speech than native listeners, particularly for talkers with the same native language background. Using auditory free classification, native listeners of English and native Korean listeners classified talkers by perceived native language. Talkers consisted of nonnative talkers from six native language backgrounds and native talkers. Results demonstrated that native listeners were nearly perfect at grouping the native talkers together, but Korean listeners were much less accurate. Further, Korean listeners did not show an advantage for grouping Korean-accented talkers together. These results suggest that nonnative listeners' less robust linguistic representations of the target language can hinder their abilities to attend to the acoustic-phonetic features that index dialect and accent categories. [Work supported by NIH-NIDCD Grant R21DC010027.]

4pSCb8. Categorization of regional, international, and nonnative accents. Amal Akbik, Eriko Atagi, and Tessa Bent (Speech and Hearing, Indiana Univ., 200 S Jordan Ave., Bloomington, IN 47405, aakbik@indiana.edu)

Auditory free classification—a task in which listeners classify auditory samples into unconstrained groups—has provided insight into perceptual representation and categorization for several sources of speech variability including U.S. regional dialects, nonnative accents, and foreign languages. Within these studies, phonological markedness and geography have emerged as central organizing principles. However, previous studies were limited by including only one source of variability. To address this gap, the perception of U.S. regional dialects, international English dialects, and nonnative accents was investigated within one classification task. Listeners categorized talkers based on perceived location of origin. Cluster analysis demonstrated a perceptual divide between native and nonnative talkers. Native talkers were further delimited by geographic proximity into Southern Hemisphere, U.S., and United Kingdom groups. One exception was the consistent grouping of Southern U.S. talkers with talkers from England. Nonnative talkers were grouped into three major branches: French and German, Asian, and other. The "other" branch primarily consisted of less familiar accents. The results suggest that native and nonnative accents are perceived as separate categories regardless of accent markedness. Additionally, when listeners are presented with a wide range of dialects and accents, geography remains an important organizing principle. [Research supported by IU Hutton Honors College.]

4pSCb9. Is consonant harmony assimilatory? Ian Maddieson (Dept. of Linguist, UNM, Univ. of New Mexico, MSC03-2130, Albuquerque, NM 87131-0001, ianm@berkeley.edu)

Ohala (1990) claimed that vowel harmony is in origin a product of vowel-to-vowel assimilation across intervening consonants. Gafos (1999) essentially argued that consonant harmony may similarly be assimilatory. For this to be the case, intervening segments—typically vowels—must be capable of transmitting the harmonizing property. For some properties, such as nasality or lip-rounding such "spreading" is non-problematic as these can be properties of either consonant or vowels. An alternative view, e.g., in Hansson (2010), is that consonant harmony (albeit more narrowly defined) is a correspondence or copying process, not an assimilatory effect. In this paper a range of attested varieties of consonant harmony will be evaluated in terms of how plausibly an assimilatory component might be involved. The analysis indicates that consonant harmony patterns vary along a scale of their likelihood to be explicable as assimilatory in nature. Processes such as sibilant harmony may have an assimilatory part, as suggested by Whalen *et al*. (2011) and supported by a limited acoustic study reported here. However, harmony involving certain phonatory and laryngeal features, such as voicing (given that vowels are prototypically already voiced) or ejective production, does not plausibly involve assimilatory transmission of the harmonizing property.

4pSCb10. French listeners' perceptions of prominence and phrasing are differentially affected by instruction set. Caroline Smith (Linguistics, Univ. of New Mexico, MSC 03 2130, Albuquerque, NM 87131-0001, caroline@unm.edu)

Listeners' perception of prosodic structure may differ depending on whether they are instructed to attend to the meaning of a spoken passage, or to the acoustics. Real-time perceptions of prominence and phrasal boundaries were obtained from Rapid Prosody Transcription [Cole *et al*. (2010)]. Twenty naive French listeners were divided into two groups that were given either meaning- or acoustically based instructions for listening to passages of spontaneous speech. While listening, they read an orthographic unpunctuated transcript of the speech. Half of each group first labeled words they perceived as prominent in five passages, then phrasal boundaries in five different passages; the other half performed the tasks in the opposite order. Consistent with previous results, listener agreement (measured by kappa) was higher for labeling boundaries than for prominence. The mean kappa was 0.80 for both meaning- and acoustically based responses, but the difference between prominence and boundaries was greater in meaning-based responses. Listeners labeled more words as prominent than they labeled boundaries, with a greater difference in meaning-based responses than acoustically based, although the frequency of labeling did not differ overall between acoustic- and meaning-based instructions. The divergence between prominence and boundary labeling challenges the assumption that prominence in French derives from pre-boundary position.

4pSCb11. The influence of production latencies and phonological neighborhood density on vowel dispersion. Benjamin Munson (Speech-Lang.-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, munso005@umn.edu)

A number of studies have shown that in read laboratory speech talkers produce vowels in words from dense phonological neighborhoods closer to the periphery of the F1/F2 space than vowels in words from sparse neighborhoods [e.g., Wright (2004)]. The reason for this pattern is hotly debated: some argue that it reflects listener-directed partial compensation for the negative effect of high neighborhood density on perception [Munson and Solomon (2004)]. Others argue that this effect is due to the coactivation of similar words during speech production [Baese-Berk and Goldrick (2009)]. If the patterns are due to the coactivation of phonologically similar words in production, then the degree of hyperarticulation should be related to another measure presumably related to the coactivation of words, production latencies. Specifically, hyperarticulation should be greatest when the production latencies are shortest, as coactivation would presumably speed production [Vitevitch (2002)]. Production latencies and vowel dispersion were measured in read productions of single words varying in phonological neighborhood density. A preliminary analysis of data from 11 adults partially supports Baese-Berk and Goldrick's hypothesis: vowel-space dispersion was better predicted by production latencies than by neighborhood density, albeit in the opposite-than-predicted direction. Analysis of a larger cohort of talkers is ongoing.

4pSCb12. The influence of multiple narrators on adults' listening comprehension. Brittan A. Barker and Cornetta Mosely (COMD, Louisiana State Univ., 63 Hatcher Hall, Baton Rouge, LA 70803, barkerb@lsu.edu)

Research has demonstrated that variable, talker information—such as the number of talkers—affects listeners' perception and processing of linguistic information during various laboratory tasks. In particular, the detrimental effects of multiple talkers are highlighted during online speech perception tasks with little contextual support [isolated word recognition; e.g., Mullennix *et al*. (1989), Ryalls and Pisoni (1997), Sommers and Barcroft (2011)]. Nonetheless, it is unclear how multiple talkers might affect listeners' perception of linguistic information in more complex spoken language tasks utilizing real-time, fluent speech. The present experiments were conducted determine if information contributed by multiple talkers influences adults' auditory story comprehension in the presence of both quiet and background noise. The accuracy and reaction time data did not support the hypothesis that talker information affects the perception of linguistic information during auditory story comprehension. Thus these data bring to light theoretical perspectives that emphasize the importance of looking across experimental tasks to better understand talker-specific information's pattern of influence on spoken language processing [e.g., Sommers and Barcroft (2006), Werker and Curtin (2005)].

4pSCb13. Multi-subject atlas built from structural tongue magnetic resonance images. Jonghye Woo (Dept. of Neural and Pain Sci., Univ. of Maryland School of Dentistry, Baltimore, MD, jschant@gmail.com), Junghoon Lee, John Bogovic (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD), Emi Z. Murano (Dept. of Otolaryngol., Johns Hopkins Univ., Baltimore, MD), Fangxu Xing (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD), Maureen Stone (Dept. of Neural and Pain Sci., Univ. of Maryland School of Dentistry, Baltimore, MD), and Jerry L. Prince (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

Magnetic resonance imaging (MRI) is a widely used technology for non-invasive tongue imaging. MRI can detail tongue and muscle shapes and their variability in both healthy and diseased populations. Such detail can aid significantly in the interpretation of muscle interactions in the tongue, and their relation in normal and disordered speech production. However, the size or shape of the tongue and muscles may vary from one subject to another. In addition, there exists no comprehensive and systematic framework to assess the difference and variability of tongue and muscles in a normalized space. In the present work, we built a multi-subject atlas from 20 normal subjects that are acquired using structural MRI to offer a normalized space on which all subjects from a target population can be mapped and compared. In order to find accurate one-to-one correspondences, we bound the tongue so that each volume had the same vocal tract features. For registration, we utilize symmetric diffeomorphic image registration with cross-correlation, which is widely used in brain image analysis. The atlas facilitates a template-based segmentation in assigning anatomical labels in the images. The tongue atlas is unprecedented and opens new vistas for exploring normal and diseased oral structures and function.

4pSCb14. An examination of the articulatory characteristics of prominence in function and content words using real-time magnetic resonance imaging. Zhaojun Yang, Vikram Ramanarayanan (Elec. Eng., Univ. of Southern California, 1124 W. 29th ST, Apt 2, Los Angeles, CA 90007, zhaojuny@usc.edu), Dany Byrd (Linguistics, Univ. of Southern California, Los Angeles, CA), and Shri Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

We examine the functional coupling between articulatory characteristics of prominence, such as articulator speed, and its acoustic characteristics, such as F0 and acoustic energy, for content words (nouns) and function words (e.g., articles, prepositions and conjunctions) using real-time magnetic resonance imaging data. We use Granger causality ideas to test the degree and direction of causal influence between the chosen articulatory and acoustic measures for function and content words. We further apply functional canonical correlation analysis to these measures to understand the covariant behavioral modes of the articulatory and acoustic measures. After controlling for word duration, we observe that articulatory speed generally has a significant causal influence on F0, especially for longer content words, but observe no such effect in the opposite direction. Notably, we do not observe this effect for function words in most cases. We further observe a tighter coupling of canonical weight functions of articulatory speed and acoustic-prominence characteristics (F0 and energy) for the content words as compared to the function words considered. These observations provide support for the hypothesis that prominence realized during content words may result from a close coupling between articulatory and acoustic characteristics such as articulatory speed and F0, with suggestions of a directional causal relationship. [Work supported by NIH.]

4pSCb15. Do formant frequencies correlate with Japanese accent? Yukiko Sugiyama (Sci. and Technol., Keio Univ., Hiyoshi 4-1-1, Kohoku-ku, Yokohama 223-8521, Japan, yukiko_sugiyama@mac.com) and Tsuyoshi Moriyama (Media and Image Technol., Tokyo Polytechnic Univ., Atsugi, Japan)

Formant frequencies were examined as a possible acoustic correlate to Japanese accent. A perception study conducted by the first author of the present study used synthetic speech stimuli in which the harmonic structure of each spectrum (F0) was removed from speech produced in a normal manner and replaced by white noise. The stimuli created this way ensured that the only property altered from the normal speech would be the presence or absence of the F0. The result found that Japanese listeners reliably identified minimal pairs of words that differed only by accent, indicating that F0 is not the only acoustic cue to Japanese accent. Since previous studies on whispered speech report a positive correlation between the pitch height intended by speakers and formant frequencies, formant frequencies were measured as a possible correlate to accent. However, the analysis did not find any correlation between the F0 movements observed in the original normal speech and the formant frequencies measured in the synthetic speech. This result suggests that acoustic properties other than the F0 are also affected in whispered speech and claims made about the positive correlation between the intended pitch and formant frequencies in whispered speech do not hold in normal speech.

4pSCb16. The perception of formant-frequency range is affected by veridical and judged fundamental frequency. Santiago Barreda and Terrance M. Nearey (Dept. of Linguist, Univ. of AB, Edmonton, AB T6G 2E7, Canada, sbarreda@ualberta.ca)

The vowels produced by different speakers vary in terms of their fundamental frequency (f0) and formant frequencies (FFs). Variation in the production of a given vowel category between speakers of different sizes is

primarily according to a single multiplicative parameter (related to speaker vocal-tract length). This parameter, which we refer to as FF-scaling, has an associated perceptual quality that listeners may use to determine apparent speaker characteristics and vowel quality. In a previous experiment [Barreda and Nearey, J. Acoust. Soc. Am. **129**, 2661 (2011)], listeners were trained to identify a limited set of voices based on FF-scaling and f0 differences. The current study presented listeners with large number of voices ($n = 4000$) varying in FF-scaling and f0, arranged in a two-dimensional space where one dimension corresponded to each acoustic characteristic. Listeners were played a voice, and asked to indicate its location on the board, thereby providing an f0 and FF-scaling estimate for the voice. Results indicate that listeners are able to identify voice FF-scaling, and that this decision is informed primarily by veridical voice FF-scaling. However, there is a complicated relationship between perceived f0 and FF-scaling, suggesting an interdependent relationship in the perception of these characteristics.

4pSCb17. Relationship between the durations of rhythm unit with primary and secondary stresses in English speech. Shizuka Nakamura (Grad. School of Lang. and Culture, Osaka Univ., 1-8 Machikaneyama-cho, Toyonaka-Shi, Osaka 5600043, Japan, shizuka@akane.waseda.jp)

In this author's previous study [Nakamura, J. Acoust. Soc. Am. **131**(4, Pt. 2), 3347 (2012)] on the acoustical analysis of the duration structure of rhythm in English speech observed in short sentences uttered by native speakers, the durations of the following rhythm unit showed the smallest variance among native speakers: 1/4 of the preceding unstressed syllable(s) + stressed syllable + 3/4 of the succeeding unstressed syllable(s). The durations of the rhythm unit with a secondary stress were concentrated at half of those of the unit with a primary stress. Therefore, the rhythm can be described by a series of rhythm units with primary and secondary stresses where the latter unit is half the duration of the former. In this study, the relationship between the rhythm units with primary and secondary stresses was investigated from viewpoints of the position of syllables with primary and secondary stresses in a sentence, correlation among rhythm units in a sentence, and individual differences among native speakers. The results show that rhythm units in a sentence that are not only adjacent but also remote can adjust their durations mutually to realize the two to one ratio of the duration of the rhythm unit with primary and secondary stresses.

4pSCb18. Gestural reorganization under rate pressure interacts with learned language-specific phonotactics. Ioana Chitoran (Universite Paris 7 Denis Diderot, 175 rue du Chevaleret, Paris 75013, France, ioana.chitoran@dartmouth.edu) and Mark Tiede (Haskins Labs., New Haven, CT)

Studies of articulatory reorganization occurring under rate-driven production pressure can provide a window into speech planning. Previous work shows evidence for stable coordinative structures in speech in which VC patterns reorganize to CV, VCC to CCV, and coronal-labial to labial-coronal order. Such stable modes are argued to result from general physical-biological constraints imposed by the articulatory/auditory system. Here we examine whether stable modes can also arise from linguistic patterns learned on a language-specific basis. The case study is Georgian, which licenses complex onsets disregarding sonority, following instead a phasing pattern whereby degree of overlap varies with order of constriction location (front-to-back /pt/ sequences are more overlapped than back-to-front /tp/). We analyze preliminary data from native speakers repeating the Georgian words [pata] and [tapa] as they tracked an accelerating metronome. Results show: (1) pAta > patA > pta (stress shift followed by elision, licensed by Georgian phonotactics); (2) tApa > tAp (elision only; consistent with Georgian order constraints, not with biomechanical constraints). These patterns are contrasted with similar data from French in which elision is not observed, consistent with French phonotactics. The data thus provide an example in which language-specific structure rather than biomechanical constraints alone mediate gestural reorganization. [Work supported by Fulbright-Hays.]

4pSCb19. The effect of interpretation bias on the production of disambiguating prosody. Wook Kyung Choe and Melissa A. Redford (Dept. of Linguist, 1290 Univ. of Oregon, Eugene, OR 97403-1290, wchoe1@uoregon.edu)

Syntactically ambiguous sentences are frequently strongly biased toward one meaning over another [see, e.g., Tanenhaus and Trueswell (1995)]. This interpretation bias influences listeners' use of disambiguating prosody [Wales and Toner (1979)]. The current study investigated the effect on production. In experiment 1, the default interpretation of a heterogeneous set of 18 syntactically ambiguous sentences was investigated in 40 participants, who completed a question-and-answer task designed to identify intended meaning without making participants aware of potential ambiguity. Results were that 90% of the participants interpreted 11 of the sentences in just one way. There was a weaker interpretation bias for the remaining 7 sentences. In experiment 2, ten speakers were provided with and taught the alternate meanings of the 18 sentences from experiment 1, and then asked to disambiguate the meanings using prosody. Temporal and F0 measures indicated that while all speakers differentiated between meanings in production, only sentences with weak interpretation biases were consistently prosodically disambiguated. Prosodic cues to structure were applied inconsistently to differentiate meaning in sentences with strong interpretation biases. We conclude that disambiguating prosody is grammaticalized only when required by the interpretative norms of the speech community.

4pSCb20. Mechanisms for remembering roots versus affixes in complex words. Anne Pycha (Linguistics, Univ. of Wisconsin, Milwuakee, 3243 N. Downer Ave., Curtin Hall 537, P.O. Box 413, Milwaukee, WI 53211, pycha@uwm.edu)

Previous research has demonstrated that listeners remember low-frequency words (fob) through explicit recollection, but high-frequency words (money) through implicit familiarity [Joordens and Hockley (2000)]. We hypothesize that a similar asymmetry in remembering occurs in morphologically complex words (bleakish), where root frequency (bleak) is always low relative to affix frequency (ish). In our experiment, which modifies a technique developed by Goldinger *et al*. (1999), participants hear both complex and simple words at study. At test, they hear old words in which a portion of the stimulus is masked with soft or loud background noise. For complex words, the masked portion is either the root or the affix (**bleak**ish, bleak**ish**); for simple words, it is the corresponding pseudo-morpheme (**rel**ish, rel**ish**). Participants indicate whether they heard the word previously by making an old/new judgment, followed by a remember/know judgment [Tulving (1985)]. Preliminary results indicate that listeners are more likely to make "old" judgments when morphemes occur in soft (versus loud) background noise, but that this illusion effect is stronger for roots than affixes. Thus, clarity of perceptual input influences the memory of a complex word, but in an asymmetric fashion, suggesting that listeners remember roots and affixes via different mechanisms.

4pSCb21. Lexical bias and prosodic cues: An eye-tracking study of compound/phrase disambiguation. Jessica L. Gamache (Michigan State Univ., 461 Rampart Way, Apartment 106, East Lansing, MI 48823, gamache1@msu.edu)

Despite compounds and phrases exhibiting distinct prosodic cues (e.g., pitch and duration), adults fail to use these cues in certain contexts. In two eye-tracking experiments, I examined English speakers' attention to prosody in adjective-noun string disambiguation to explore a lexical bias previously reported (e.g., "hot dog" is interpreted as the food, regardless of prosody.) In experiment 1, 24 participants were presented with pictures of the compound and phrasal representations for possible compounds (20 known, 10 novel) accompanied by an audio presentation of either a phrasal or compound production. Replicating previous findings, participants sometimes ignored prosody and exhibited biases towards compound pictures for common compounds and towards phrasal pictures for novel compounds. The fixation data reveals similar patterns in which participants appeared to not consider the alternative picture, even when it matched available prosodic cues. In experiment 2, the phrasal/compound pictures were decoupled and placed with unrelated distractors and participants were asked whether either picture matched the audio presentation of a phrasal or compound production. This experiment helps to determine whether the exhibited lexical bias is an artifact of experimental designs using minimal pairs or whether prosody is still ignored when frequency and novelty are not directly competing.

4pSCb22. How robust are lexical effects on phonetic categorization? Mary M. Flaherty and James R. Sawusch (Psychology, SUNY Buffalo, 392 Park Hall, North Campus, Buffalo, NY 14260, maryflah@buffalo.edu)

Previous work has shown that lexical knowledge (whether a stimulus token makes a word or nonword) influences phonetic categorization [e.g., Fox (1984)]. Recent work in our lab examined the effect of lexical influences on speech perception using two tasks (phoneme identification and AXB discrimination) and uncovered some unexpected findings. Listeners who performed identification first showed a robust effect of lexical status on phonetic categorization. However, listeners who performed the discrimination task first showed no effect of lexical status. Since prior research has shown that the lexical effect is fairly robust [see Pitt and Samuel (1993)], the finding that the influence of lexical status can be eliminated by first placing listeners in an AXB discrimination task is the focus of the current research. The AXB task may focus attention on the prelexical, phonetic representation and differences within each phonetic category. This, in turn, eliminates the influence of higher level processes on phonetic perception in the identification task. The present study seeks to replicate this finding and investigate whether other lexical influences on perception (the influence of lexical neighborhood) are also altered by experience with AXB discrimination. Results are discussed in terms of the flow of information during phonetic perception and word recognition.

4pSCb23. Neural processing of voices—Familiarity. Lisa Gustavsson, Petter Kallioinen, Eeva Klintfors (Phonet./Linguist., Stockholm Univ., Stockholm 106 91, Sweden, lisag@ling.su.se), and Jonas Lindh (Inst. of Neurosci. and Physiol., Gothenburg, Sweden)

Brain responses to familiar and unfamiliar voices were investigated with ERPs (Event Related Potentials). Presentation of a stream of one syllable utterances from a female voice established a standard expectation, and similar samples from four other male voices where inserted as unexpected deviants in a typical mismatch paradigm. The participants were 12 students from the basic course in linguistics. Two of the deviant voices were familiar voices of their teachers. The two other deviant voices were matched (same age, sex, and dialect) but unfamiliar to the participants. A typical MMN (Mismatch Negativity) was elicited, i.e., a more negative response to the deviants compared to the standards. In contrast to verbal reports, where only one participant identified any of the deviant voices, the MMN response differed on group level between familiar and unfamiliar voices. MMN to familiar voices was larger. Using teachers' voices ensured naturalistic long term exposure, but did not allow for random assignment to conditions of familiarity making the design quasi-experimental. Thus, acoustic analysis of voice characteristics as well as follow up studies with randomized exposure to voices are needed to rule out possible confounds and establish a causal effect of voice familiarity.

4pSCb24. Effects of prosodic strengthening and lexical boundary on /s/-stop sequences in English. Yoonjeong Lee (Linguistics, Univ. of Southern California, C-128, 3701 Overland Ave., Los Angeles, CA 90034, yoonjeol@usc.edu)

This study examined how the effects of prosodic strengthening (from prosodic boundary and accent) and lexical boundary (e.g., "ice # can" vs. "eye # scan") are acoustic-phonetically realized on English /s/-stop sequences in a sentence. First, the domain-initial strengthening effect was not strictly confined to the first segment, but could extend into the second consonant and, at least partially, into the following vowel in the #/sCV/ sequence (e.g., in "scan"). Second, the accent-induced strengthening effect was robust in all acoustic measures for the #/sCV/ sequence. Third, prosodic strengthening arising with boundary and accent gave rise to the "shortened" VOT for the voiceless stop in the #/sCV/ sequence, suggesting that prosodic strengthening can operate on the phonetic manifestation of a phonological rule to reinforce the language-specific phonetic feature, which is, in this case, {-spread glottis}. Fourth, domain-initial strengthening and accent-induced strengthening differ substantially in some aspects, suggesting that they may be encoded separately in speech production process. Finally, "ice # can" and "eye # scan" were indeed very differently realized, suggesting that the underlying lexical boundary is signaled by fine-phonetic details even when the sequences occurred phrase-internally where they appeared to be homophonous, at least impressionistically, and syllabified the same.

4pSCb25. Syntactic predictability influences duration. Claire Moore-Cantwell (Linguistics, UMass Amherst, 26 Wright Ave., Northampton, MA 01060, cmooreca@linguist.umass.edu)

Building on work by Gahl and Garnsey, 2004, this paper demonstrates that speakers "buy time" during the planning of upcoming low-probability syntactic structures by producing prosodic boundaries with longer duration before low-probability than before high-probability structures. Subject extraction cleft sentences ("It was Edward who (t) loved Lucy.") are more common in corpora than object extraction cleft sentences ("It was Edward who Lucy loved (t).") [Roland *et al*. (2007)], and are also easier to process [e.g., Gibson (1998)]. The duration of the clefted constituent ("Edward") was measured in planned productions of subject- and object-extraction clefts in English. In order to disentangle the probability of each structure from its difficulty level, the probabilities were manipulated within the experiment through training. Participants read aloud: First, two SE and two OE clefts; second, eight SE or eight OE clefts; and finally, another two SE and two OE clefts. Before training, the clefted constituent was longer in OE clefts (mean 407 ms) than in SE clefts (370ms, $t = 2.4$, $p = 0.02$). This difference was no longer present after OE training (OE: 385 ms, SE: 397 ms), but was still present after SE training (OE: 448 ms, SE: 388 ms).

4pSCb26. Coarticulation in a whole event model of speech production. Bryan Gick (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T1Z4, Canada, gick@mail.ubc.ca) and Ian Stavness (Comput. Sci., Univ. of Saskatchewan, Saskatoon, SK, Canada)

Previous models of coarticulation have used varying combinations of advance planning and on-line calculation of weighted averages to determine how temporally overlapping speech sounds interact [see Farnetani and Recasens, Hbk. Phonet. Sci. Wiley-Blackwell (2010)]. A robust model of coarticulation should be able to predict such local interactions, as well as to describe changes resulting from rate variation, which should influence degree of temporal overlap between adjacent events. Using a model of tongue-jaw-hyoid biomechanics [Stavness *et al*., J. Biomech. (2012)], the present paper demonstrates that typical cases of lingual coarticulation can be attributed to the intrinsic biomechanics of the human body in an entirely feed-forward model with no additional machinery. Biomechanical modeling outcomes are compared to speech articulations at different speech rates, and show that naturalistic coarticulatory patterns emerge simply by varying degrees of temporal overlap in a biomechanically realistic model. The built-in mechanics of the human body can handle coarticulatory interactions with no extrinsic model at all, save one that identifies a) the right body parts, and b) the time-course of events. Results are interpreted in a "Whole Event" model of speech. [Research funded by NSERC.]

4pSCb27. Perceptual integration of indexical information in bilingual speech. Charlotte Vaughn and Susanne Brouwer (Linguistics, Northwestern Univ., 2016 Sheridan Rd, Evanston, IL 60208, crvaughn@u.northwestern.edu)

The present research examines how different types of indexical information, namely talker information and the language being spoken, are perceptually integrated in bilingual speech. Using a speeded classification paradigm [Garner (1974)], variability in characteristics of the talker (gender in experiment 1 and specific talker in experiment 2) and in the language being spoken (Mandarin vs. English) was manipulated. Listeners from two different language backgrounds, English monolinguals and Mandarin-English bilinguals, were asked to classify short, meaningful sentences obtained from different Mandarin-English bilingual talkers on these indexical dimensions. Results for the gender-language classification (Exp. 1) showed a significant, symmetrical interference effect for both listener groups, indicating that gender information and language are processed in an integral manner. For talker-language classification (Exp. 2), language interfered more with talker than vice versa for the English monolinguals, but symmetrical interference was found for the Mandarin-English bilinguals. These results suggest both that talker-specificity is not fully segregated from language-specificity, and that bilinguals exhibit more balanced classification along various indexical dimensions of speech. Currently, follow-up studies investigate this talker-language dependency for bilingual listeners who do not speak Mandarin in order to disentangle the role of bilingualism versus language familiarity.

4pSCb28. Prosodic characteristics of two focus types in emphatic context in Thai. Alif Silpachai (Linguist, Univ. of Southern California, 3170 Aintree Lane, Apt/Ste., Los Angeles, CA 90023, silpacha@usc.edu)

This study presents an acoustic analysis of narrow focus (early focus) and broad focus, each in emphatic context (tune) in Thai, with the goal of providing a basic characterization of their prosody. To investigate prosodic realizations, target words from each of the 5 lexical tones in Thai were

placed in subject positions of sentences with SVO structure. Each target word was placed in a sentence in which each syllable contained the same lexical tone as that of the target word. Preliminary results show that F0 measures, especially F0 maximum, minimum, and range, differed between focus types. In particular, narrow focused words were distinguished from non-narrow focused by higher F0 maximum, minimum, and range, while post-focal words contained lower F0 measures. Syllable duration also played a role in signaling narrow focus: focal words in narrow focus sentences were significantly longer than their non-focal counterparts in broad focus sentences. Interestingly, a pitch reset seemed to occur after focused words. Findings from four additional Thai speakers will be presented and there will be a discussion of their relevance to the intonational phonology of Thai.

4pSCb29. Speech rhythm and speech rate affect segmentation of reduced function words in continuous speech. Tuuli Morrill, Laura Dilley, J. Devin McAuley (Michigan State Univ., Oyer Ctr., 1026 Red Cedar Rd., East Lansing, MI 48824, tmorrill@msu.edu), and Mark Pitt (Psychology, The Ohio State Univ., Columbus, OH)

Recent work [Dilley and Pitt, Psychol. Sci. (2010)] has demonstrated that reduced function words in speech can perceptually disappear if the rate of surrounding speech is slowed, even when the acoustic properties of the function word (FW) and its immediate phonetic environment are held constant. An experiment was conducted to determine whether this disappearing word effect could be elicited through a manipulation involving speech rhythm, realized as binary and ternary alternations of high and low tones, as well as through manipulations to context speech rate. 74 participants transcribed 32 sentences containing a FW in which the preceding speech within the utterance was resynthesized with a binary or ternary speech rhythm presented at one of three context speech rates. A binary rhythm in the preceding speech context yielded lower FW report rates than the ternary rhythm. These results suggest that listeners' expectations about speech rhythm and/or syllable grouping affected the number of syllables and words perceived, indicating that such properties may play an important role in word segmentation and lexical access. [Work supported by NSF grant BCS-0847653.]

4pSCb30. Prosody and syntactic structures in continuous speech in French. Sarah Massicotte-Laforge (Psychologie, Université du Québec à Montréal, Groupe de recherche sur le langage, Département de psychologie, section développement, Université du Québec à Montréal, C.P.8888, Succursale Centre-Ville, Montréal, QC H3C 3P8, Canada, massicotte-laforge.sarah@courrier.uqam.ca), Andréane Melançon, and Rushen Shi (Psychologie, Université du Québec à Montréal, Montréal, QC, Canada)

Infant-directed speech contains dominantly multi-word utterances. Segmenting speech into linguistic units is crucial for language acquisition. This study inquires if prosodic cues exist in speech and mark syntactic categories. Participants were Quebec-French speakers. In experiment 1 participants read determiner+noun and pronoun+verb utterances. Nouns and verbs were pseudo-words (e.g., mige, crale) counterbalanced in their occurrences in the utterances, and their prosodic properties (duration, pitch, intensity) were measured. Results showed that the two types of utterances did not differ in prosody; noun versus verb productions of these pseudo-words were equivalent prosodically. Experiment 2 tested whether larger utterances were produced with prosodic cues supporting syntactic units. The same pseudo-words were the final words (counterbalanced) in (1) [determiner+adjective+noun] and (2) [[determiner+noun]+verb] structures. Results showed that the last word as nouns versus verbs differed significantly in duration, pitch and intensity. Moreover, the initial consonant of verb productions was longer, with a distinct preceding pause. The word preceding the verb (2) exhibited boundary cues, differing significantly from the word preceding the noun in (1) in duration, pitch, and intensity. We suggest that these acoustic cues may help children first parse larger utterances and then acquire the syntactic properties of phrases and words based on their distribution.

4pSCb31. Tonogenesis in contemporary Korean with special reference to the onset-tone interaction and the loss of a consonant opposition. Mi-Ryoung Kim (Practical English, Korea Soongsil Cyber Univ., 474 East 14th Alley #9, Eugene, Oregon 97401, kmrg@mail.kcu.ac)

Recent studies show that, besides their effect on the fundamental frequency (f0) contour of the following vowel, Korean stops are undergoing a sound change in which a partial or completer voice onset time (VOT) merger is taking place between aspirated and lax stops. The purpose of this study is to see whether the sound change holds across the three major dialects of Korea (Seoul, Pusan, and Gwangju). The three acoustic parameters, VOT, f0, and H1-H2, were examined. The results show that the effects of onsets on f0 (i.e., onset-tone interaction) were robust across dialects whereas the merging of aspirated and lax stops was not. With respect to H1-H2, lax consonants showed higher breathiness than aspirated counterparts whereas tense consonants did not. With respect to VOT, most aspirated and lax stops were produced with long-lag voicing whereas tense stops were produced with short-lag voicing. However, interspeaker variations were noticeable even within each dialect, indicating that the sound change is still ongoing among Korean speakers. The phenomena correspond to typical tonogenesis properties, characterized by onset-tone interaction and merging of a consonantal opposition. The findings suggest that the sound change in contemporary Korean can be viewed as undergoing tonogenesis.

4pSCb32. Towards a model of Singaporean English intonational phonology. Adam J. Chong (Linguistics, UCLA, 1380 Veteran Ave., Apt. 103, Los Angeles, CA 90024, ajchong@ucla.edu)

Singaporean English (SgE) is a variety of English spoken in Singapore. Recent research has sought to identify the systematic features that make SgE distinct from other varieties of English. Although the intonation of SgE has been described previously [Deterding (1994), Lim (2004), Ng (2011)], no phonological model has yet been proposed. This paper proposes a model of intonational phonology for SgE within the Autosegmental-Metrical phonology framework. Three native speakers were recorded reading declarative and question sentences of varying length and stress pattern. Preliminary results suggest that SgE has three prosodic units above the word: the Accentual Phrase (AP), Intermediate Phrase (ip) and Intonational Phrase (IP). An AP is slightly larger than a word and is characterized by a general LH (rising) contour. The L can be attributable to either an L* tone on a lexically-stressed syllable or an L initial boundary tone if the stressed syllable occurs late in the AP. The AP-final syllable always has a phonologically High boundary tone (Ha). The initial AP is realized in a large pitch range, and subsequent APs within the same ip are realized in successively reduced pitch ranges. Tones of larger prosodic units will also be discussed.

4pSCb33. Calibrating the detection of spontaneous speech: From sentences to noun phrases. Sara Parker and Jennifer Pardo (Psychology, Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, saraphyn@gmail.com)

Many studies have examined the differences between speech that is produced spontaneously as opposed to read from a prepared script. Most of these studies have focused on prosodic measures taken from clauses, sentences, or connected discourse. Furthermore, studies have shown that listeners are able to identify the context of production when presented with sentence-length utterances. The current study examined whether a listener can identify the context for utterances that are briefer than a sentence. A set of 20 talkers (10 male) produced spontaneous descriptions of maps that they then read aloud in a separate session at least one week later. Pairs of sentences that matched in fluency across both contexts were selected, and listeners judged which member of a pair was produced spontaneously. In separate blocks, listeners heard either full sentences, sentence beginnings, sentence endings, or two-word noun phrases excised from sentences. Overall, listeners could identify the spontaneously produced utterances, but only for excerpts longer than two-word noun phrases. These findings indicate that the information present in two-word noun phrases is not sufficient to support perception of spontaneous versus read speaking style.

4pSCb34. Effects of musical experience on perception of audiovisual synchrony for speech and music. Dawn M. Behne, Magnus Alm, Aleksander Berg, Thomas Engell, Camilla Foyn, Canutte Johnsen, Thulasy Srigaran, and Ane Eir Torsdottir (Psych. Dept., NTNU, Trondheim NO7491, Norway, dawn.behne@svt.ntnu.no)

Perception of audiovisual synchrony relies on matching temporal attributes across sensory modalities. To investigate the influence of experience on cross-modal temporal integration, the effect of musical experience on the perception of audiovisual synchrony was studied with speech and music stimuli. Nine musicians and nine non-musicians meeting strict group criteria provided simultaneity judgments to audiovisual /ba/ and guitar-strum stimuli, each with 23 levels of audiovisual alignment. Although results for the speech and music stimuli differed, the two groups did not differ in their responses to the two types of stimuli. Consistent with previous research, responses from both groups show less temporal sensitivity to stimuli with video-lead than audio-lead. No significant between-group difference was found for video-lead thresholds. However, both for the speech and music stimuli, musicians had an audio-lead threshold significantly closer to the point of physical synchrony than non-musicians, indicating the musicians' greater acuity for audiovisual temporal coherence. Overall this leads to a non-significant tendency for a narrower window of synchrony for musicians than non-musicians. Findings are consistent with predictions that cross-modal temporal experience increases threshold acuity for audio-lead, but not for video-lead, and also support theories suggesting greater efficiency with relevant experience.

4pSCb35. Speech rhythm in Korean: Experiments in speech cycling. Younah Chung (Linguistics, UCSD, 8520 Costa Verde Blvd., APT 3420, San Diego, CA 92122, yachung@ucsd.edu) and Amalia Arvaniti (English Lang. & Linguist, Univ. of Kent, Canterbury, United Kingdom)

Korean has not been unanimously classified for rhythm class, and it lacks stress. Thus, it does not fit into views that rhythm rests on alternations of metrical strength. The goal was to examine what, if any, elements are used in Korean for rhythm purposes. It was hypothesized that the onsets of accentual phrases act as beats. The materials were 6 sentences; each was 9 syllables and three APs long. The number of syllables in each AP varied. Syllable composition also varied between CV and CVC. Native speakers repeated each sentence, fitting each repetition into beat intervals at three different metronome rates. Each AP was expressed as a ratio of the entire cycle. Two experiments were conducted. The first experiment suggests that speakers keep AP onsets in phase although syllable count and composition also affect phase. The results support our hypothesis that AP onsets operate similarly to stresses. The second experiment that used waltz rhythm showed that it is the only level of prominence, and no differentiation between the strength of these beats, such that it would produce waltz rhythm, is possible. The results suggest that Korean rhythm is not characterized by multiple levels of alternation between strong and weak constituents.

4pSCb36. Acoustic vowel space size and perceived speech tempo. Melanie Weirich and Adrian P. Simpson (Institut für Germanistische Sprachwissenschaft, Friedrich-Schiller-Universität Jena, Fürstengraben 30, Jena 07743, Germany, melanie.weirich@uni-jena.de)

"Females speak faster than males." Although several studies have proved this stereotype to be wrong [Byrd (1994)], it is still a widespread belief in many languages and within both genders. The interesting question is why. Two findings are particularly relevant regarding this stereotype: First, females reveal a greater acoustic vowel space than males [Hillenbrand *et al*. (1995)]. Second, a stimulus with a moving f0-contour is perceived as faster than the same stimulus with a monotonous contour [Lehiste (1976)]. From that, we might propose that if a dynamic f0 contour triggers the perception of a faster speaking rate, then a larger acoustic vowel space might have the same effect. The reason for female speakers being perceived as speaking at a faster tempo, then, is that they traverse on average a larger acoustic vowel space within the same time-frame than male speakers do. Furthermore, we could also expect a relationship between vowel space size and perceived speech tempo within the same gender. A perception test was conducted with temporally aligned stimuli from 56 female speakers who vary in their vowel space sizes. Results reveal a significant positive correlation between vowel space size and perceived tempo ($r = 0.36$, $p < 0.001$).

4pSCb37. English native monolingual and simultaneous English/Spanish bilingual listeners' perception of foreign accented speech: Cross-language effects on accented speech perception. Somang Moon and Su-Hyun Jin (Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, somang.moon@gmail.com)

The current study was designed to explore whether English monolingual listeners (ENM) perceive foreign- accented speech differently from English/Spanish simultaneous bilingual listeners (ESB) who learned both languages simultaneously at early age. Previous studies suggest that listener's perception of foreign-accented speech is affected by listener's L1 phonological systems. When exposed to two languages simultaneously, bilinguals might be able to exploit the phonetic categories of the two languages in speech perception [Best (1994), Goetry and Kolinsky (2000)]. It would be possible that simultaneous exposure to multi-languages at the early age helps SB listeners to tolerate foreign accents in speech more resulting in better understanding of accented speech. It would be also possible that if two phonological systems of SB listeners interact each other resulting in poorer understanding of foreign accented speech than ENM listeners. ENM and SB listeners completed two speech perception tasks: accent ratings of an English passage and identification of English vowels spoken by Korean-native speakers. Results might suggest the effect of language exposure on accented speech perception.

4pSCb38. An investigation of the three tone system in Tsuut'ina (Dene). Joyce McDonough, Jared O'Loughlin (Dept. of Linguist, Univ. of Rochester, Rochester, NY 14625, joyce.mcdonough@rochester.edu), and Christopher Cox (Dept. of Linguist, Univ. of Alberta, Edmonton, AB, Canada)

This study is part of the documentation and conservation of Tsuut'ina (formerly Sarcee, Sarsi; ISO 639-3: srs), a northern Dene (Athabascan) language by a collaboration of academic and community members. Tsuut'ina is a tone language. Contrary to Dene tonogenesis theory and unlike reports on all other Dene tone languages, Tsuut'ina is reported to have three tones, H, L, M. The tonal system in Dene family has been argued to arise from the loss of laryngealized sonorants in monosyllabic stem codas and incorporation of laryngealization into the nucleus of the stem, resulting in H and L tonal contrasts. The Dene languages additionally exhibit "tonal reversal", a tendency for the Dene tone languages to show "reversed" tonal patterns that postdate the original tonogenesis. In this study we investigate the tonal distribution, realization patterns and tonal alignment in data collected from two fluent speakers reciting prepared wordlists and short discourses. Preliminary investigation indicates that, as reported, three tonal patterns emerge, M tone associated most often with a falling tone, with distinct distribution patterns arguably related to morphological factors. Furthermore M tone is more highly variable. We lay out distribution patterns and interactions with morphology and statistical analyses associated with the data.

4pSCb39. Prosodic correlates of smiled-speech. Caroline Émond (Linguistique, Université du Québec à Montréal, C.P. 8888, Succ. Centre-ville, Montréal, Québec, QC H3C 3P8, Canada, caroemond@hotmail.com) and Marty Laforest (Lettres et Commun. sociale, Université du Québec à Trois-Rivières, Trois-Rivières, QC, Canada)

Smiling is a visible expression and an audible one too when it is synchronous with speech. Very few studies have documented the perceptual prosodic cues associated with perceived smiling speech. The first aim of this paper is to study the perception of smiled-speech according to the listeners' gender. The reaction time and the intensity of the perceived smiled-speech were also investigated. The second aim is to identify a combination of prosodic parameters which would allow a phonetic description of smiled-speech. 140 utterances were extracted from spontaneous data (Montréal 1995 corpus) and used as stimuli for a perception test administered to 40 Québec French listeners (20 men, 20 women). Results show that men and women do not perceived smiled-speech in the same way, and women are quicker than men to make their decisions. Moreover, reaction times are faster for utterances perceived as smiling with a high degree of intensity, for both men and women, than those with lower intensity. Perceived prosodic parameters related to pitch height, pitch range, rhythm, and speech rate in relation to smiled-speech and its intensity are also discussed.

4pSCb40. Vowel production in Mandarin accented English and American English: Kinematic and acoustic data from the Marquette University Mandarin accented English corpus. An Ji (Elec. & Comput. Eng., Marquette Univ., Milwaukee, WI), Jeffrey J. Berry (Speech Pathol. & Audiol., Marquette Univ., P.O. Box 1881, Milwaukee, WI 53201-1881, jeffrey.berry@marquette.edu), and Michael T. Johnson (Elec. & Comput. Eng., Marquette Univ., Milwaukee, WI)

Few electromagnetic articulography (EMA) datasets are publicly available, and none have focused systematically on non-native accented speech. We introduce a kinematic-acoustic database of speech from 40 (gender and dialect balanced) participants producing upper-Midwestern American English (AE) L1 or Mandarin Accented English (MAE) L2 (Beijing or Shanghai dialect base). The Marquette University EMA-MAE corpus will be released publicly to help

advance research in areas such as pronunciation modeling, acoustic-articulatory inversion, L1-L2 comparisons, pronunciation error detection, and accent modification training. EMA data were collected at a 400 Hz sampling rate with synchronous audio using the NDI Wave System. Articulatory sensors were placed on the midsagittal lips, lower incisors, and tongue blade and dorsum, as well as on the lip corner and lateral tongue body. Sensors provide five degree-of-freedom measurements including three-dimensional sensor position and two-dimensional orientation (pitch and roll). In the current work we analyze kinematic and acoustic variability between L1 and L2 vowels. We address the hypothesis that MAE is characterized by larger differences in the articulation of back vowels than front vowels and smaller vowel spaces compared to AE. The current results provide a seminal comparison of the kinematics and acoustics of vowel production between MAE and AE speakers.

4pSCb41. Phonetic alignment and phonological association in Tashlhiyt Berber. Timo B. Röttger (IfL Phonetik, Univ. of Cologne, Herbert-Levin-Str. 6, Köln D-50931, Germany, timo.roettger@uni-koeln.de), Rachid Ridouane (Laboratoire de Phonétique et Phonologie (UMR 7018), CNRS/Sorbonne Nouvelle, Paris, France), and Martine Grice (IfL Phonetik, Univ. of Cologne, Köln, Germany)

Although Tashlhiyt Berber uses intonation to mark sentence modality, the location of f0 events is severely constrained by its notorious predominance of consonantal nuclei (cf. (1) where syllable nuclei are underlined) (1) [ts.sk.ʃ f.tstt] "you dried it (fem.)" Here we report on the alignment of f0 peaks in disyllabic target words in polar questions and contrastive statements in the language. Data from four native speakers revealed that questions tend to have later f0 peaks than statements. This was reflected in discrete association patterns when more than one tone bearing unit was available: in questions the f0 peak occurred significantly more often on the final syllable than in statements. Interestingly, if no association distinction was made, there was a difference in alignment of this peak within a tone bearing unit: the peak was aligned significantly later in questions. Thus, discrete phonological association patterns were mirrored by phonetic alignment detail. These data question the traditional dichotomy between phonological association and phonetic alignment.

4pSCb42. Articulatory parameterization in Trique tone production: Distinguishing co-production from coarticulation. Christian DiCanio and Hosung Nam (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, dicanio@haskins.yale.edu)

The production of a tone in a tonal language is typically influenced by adjacent tonal targets. Within the literature, all such influences on F0 are considered part of tonal coarticulation. Yet, conflating all these effects under "coarticulation" results in an assortment of different processes within a tone language which lack a common motivating principle. In this talk, we present original tone production data from Itunyoso Trique (Oto-Manguean). The data consists of five repetitions of 24 sentences spoken at two speech rates (fast/normal) by eight native speakers. The medial target word was one of four tones (/45/,/4/,/32/,/2/), while the adjacent words were one of six tones (/45/,/43/,/32/,/3/,/2/,/1/). F0 data was extracted and time-normalized. Two patterns were observed. First, adjacent tones influenced F0 at the onset and offset of target tones. Second, global changes in F0 contour occurred for certain tones. All such effects were stronger during fast speech rate. We argue that these effects, often grouped together as coarticulation, have distinct explanations within an Articulatory Phonology framework. Transitional effects at tonal onsets and offsets are modeled by temporally modulating gestural activation intervals, resulting in articulatory undershoot between tones, whereas global changes in F0 contour are modeled by modulating gestural target parameters.

4pSCb43. Perception modeling of native and foreign-accented Japanese speech based on prosodic features of pitch accent. Ashleigh R. Gonzales (Linguistics, Simon Fraser Univ., 8888 University Dr., 9201 Robert C. Brown Bldg., Burnaby, BC V5A 1S6, Canada, agonzale@sfu.ca), Shunichi Ishihara (School of Culture, History and Lang., Australian National Univ., Canberra, ACT, Australia), and Chiharu Tsurutani (School of Lang. and Linguist, Griffith Univ., Nathan, QLD, Australia)

This study investigates the influence acoustic measures of pitch accent have on L1 and Australian English (AusE) L2 Japanese speech perception, expanding Tsurutani (2010) and Ishihara, Tsurutani, and Tsukada (2011), and motivated by Munro and Derwing (2001), which studies the role of speaking rate on judgments of L2 speech. We establish native and advanced AusE listeners of Japanese differ in their judgments of foreign accent in terms of accentedness and comprehensibility [Munro and Derwing (1995, 1999)] through a listening task. Selected acoustic measures of pitch accent from the speech stimuli, which displayed significant variance across listener groups—delta-pitch, max and mean max delta-intensity, and duration per mora—are correlated with L1 and L2 listener data. Testing for a relationship between each of the acoustic measures and listener judgments, the regression analyses show a considerable relationship between comprehensibility judgments and duration and intensity features, ranging from adjusted $R2 = 14.3\%$ to 24.6% across listeners, and indicating the degree of variance between judgments can be attributed to these acoustic measures. We can interpret that comprehensibility is linked to intensity and duration, which supports the authors' prior findings that timing is considered more important than pitch in the detection of foreign-accented speech.

4pSCb44. Can co-speech hand gestures facilitate learning of non-native tones? Katelyn Eng, Beverly Hannah, Lindsay Leung, and Yue Wang (Linguistics, Simon Fraser Univ., 8888 University Dr., RCB 9213, Burnaby, BC V5A 1S6, Canada, yuew@sfu.ca)

Speech perception research has indicated that information from multiple input modalities (e.g., auditory, visual) facilitates second language (L2) speech learning. However, co-speech gestural information has shown mixed results. While L2 learners may benefit from this additional channel of information, it may also be inhibitory as learners may experience excessive cognitive load. This study examines the role of metaphoric hand gestures in L2 lexical tone learning using previously established laboratory training procedures. Training stimuli include Mandarin tones produced by native Mandarin speakers, with concurrent hand gestures mimicking pitch contours in space. Native Canadian English speakers are trained to perceive tones presented in one of three modalities: audio-visual (AV, speaker voice and face), audio-gesture (AG, speaker voice and hand gestures) and audio-visual-gesture (AVG). The effects of training are assessed by comparing the pre-training and post-training tone identification results. Greater improvements for the AVG compared to AV group would indicate the facilitative role of gestures. However, greater improvements for the AG or AV compared to AVG group would support the cognitive overload account. Findings are discussed in terms of how sensory-motor and cognitive domains cooperate functionally in speech perception and learning. [Equal contributions by KE, BH, and YW; work supported by SSHRC.]

4pSCb45. Consonant harmony in Moroccan Arabic: Similarity and incomplete neutralization. Georgia Zellou (Linguistics, Univ. of Pennsylvania, 800 N. 48th St., #26, Philadelphia, PA 19139, gzellou@sas.upenn.edu)

Moroccan Arabic (MA) displays a synchronic consonant harmony alternation where underlying alveolar sibilants can assimilate in place of articulation to a following palatal sibilant, e.g., seʒera ~ ʃeʒera "tree". This study investigates the phonetic realization of the assimilated sibilant variant in consonant harmony forms. This consonant harmony process is typologically unusual since avoidance of similarity of root consonants has been proposed to be a pervasive tendency for the Semitic languages, including Arabic. Hence, it is predicted that even though a phonological change has resulted in adjacent stem consonants with identical features, similarity avoidance tendencies will act at the level of the phonetic representation to ensure that adjacent consonants are not articulatorily identical. An acoustic investigation using a center of gravity (COG) measure of MA sibilants was conducted on monolingual MA speakers to test this hypothesis. The results indicate that the harmonized palatal sibilants (i.e., ʃeʒera) are produced with a higher COG, suggesting a further front place of articulation, compared to regular (non-harmonized) palatal sibilants. In other words, the harmonized sibilants exemplify a case of incomplete neutralization, where the phonetic trace of a disappeared consonant remains. Furthermore, these results suggest that similarity avoidance in MA is maintained through sub-phonemic, gradient differences.

4pSCb46. The role of prosody in speech segmentation: Comparisons between monolinguals and French-English bilinguals. Meghan Spring, Linda Polka (School of Commun. Sci. and Disord., McGill Univ., 46 Tiffany Crescent, Kanata, Ontario K2K1W2, Canada, meghan.spring@mail.mcgill.ca), and Suzanne Curtin (Dept. of Psych., Univ. of Calgary, Calgary, AB, Canada)

Monolinguals harness language-specific prosodic cues for the purpose of segmenting out words from the speech stream. However, if and how bilinguals are able to do so in both their languages is less certain. In the current

study, 26 English monolinguals, 28 French monolinguals, and 41 English-French adult bilinguals heard streams of both English- and French- accented nonsense syllables. While there were clear differences between the monolingual English and French groups, there was no difference between the performance of English-dominant and French- dominant bilinguals, nor between simultaneous versus sequential bilinguals. As a group, English-French bilinguals did show evidence of different segmentation strategies between language streams. It is therefore concluded that in certain conditions, bilinguals appear to be able to switch stress-based segmentation strategies between their languages. The use of the Hearing in Noise Test (HINT) as a promising new method for measuring language dominance in bilinguals is also discussed.

4pSCb47. Perceived prosodic boundaries in Taiwanese and Swedish. Grace Kuo (Linguistics, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095, gracekuo@humnet.ucla.edu)

Earlier studies have shown that listeners are not only able to detect the presence or the absence of a prosodic boundary but also able to distinguish between different boundary types. This study examined whether Taiwanese listeners ($n = 18$) and English listeners ($n = 7$) were able to predict the occurrence and the strength of the upcoming prosodic boundaries in Taiwanese and Swedish. For this purpose, we conducted a perceptual rating experiment, whose stimuli consisted of fragments with different boundaries (word, phonological phrase/tone sandhi domain, and intonational phrase), length (2-second and one-word) and quality (low-pass filtered and unfiltered.) Results show that both Taiwanese and English listeners can detect the occurrence and distinguish the boundaries in a foreign language when they are presented with longer fragments. Our finding strengthens the notion proposed in Carlson *et al*. (2005) that lexical information is not a necessary cue for prosodic boundary detection Another supporting evidence is that they could do the task nearly as well when the utterances were low-pass filtered. Significant correlations between ratings and the following relevant measures are found: f0 and voice quality.

4pSCb48. Decrease of pitch perception ambiguity in tone language processing. Xiao Perdereau (Burgundy Univ., 9, A. Savary, BP 47870, Dijon 21078, France, xiao.chen-perdereau@u-bourgogne.fr)

Native tone language speakers were presented with speech materials in their language produced by non-native speakers. The speech materials were selected sound streams according to acoustic characteristics. They were made of monosyllabic words, disyllabic words and polysyllabic short sentences in spoken Mandarin. Participants were required to recognize the speeches in as short time as possible. Results revealed that the essential time to identify the speech is longer for shorter sequences, suggesting that ambiguities reside mostly in the lexical tone level. The ambiguity due to pitch perception decreases when the segmented speech events increase. Although it contributes to word meaning, pitch perception is less important in a polysyllables group of words or sentence processing than in monosyllabic word identification. We will also present some applications of these findings.

4pSCb49. Towards a model of intonational phonology of Turkish: Neutral intonation. Canan Ipek and Sun-Ah Jun (Linguistics, USC, 123 S Figueroa Apt. 835, Los Angeles, CA 90012, canan.ipek@gmail.com)

This study proposes an Autosegmental-Metrical model of Turkish intonation based on sentences produced in neutral focus, as part of our ongoing research investigating Turkish intonational phonology. Tonal patterns of utterances were examined by varying the length of a word and a phrase, the location of stress, syntactic structures, and sentence types. Preliminary results suggest that Turkish has a H* pitch accent, realized on the stressed syllable of most content words. Each content word forms one Prosodic Word (PW) whose left edge is marked by an L tone. There are two prosodic units higher than PW: an Intermediate Phrase (ip) marked by a final rising (LH) tone and an Intonational Phrase (IP) marked by various types of a final boundary tone. These three prosodic units are also distinguished by the degree of juncture. Interestingly, the ip-final LH boundary tone marks the right edge of a heavy syntactic constituent regardless of the length of the unit. Furthermore, the left edge of a nuclear pitch accent is also marked by a rising tone (LH), which is realized on the last syllable of the immediately preceding PW. The ip-final LH tone and the pre-nuclear LH tone are phonetically different and perceptually distinct.

4pSCb50. Fundamental frequency as cue to intonation: Focus on Ika Igbo and English rising intonation patterns. Joy O. Uguru (Linguist, Igbo and other Nigerian Lang., Univ. of Nigeria, No. 01 Louis Mbanefo st., Nsuukka +234, Nigeria, joyolug@yahoo.com)

This paper shows that fundamental frequency, Fo, can be a cue to type of intonation. The work centers on three main intonation patterns in Ika Igbo and English. Ika Igbo is a language that manifests intonation in addition to lexical tone. These intonation patterns are Low Rise (LR), High Rise (LR), and Fall Rise (FR). The Fos of these intonation patterns were analyzed acoustically in utterances with similar phonemes and tunes in both languages. Eighteen utterances were used for the study. The analyses show that the Fos of LR and FR intonation were generally lower than those of HR. Hence, it can be concluded that high intonation has high Fo while low intonation has low Fo. It can therefore be concluded that Fo is a cue to type of intonation.

4pSCb51. Downstep exceptions in Ibibio. Afton L. Coombs (Linguistics, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, acoombs@usc.edu)

Downdrift and downstep are processes which may cause lowering of high tone syllables. Downdrift is intonational, occurring at phrasal or utterance level, while downstep is a phonological process which acts from one tone-bearing unit to the next such that H tones lower successively. The relationship between these larger tonal lowering processes and individual tone units is complicated, however, by processes which may raise or preserve original high tone pitches. Ibibio, a Niger-Congo language spoken in southeastern Nigeria, is a terraced tone language with contrastive H and L tones. H tones in Ibibio experience automatic and non-automatic downstep, lowering both in sequences of high tones and around intervening lows. This study aims to determine those factors which counteract or overrule the downstep process. Average pitch readings were taken of entire syllables and compared with readings of other syllables within the same word. The main finding of this study is that while single words show acoustically measurable downtrends, they also show non-lowering and even raising of high tones, specifically in HHL contexts. This complicates how downtrends act across tone-bearing syllables, and may indicate that a high tone is raised in order to increase contrast with a following low.

4pSCb52. Effect of speech variable rate on the coarticulation in the right vocalic context of Arabic utterances VCV. Leila Falek, Hocine Teffahi (Electron. and Comput. Sci. Faculty, USTHB FEI Algiers, USTHB FEI Algiers Algieria, Algiers 16111, Algeria, falek.leila@gmail.com), and Amar Djeradi (faculté d'electronique, USTHB, Algiers, Algeria)

Our study consists of analysing Arabic utterances VCVα in brief vocalic context with Vα and speech rate as variables in order to observe the impact of the "right" context and speech rate on the coarticulation. Thus, we have to look for some invariance in the speech signal explaining the coarticulation phenomenon related to speech rate. So, we have analyzed the formant tracking of the arabic pharyngeal / / (or (ع) in arabic) in vocalises contexts /a/, /u/ and /i/ with variables speech rates (normal, fast and slow) in interrogative sentences. That is in order to confirm the anticipatory phenomenon and observed the influence of speech rate variation on the vocal tract in articulation. The observed results have shown in our case the existence of anticipatory articulation and that it depends on speech rate. (Non existence in slow rate, however, with more prominence in normal or fast rate.)

4pSCb53. Acoustic features of English sentence production for English and Chinese native speakers: Intonational and temporal patterns. Ashley Woodall, Chang Liu, Brenna Thomas, and Katherine Reistroffer (The Univ. of Texas at Austin, 2504 A Whitis Ave A1100, Austin, TX 78712, ash.woodall@gmail.com)

Intonation is utilized by languages in order to differently convey intention, meaning, and emotion. Chinese, a tonal language, assigns F0 formant frequencies to lexically important components within words; whereas English, a stress language, changes F0 patterns according to the speaker's intended meaning or emotion, especially changing near the end of a phrase. As Chinese speakers produce other languages, especially a non-tonal language such as English, it is uncertain whether their intonation is the same as a native speaker. The purpose of this study is to compare intonation across Chinese and English speakers. An acoustical analysis was completed on 16

English and 32 Chinese speakers producing List 1 of the Hearing In Noise Test (HINT) sentences. Preliminary results show that Chinese speakers produce sentences with longer absolute duration than English speakers. Intonation features of sentence production such as F0 contour and temporal contours, as well as temporal features of English sentences such as temporal gap and sentence and word duration will also be compared. Findings will show the effect of L1 tonal language on producing a stress language, such as English.

4pSCb54. Rate variation as a talker-specific/language-general property in bilingual speakers. Midam Kim (Linguistics, Northwestern Univ., 425 Hurricane Ln, Lawrence, Kansas 66049, midamkim@gmail.com), Lauren Ackerman, L. Ann Burchfield, Lisa Dawdy-Hesterberg, Jenna Luque, Kelsey Mok, and Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL)

Nonnative talkers tend to exhibit slower speech rates than native talkers at the group level. Here we ask whether individual variation in rate is language-general to the extent that L1 rate is a significant predictor of L2 rate within bilinguals. 62 nonnative English talkers participated in three speech production tasks in both their L1 (14 Cantonese, 14 Mandarin, 11 Korean, 4 Portuguese-Brazilian, 6 Spanish, 13 Turkish) and L2 (English), namely, reading a paragraph, spontaneously answering questions, and spontaneously describing a picture story. Two measurements of rate were automatically extracted from the recordings: speech rate (syllables per second), and articulation rate (syllables per second excluding silent pauses). As expected, L2 speech and articulation rates were overall slower than L1 speech and articulation rates for all tasks. Importantly, L2 speech rates and articulation rates were positively related to L1 speech rates and articulation rates, respectively. There were also significant differences in L2 speech rates and L2 articulation rates depending on L1 background and tasks. However, the positive relationship between L1 and L2 rates still holds with these other effects taken into consideration, suggesting that overall rate variation is partially an individual-specific property that transcends L1 and L2 within bilinguals. Acknowledgments: Vanessa Dopker and Chun Liang Chan. [Work supported by Grant R01-DC005794 from NIH-NIDCD.]

THURSDAY AFTERNOON, 6 JUNE 2013 — 510A, 1:00 P.M. TO 5:00 P.M.

Session 4pSP

Signal Processing in Acoustics, Acoustical Oceanography, and Architectural Acoustics: Sampling Methods for Bayesian Analysis and Inversions in Acoustic Applications

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

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180

Invited Papers

1:00

4pSP1. Sampling methods for uncertainty quantification in source localization and geoacoustic inversion in the ocean. Tao Lin and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd, Newark, NJ 07102, michalop@njit.edu)

Iterative and sequential Bayesian filtering approaches have been successfully employed for the estimation of select features of received acoustic signals—namely, arrival times and amplitudes of paths that have interacted with the propagation medium. These are subsequently utilized in source localization and environmental property estimation. Sequential filtering has the advantage of relating arrival times across spatially separated hydrophones of a receiving array, providing “tighter” estimates of arrival times and amplitudes and, thus, probability densities with a reduced “spread” in inversion. We demonstrate that sequential methods are superior to solely iterative ones by linking estimates of times and amplitudes to propagation models and estimating source location and environmental parameters. The inversion component of the problem is approached with an efficient approach, which relies on a novel implementation of linearization of the relationship that links parameters of the propagation medium to the received sound field. [Work supported by ONR].

1:20

4pSP2. Bayesian localization of an unknown number of ocean acoustic sources. Stan E. Dosso (School of Earth & Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

This paper considers localizing an unknown number of ocean acoustic sources when properties of the environment are poorly known. A Bayesian formulation is developed in which environmental parameters, noise statistics, and the number, locations, and complex spectra (amplitudes and phases) of multiple sources are considered unknown random variables constrained by acoustic data and prior information. The number of sources is determined during a burn-in stage by minimizing the Bayesian information criterion using hybrid optimization with an efficient source birth/death scheme. Optimal estimates and marginal posterior probability distributions for source locations are computed employing a variety of sampling approaches. Environmental properties and source locations and are treated as explicit parameters and marginalized using Markov-chain Monte Carlo sampling methods. In particular, environmental parameters are treated using Metropolis-Hastings sampling applied efficiently in a principal-component space, and source locations are treated using Gibbs sampling since the corresponding conditional probability distributions can be computed efficiently using normal-mode methods. Source and noise spectra are sampled implicitly by applying analytic maximum-likelihood solutions expressed in terms of the explicit parameters. This represents an empirical Bayesian approximation within a hierarchical formulation, and significantly reduces the dimensionality and improves sampling efficiency in the inversion.

1:40

4pSP3. Probabilistic two dimensional joint water-column and seabed inversion. Jan Dettmer and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8W 3P6, Canada, jand@uvic.ca)

This paper develops a probabilistic two-dimensional (2D) inversion for geoacoustic seabed and water-column parameters in a strongly range-dependent environment. Range-dependent environments in shelf and shelf-break regions are of increasing importance to the acoustical-oceanography community, and recent advances in nonlinear inverse theory and sampling methods are applied here for efficient probabilistic inversion in 2D. The 2D seabed and water column are parameterized by highly efficient, self-adapting irregular grids which match the local resolving power of the data and provide parsimonious solutions requiring few parameters to capture complex environments. The self-adapting parameterization in the water-column and seabed is achieved by implementing the irregular grid as a trans-dimensional hierarchical Bayesian model which is sampled with the Metropolis-Hastings-Green algorithm. To improve sampling, population Monte Carlo is applied with a large number of interacting parallel Markov chains employing a loadbalancing algorithm on a computer cluster. The inversion is applied to simulated data for a vertical line array and several source locations to several kilometers range. Complex pressure fields are computed using a parabolic equation model and results are considered in terms of 2D ensemble parameter estimates and marginal uncertainty distributions. [Work supported by NSERC.]

2:00

4pSP4. Using nested sampling with Galilean Monte Carlo for model comparison problems in acoustics. Paul Goggans, Wesley Henderson (Elec. Eng., Univ. of Mississippi, Anderson Hall Rm. 302, University, MS 38677, goggans@olemiss.edu), and Ning Xiang (Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Nested sampling is increasingly being used to calculate the evidence for competing models in Bayesian model comparison problems arising in acoustics applications. Use of nested sampling offers advantages in robustness over alternative methods of evidence calculation and enables evidence calculation for models with many parameters. The most challenging aspect of implementing nested sampling is sampling from the prior for the parameters constrained by a threshold likelihood value. For models with just a few parameters, sampling from the constrained prior can be accomplished with a simple Monte Carlo algorithm implementing a random walk, however, this simple method is inefficient and fails as the number of model parameters increases. John Skilling, the originator of nested sampling, has proposed the “Galilean” Monte Carlo method for efficiently sampling from the constrained prior when there are many parameters. Unlike the random walk method, the Galilean Monte Carlo method moves samples with a vector velocity, reflecting them from the likelihood constraint surface when necessary. This directed sampling gives the method its greater efficiency. In this paper we discuss our experience in implementing Galilean Monte Carlo in nested sampling and compare Galilean and random walk Monte Carlo for a model comparison problem in room acoustics.

2:20

4pSP5. Energy based Markov Chain Monte Carlo algorithms for Bayesian model selection. Tomislav Jasa (Thalgorithm Inc., 1688 Tarn Rd., Toronto, ON l4x1b1, Canada, jasa@ini.phys.ethz.ch), Jonathan Botts, and Xiang Ning (Grad. Program in Architectural Acoust., Rennselear Polytechnic Inst., Troy, NY)

Markov Chain Monte Carlo (MCMC) algorithms for Bayesian model selection have been increasingly applied to acoustics applications. One of challenging tasks required in Bayesian model selection is the exploration of high-dimensional multi-variate spaces such that a key quantity, termed the Bayesian evidence, can be estimated in order to rank a set of competing models. This work presents a class of energy-based MCMC algorithms specifically designed to estimate the Bayesian evidence. As illustrative examples, the energy-based MCMC algorithms are applied to the problem of filter design as used in human head-related transfer functions and in acoustic impedance boundaries within the finite-difference time-domain framework for room-acoustics simulations.

2:40

4pSP6. Nested sampling in practice. Jonathan Botts (Dept. of Media Technol., Aalto Univ., 110 8th St., Greene Bldg., Troy, New York 12180, botts.jonathan@gmail.com)

For problems requiring both model comparison and parameter estimation, nested sampling is an attractive choice because it provides an estimate of the normalized posterior. The critical assumption of nested sampling is that exploration proceeds regularly toward the region of maximum likelihood. However, in practice, achieving such regular compression is far from trivial, particularly for realistic, multi-modal problems. This paper offers a comparison of both systematic and random walk exploration for generating samples within a constrained prior. Ranges for size of ensemble and amount of exploration required for regular compression are also established for different acoustic data analysis problems.

Contributed Papers

3:00

4pSP7. Geoacoustic inversion via trans-dimensional sampling over seabed and error models. Gavin Steininger, Jan Dettmer, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. (Ring Rd.), Victoria, BC V8P 5C2, Canada, gavin.amw.steininger@gmail.com), and Charles W. Holland (Appl. Res. Lab., The Penn State Univ. Univ., State College, PA)

This paper develops an efficient Bayesian sampling approach to geoacoustic scattering and reflection inversion based on trans-dimensional (trans-D) sampling over both the seabed model (number of sediment layers) and error model (autoregressive order to represent residual correlation). Sampling is carried out using a population of interacting Markov chains employing a range of sampling temperatures (parallel tempering). The approach is applied to both simulated and measured data. The advantages of trans-D autoregressive model sampling over alternative methods of error model selection is explored in terms of the reduction in posterior uncertainty of geoacoustic parameters and evaluation of residual correlation. The seabed is modeled as a stack of homogeneous fluid sediment layers overlying an elastic basement. Including elastic (shear) parameters in the basement makes this layer distinct from the overlying sediment layers and requires a novel formulation of the partition prior distribution for trans-D sampling. [Work supported by ONR.]

3:20

4pSP8. Bayesian-based estimation of acoustic surface impedance: finite difference frequency domain approach. Alexander Bockman (Massachusetts Inst. of Technol. Lincoln Lab., 244 Wood St., Lexington, MA 02420, alexander.bockman@ll.mit.edu), Cameron Fackler, and Ning Xiang (Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Design for acoustic performance in an interior fluid domain requires accurate description of boundary materials' specific acoustic impedance. The standard approach for the estimation of this material characteristic is the two-microphone, impedance-tube method. Modifications to the processing of the sampled acoustic field have been proposed to allow for more general test geometries. While analytical methods may be applied to a small class of ideal geometries, numerical methods provide greater geometric flexibility. In general, solutions to the wave equation forward problem are found from boundary element, finite element, or finite difference methods. The inverse problem of parameter estimation is solved by evaluating accuracy of prediction of the acoustic field for given distributions of the specific acoustic impedance parameter against observed data. In this presentation a Bayesian-network sampling approach is used to estimate specific acoustic impedance of a micro-perforated panel in an impedance tube test geometry. The choice of geometry and material allow for direct comparison to the two-microphone, impedance-tube method within the appropriate frequency range, and a theoretical model for the material beyond that frequency range. The potential to extend the frequency range of operation of the impedance tube is explored. Sensitivity of the method to nuisance parameters is discussed.

3:40

4pSP9. A Bayesian based equivalent sound source model for a military jet aircraft. David M. Hart (Physics, Brigham Young Univ., 363 N. 835 E., Lindon, UT 84042, dmh1993@studentbody.byu.edu), Tracianne B. Neilsen, Kent L. Gee (Physics, Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

The two-source model for jet noise holds that turbulent mixing noise in jets is generated by uncorrelated, fine-scale (FSS) and partially correlated, large-scale (LSS) turbulent structures [Tam *et al*., J. Fluid Mech. **615**, 253–292, (2008)]. The noise from an F-22A Raptor is modeled with an equivalent source consisting of two line arrays of monopole sources. These arrays, one correlated and one uncorrelated, with Rayleigh distributed amplitudes, account for both FSS and LSS sound propagation [Morgan, J. Acoust. Soc. Am. **129**, 2442 (2011)]. The equivalent source parameters are selected based on Bayesian methods implemented with simulated annealing and fast Gibbs sampler algorithms. This method yields the best fit parameters, and the sensitivity of the solution is indicated by the generated posterior probability distributions. Analysis of the resulting equivalent sources shows that the directional, correlated line array has a greater effect on the near field sound, and the sensitivity of the array's parameters increases as the frequency increases. This equivalent source model can generate results up to 2500 Hz and accurately predict both near field and far field measurements. The analysis suggests that the shape of the source distribution changes as the frequency increases. [Work sponsored by the Office of Naval Research.]

4:00

4pSP10. Identification of acoustic sources with uncertain data. Vincent Martin and Frédéric Cohen-Tenoudji (Institut Jean Le Rond d'Alembert, UMR CNRS/UPMC 7190, 4 Place Jussieu, Paris 75252 Paris Cedex 05, France, vincent.martin@upmc.fr)

In inverse acoustic problems where attempting to identify the vibratory velocities of sources at the origin of an acoustic radiated field, we have the measured radiated field (called objective) on an antenna with numerous sensors and a propagation model. If both are erroneous, misidentification follows. Here, the problem is formulated in the frequency domain and solved in the least mean square sense. An impaired objective including an unstructured error has virtually no chance of satisfying the propagation equation. Accordingly, with an accurate radiation model, we cannot identify source velocities able to generating this objective. With the same model but now with unknown parameters (in the case of only one parameter it could be the speed of sound within the medium), it is expected intuitively that the parameter value aiming at the perturbed objective does not reach it but ultimately generates a pressure satisfying the wave equation, with a value near the correct pressure. The error in the model is structured in the sense that the model keeps a form satisfying the equations of physics. Currently, it is reported that this intuitive expectation is observed quantitatively through the geometric interpretation of over-determined inverse problems dealt with in L2.

4:20

4pSP11. Nested sampling-based design of multilayer microperforated panel sound absorbers. Cameron Fackler and Ning Xiang (Grad. Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, facklc@rpi.edu)

A model-based design approach for microperforated panel absorbers comprised of multiple panel layers is developed. Microperforated panels (MPPs) are becoming increasingly popular as sound absorbers, capable of providing broadband absorption with high absorption coefficients, without the use of traditional porous materials. To increase the bandwidth of the intrinsically peaked narrowband absorption of a single MPP, multiple such panels can be combined into composite sound absorbers. We propose a method based on Bayesian inference to design multilayered MPP absorbers capable of producing a user-specified absorption profile. Using nested sampling to accumulate Bayesian evidence and to implement Occam's razor, the method produces a design requiring the fewest number of MPP layers while meeting the specified design requirements.

4:40

4pSP12. Assessing model uncertainties for joint inversions of seismological data using a genetic algorithm. Priscilla Brownlow (Grad. Program in Acoust., Penn State Univ. Univ., 307B Dunham Hall, White Course Apartments, University Park, PA 16802, pdb153@psu.edu), Richard Brazier, Andrew Nyblade, Jordi Julia, and K. B. Boomer (Dept. of Geosciences, Penn State Univ., University Park, PA)

Error bars were generated for velocity models using receiver functions and surface wave dispersion curves for four seismic stations in southern Africa, with a genetic algorithm adapted from the code NSGA-II. Each receiver function and dispersion curve was originally created by Eldridge Kgaswane (2009). We examined these stations, and through a series of statistical resampling, we were able to place an uncertainty on each layer's velocity in the lithosphere. Each station was set to an initial model, which was perturbed to generate a series of best-fit models for the corresponding receiver functions and dispersion curves. For each layer of depth, a series of solutions evolve over a set number of generations using "survival of the fittest" to come up with these best-fit models. These were constrained to only consider geologically viable models, such as the velocity range in each layer and smoothing. Afterward, the error bounds on velocities were able to be placed on each layer. The velocity vs. depth plot gives the uncertainty from 1 to 2.5 km in depth. Now a better estimate of the velocities of the waves can be made, which leads to a better estimate of the composition of the lithosphere under southern Africa.

Session 4pUW

Underwater Acoustics and Signal Processing in Acoustics: Sparse Process Modeling Techniques for Acoustic Signal Processing

Paul J. Gendron, Cochair
Maritime Systems Div., SSC Pacific, A460, Bldg. 1, Bayside Campus, 53560 Hull St., San Diego, CA 92152

Geoffrey F. Edelmann, Cochair
U. S. Naval Res. Lab., 4555 Overlook Ave SW, Code 7145, Washington, DC 20375

Chair's Introduction—12:55

Invited Papers

1:00

4pUW1. Reweighted sparse source-location acoustic mapping in shallow water. Pedro A. Forero and Paul A. Baxley (Maritime Systems Div., SSC Pacific, 53560 Hull St., BS/160/Rm. 190, San Diego, CA 92152, pedro.a.forero@navy.mil)

Various applications for monitoring and surveillance in littoral waters rely on passive sonar for localizing acoustic sources in shallow-water environments. Although adaptive matched-field processing (MFP) has been successfully used for localization, its performance is degraded when localizing multiple sources at low signal-to-noise ratios (SNRs) and in the presence of model mismatch. Robust MFP using, e.g., the white-noise constraint offers an alternative to cope with the mismatch issue but remains ineffective in the multisource and low SNR setup. This work capitalizes on sparsity for constructing a source location map for shallow water environments. Sparsity naturally arises since only locations corresponding to acoustic sources are expected to appear in the map (nonzero entries), while the remaining map locations are empty (zero entries). A high-resolution map is constructed via a two-step approach that capitalizes on a model for the acoustic propagation environment while being robust to model mismatch. During the first step the robust map is obtained by solving a regularized least-squares (LS) problem. Then, the map coefficients are used to devise a modified criterion with a weighted regularizer yielding a lower-ambiguity map, facilitating detection of quiet sources in the presence of loud interferers.

1:20

4pUW2. Application of statistical reduced isometry property to design of line arrays for compressive beamforming. Charles F. Gaumond and Geoffrey F. Edelmann (Acoust. Div., Naval Res. Lab., Code 7162, 4555 Overlook Ave. SW, Washington, DC 20375, charles.gaumond@nrl.navy.mil)

The Statistical Reduced Isometry Property (StRIP) and Statistical Null Space Property (SNSP) are presented and reduced to numerical algorithms. These properties are used to predict the utility of a specific subsampled array for use in compressive sensing. Three examples of subsampling an equally spaced array are presented: random, Golomb and Wichmann. The Golomb array uses a Golomb ruler that has no repeated sensor element spacings. The Wichmann array includes at least one of every possible interval of sensor element spacings. The SNSP is shown to be insensitive to subsampling in the type of cases shown. The Golomb array is predicted to have superior performance to the Wichmann for comparable subsampling. The use of these two subsamplings for beamforming using at-sea data from the Five Octave Research Array (FORA) is shown. [Research funded by the Office of Naval Research.]

1:40

4pUW3. Time-frequency localization issues in the context of sparse process modeling. Ananya Sen Gupta (Elec. and Comput. Eng., Univ. of Iowa, 4016 Seamans Ctr. for the Eng. Arts and Sciences, Iowa City, IA 52242, ananya-sengupta@uiowa.edu)

Practical applications in acoustics such as shallow water acoustic communications often involve non-stationary processes that follow a time-varying sparse support. A classic example is acoustic scatter due to multiple reflections at the moving ocean surface and sea bottom, localized in the time-frequency domain as the sparsely distributed and time-varying Delay-Doppler spread function. In this work, we connect time-frequency localization of non-stationary processes with related issues in adaptive sparse sensing in the context of modeling and tracking the Delay-Doppler spread function. To this end, we provide an overview of methodological advances in adaptive sparse sensing techniques, and compare them over experimental field data collected as 15 m depth, 200 m range and moderate to rough sea conditions. We also comment on other adaptive techniques, such as least-squared error minimization algorithms, and discuss their extensions to the sparse sensing domain. We also explore the uncertainty principle underlying time-frequency representations, particularly in terms of how it influences the related and occasionally competing challenges to sparse process modeling: (i) restricted isometry criteria (RIP) for precise sparse reconstruction and (ii) choice of temporal window to localize the non-stationary Delay-Doppler spread function.

2:00

4pUW4. Performance limits of a compressive sensing application to beamforming on a line array. Jeffrey A. Ballard (Appl. Res. Labs., The Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029, ballard@arlut.utexas.edu)

Compressive sensing is a sampling theorem that exploits the sparsity of a signal in a domain Ψ, while being spread out in a sensing domain Φ. For example, a sinusoid time domain signal in Φ can be represented by one non-zero coefficient in the frequency domain Ψ. The time-frequency relationship is similar to the space-angle relationship that exists in underwater acoustics for an array of hydrophones. Wavefront curvatures that are spread out in the space domain can be represented in the angle domain by a sparse vector. This work investigates the performance limits of using compressive sensing to resolve signals in the angle domain, a task usually accomplished by beamforming. For compressive sensing, it has been shown that the performance of recovering a signal is related to the number of measurements, the number of non-zero coefficients, and the dimension of Ψ [Candes and Wakin, IEEE Signal Process. Mag. 21-30 (March 2008)]. Typically, in underwater acoustics, the number of hydrophones and their locations are fixed, so that the performance is found to be dependent on the number of non-zero coefficients (signals in the water) and the dimension of the angle domain (beams). [Work supported by ARL:UT IRD.]

2:20

4pUW5. Sensitivity of co-prime arrays to shape perturbation. Andrew T. Pyzdek and R. Lee Culver (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, atp5120@psu.edu)

Co-prime arrays offer savings in both implementation and computation by reducing the number of array elements. For passive beamforming, a pair of specially-spaced sparse arrays organized as a co-prime array provides unambiguous source bearings through the cancellation of the grating lobes inherent in the pattern response of each array if processed by itself. In the ocean environment, however, towed line arrays are difficult to keep aligned and take on a time-varying shape. Hodgkiss [IEEE JOE (1983)] showed that array shape perturbation can lead to beam broadening, an effect which may interfere with the grating lobe cancellation of co-prime arrays. In the present paper, performance degradation of co-prime sparse arrays are examined under the condition of perturbed array shape. Simulations are used to compare a co-prime array of known element spacing and position to an array with small uncertainties in both element location and interelement spacing along the array. Possible correction methods are examined. [Work sponsored by ONR Undersea Signal Processing.]

2:40

4pUW6. Effects of multipath distortion on sparse signal parameter estimation. Sung-Hoon Byun, Sea-Moon Kim, and Hyun-Taek Choi (MOERI/KORDI, 171 Jang-dong Yeseong-gu, Daejeon 305-343, South Korea, byunsh@kiost.ac)

Shallow underwater acoustic channel is typically characterized as sparse channel and the sparsity has been actively exploited to estimate the channel accurately. However, distortion of the multipath signal components degrades the performance of sparse approximation and the amount of distortion is dependent on specific time-varying channel condition which each multipath encountered during transmission. In this research we measure the signal distortion of multipath components and analyze its impacts on the sparse channel estimation. Especially, we are interested in the effects of the spatial difference of the distortion on sparse approximation of the multichannel receiver data. To this end, we use variety of experimental data sets which have different characteristics of multipath signal distortion and analyze the relation between the amount of distortion and the signal residual obtained from sparse approximation.

3:00

4pUW7. Numerical simulations of compressive beamforming with a vertical line array in the deep ocean. Geoffrey F. Edelmann (Acoust. Div., U. S. Naval Res. Lab., 4555 Overlook Ave SW, Code 7145, Washington, DC 20375, edelmann@nrl.navy.mil), Ian Rooney (Dept. of Elec. Eng., Univ. of Massachusetts at Dartmouth, Dartmouth, MA), and Charles F. Gaumond (Acoust. Div., U. S. Naval Res. Lab., Washington, DC)

Vertical line arrays (VLA) have been previously deployed in the deep ocean to detect surface targets in the far-field. Conventional beamforming is based upon l2 minimization due to the ubiquitous nature of the fast Fourier transform, but l2 minimization is not a unique method for solving the beamforming problem. Using a basis pursuit algorithm, this research project applied l1 minimization to target detection via beamforming. The compressive beamforming technique was shown to produce narrower beams, comparable noise resistance, and a relaxed requirement on the number of elements required to produce bearing-time records. Compressive beamforming was applied to simulated data and successfully detected surface targets in deep water with 30% of the elements of the full array. [Work supported by the Office of Naval Research.]

3:20

4pUW8. A hierarchical mixture model for sparse broadband scattering functions between moving platforms. Paul J. Gendron (Maritime Systems Div., SSC Pacific, A460, Bldg. 1, Bayside Campus, 53560 Hull St., San Diego, CA 92152, paul.gendron@navy.mil)

A hierarchical mixture model is considered for sparse broadband acoustic Green's functions [J. Acoust. Soc. Am. **130**, 2346, Canadian Acoust. **40**(3)]. Such a mixture model can be employed to match arbitrary second order statistics of a channel over time-bandwidth and angle. The model matches these statistics while simultaneously admitting the degree of sparsity necessary to capture propagation between moving platforms through the ocean waveguide. The uppermost stage of the hierarchy is specified by a mean bulk relative platform speed. Conditioned on this is a structured field of beta distributions associated with the probabilities of ensonified paths over beam-Doppler and frequency. The mixture model of the response is built from Bernoulli indicator variables whose probabilities are drawn from the field of betas. Posterior mean and variance are reviewed and used in an underwater acoustic receiver structure to replace Kalman like estimators of response as well as phase looked loop structures for symbol timing. The performance in terms of mean squared error is 10 dB lower than conventional Wiener filtering schemes when the channel response is significantly sparse. Reduction of this margin occurs as either sparsity or SNR is degraded. This degradation in performance is quantified under a range of sparsity constraints associated with the beta variates. [Work supported by the Naval Innovative Science and Engineering Program and the Office of Naval Research.]

3:40

4pUW9. Bayesian sequential sparse sampling. Peter Gerstoft (Scripps Inst. of Oceanogr., UCSD, 9500 Gillman Dr., La Jolla, CA 92093-0238, gerstoft@ucsd.edu) and Christoph Mecklenbrauker (Tech. Univ. of Vienna, Vienna, Austria)

We consider the sequential reconstruction of source waveforms under a sparsity constraint from a Bayesian perspective. We assume that the wave field that is observed by a sensor array is from a spatially sparse source distribution. A spatially weighted Laplace-like prior is assumed for the source distribution and the corresponding weighted LASSO cost function is derived. We demonstrate the sequential sparse sampling using a line array and track the direction of arrival. In a real world example we track a source using a 2D array.

4:00

4pUW10. High resolution beamforming using sparse recovery from sensor array data. Ravi Menon and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr, MC-0238, UC San Diego, La Jolla, CA 92093, rmenon@ucsd.edu)

We consider the problem of adaptive beamforming using fewer snapshots than the number of sensors. Given an array of sensors in an environment and signals impinging on the array in the presence of background noise, it is of practical interest to be able to estimate the direction of arrival and power of the signals using as few snapshots as possible. In a sparse recovery framework, the signal vector is modeled as a sparse vector in the bearing domain. By casting the beamforming operation as an ℓ^1 minimization problem (as opposed to the convention ℓ^2 minimization), the signal vector can be recovered (the sensing matrix must satisfy the restricted isometry property). The results show an improvement over traditional beamforming methods and these are demonstrated using simulations.

THURSDAY EVENING, 6 JUNE 2013 — 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees on the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings beginning at 7:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Thursday are as follows:

Animal Bioacoustics	510b
Noise	511be
Speech Communication	515abc
Underwater Acoustics/Acoustical Oceanography	510d