THURSDAY AFTERNOON, 6 JUNE 2013 513ABC, 1:00 P.M. TO 5:20 P.M.

Session 4pAAa

Architectural Acoustics: Room Acoustics Computer Simulation II

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

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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

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.

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

4p THU. PM

(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
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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

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Wilfried Kausel, Cochair
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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.

4p THU. PM

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
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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
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Torsten Dau, Cochair
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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
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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
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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
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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