Session 5aID

Interdisciplinary: Plenary Lecture: Public Space Acoustics for Information and Safety

Gilles A. Daigle, Chair
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Chair's Introduction—7:55

Invited Paper

8:00

5aID1. Public space acoustics for information and safety. Hideki Tachibana (Chiba Inst. of Technol., Tsudanuma 2-17-1, Narashino, Chiba 275-0016, Japan, pon-t@iis.u-tokyo.ac.jp)

In such public-spaces as railway stations, airport terminal buildings, shopping arcade, etc., intelligible acoustic information is substantially needed as well as visual information. It is not only for announcements at ordinary times but for evacuation information at emergencies. In reality, however, there are many cases where the public address announcements are much deteriorated by reverberation and background noises in the spaces which results in poor aural intelligibility. In order to improve this kind of acoustic problems in public spaces, a comprehensive research is needed not only by building acoustics but also by speech science, electro-acoustics, signal-processing technology, and cognitive psychology. The acoustics research group in Chiba Institute of Technology, Japan, has been investigating this research topic these several years. In this presentation, the outline of this research and some of its outcome are introduced.

FRIDAY MORNING, 7 JUNE 2013 513ABC, 9:00 A.M. TO 12:00 NOON

Session 5aAAa

Architectural Acoustics: Room Acoustics Computer Simulation III

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Contributed Papers

9:00

5aAAa1. Directional sound source modeling by using spherical harmonic functions for finite-difference time-domain analysis. Shinichi Sakamoto (5th Dept., Inst. of Industrial Sci., The Univ. of Tokyo, 4-6-1, Komaba, Meguro-ku, Tokyo 153-8505, Japan, sakamo@iis.u-tokyo.ac.jp) and Risa Takahashi (ONO SOKKI Co., Ltd., Yokohama, Japan)

Directivity of a sound source originates from the source's shape and its size relative to wavelength. Therefore, the directivity varies with a frequency of a sound. In order to precisely simulate a directivity of a source in the finite-difference time-domain analysis, basically, it is necessary to model the shape of the source geometrically in detail. In the case of sources with complex shapes, however, geometrically precise modeling of the source shape requires small size of spatial discretization, and such a fine mesh discretization results in huge computational costs. In this study, applying a basic theory of Fourier analysis in which arbitrary directivity can be constructed by linear combination of spherical harmonic function, the condition of the sound source for the finite-difference time-domain method to reduce computational cost and to enable efficient analysis against a source with complex directivity characteristics is investigated. Spatial distribution of sound pressure of initial condition for respective spherical harmonic function and correction method of spectral characteristics for the finite-difference time-domain analysis are described.

9:20

5aAAa2. The use of equivalent source models for reduced order simulation in room acoustics. Yangfan Liu and J. Stuart Bolton (Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2037, liu278@purdue.edu)

The sound field in a room can be predicted by using the Boundary Element Method if the motion of the source is known along with the impedance of the room surfaces and furnishings: however, these computations are very time consuming, especially at high frequencies. It is known that the total sound field in a room consists of three components: (1) free space source radiation; (2) the incoming sound field reflected from the room surfaces; and (3) the outgoing sound field scattered from the source surface, particularly if it is large: e.g., a flat screen television. For the purpose of fast computation, an equivalent source method (ESM) can be adopted in which two sets of

equivalent sources represent the incoming and scattered sound field components, respectively. When the free space component is known, the parameters of the ESM can be estimated by least squares approximations of the impedance boundary conditions on the room and source surfaces. Accuracy and computational load can be balanced by the choice of source order and the constraints placed on equivalent source locations. Simulation results will be presented to compare the performances of different ESM configurations.

9:40

5aAAa3. New opera house in Astana: A recent opportunity to use a room acoustic scale model. Maria Cairoli, Enrico Moretti (Acoust. Lab, Biobyte, via ampere, 40, Milano, Milano 20131, Italy, mariacairoli@biobyte.net), and Anders Gade (akustik A/S, Gade & Mortensen, Charlottenlund, Denmark)

The acoustic design of the New Opera House in Astana (Kazakhstan) currently under construction was carried out by Biobyte in Milan, Italy (Enrico Moretti and Maria Cairoli) assisted by Gade & Mortensen Akustik, Denmark (Anders Gade). In order to predict the acoustic consequences of the room geometry and decide on details in the design such as diffusion treatment of curved surfaces, it was decided to build a 1:20 scale model, in which several room acoustical parameters were measured. However, the scale model also provided an opportunity to compare the performance of scale model testing and room acoustic predictions by two computer simulation programs, which were also used for predictions in the design process. Therefore, besides information about the adequacy of the diffusion treatment to avoid focusing from concave surfaces (an aspect which is not well described by computer simulation) we obtained data on measurement accuracy—or rather deviations—between the results from the scale model (DIRAC) and the two different room acoustic prediction programs (CATT and ODEON). These results will be presented and discussed in the paper.

10:00

5aAAa4. Analytic expression to derive the image source point between two symmetrical sloping planes. Santiago Ortiz (Noise Control, Ctr. for Appl. Acoust. and Non Destructive Evaluation (CAEND), CSIC, c/Serrano 144, Madrid 28006, Spain, santiago.ortiz@caend.upm-csic.es), U. Peter Svensson (Electron. and Telecommunications, NTNU-Norwegian Univ. of Sci. and Technol., Trondheim, Norway), and Pedro Cobo (Noise Control, Ctr. for Appl. Acoust. and Non Destructive Evaluation (CAEND), CSIC, Madrid, Spain)

The sound field in open cavities with sloping walls can be represented by image sources and edge diffraction components. Sloping walls will generate a systematic structure to the positions of the image source, and explicit expressions for those positions are presented here. Using such explicit expressions, rather than the general image source method, makes it straightforward to study arbitrarily high orders. When two opposite walls form parts of an infinite wedge, then their image sources will be positioned on a cylindrical surface. The addition of a floor is easy, while adding more walls requires an iterative procedure, with explicit visibility tests. First-order diffraction components can then be generated for each image source. Simulation results are compared with experimental measurements in a three dimensional open cavity with sloping walls.

10:20

5aAAa5. Acoustical archaeology—Recreating the soundscape of John Donne's 1622 gunpowder plot sermon at Paul's Cross. Benjamin Markham, Matthew Azevedo (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, mazevedo@acentech.com), and John Wall (Dept. of English, NC State Univ., Raleigh, NC)

Auralization has become a valuable tool to explore the acoustics of spaces and activities that no longer exist. Generally, acoustical archaeology has explored a fairly limited number of sources in a space to determine specific acoustical aspects of the sound of the spaces and to separate intentionally designed acoustical phenomena from the often unintended effects of the architecture. We have expanded this technique to recreate the entire soundscape of a specific event, in this case John Donne's 1622 Gunpowder Plot sermon at Paul's Cross, outside St. Paul's Cathedral in London as it was prior to the fire of 1666. This work augments ambisonic auralization techniques with techniques borrowed from computer-aided music composition and audio production to create an immersive acoustical environment for the purpose of exploring the experience of listeners at many positions in a crowd that can be varied in size in real time. The paper outlines the role of geometric acoustics modeling, real-time convolution, randomized and statistically derived sound event triggers, and other techniques employed to auralize a soundscape that includes the sermon, crowd response, and the ambient sounds of pre-Industrial London.

10:40

5aAAa6. Auralization of municipal public address announcements by applying geometrical sound simulation and multi-channel reproduction techniques. Junichi Mori (Grad. School of Eng., Chiba Inst. of Technol., 2-17-1 Tshudanuma, Narashino-city, Chiba 271-0016, Japan, junichi@acoust.cs.it-chiba.ac.jp), Sakae Yokoyama (Inst. of Industrial Sci., The Univ. of Tokyo, Tokyo, Japan), Fumiaki Satoh, and Hideki Tachibana (Chiba Inst. of Technol., Chiba, Japan)

Municipal public address system (M.P.A. system) for disaster prevention is an important system for supplying information outdoors. The speech intelligibility of the M.P.A. system, however, tends to be deteriorated by long-pass-echoes due to reflections from the surrounding buildings and the sounds from the loudspeakers covering other sub-areas. When designing such a M.P.A. system, effective tools for prediction of outdoor sound propagation are needed. For this aim, the authors have been investigating the application of the sound simulation technique based on geometrical acoustics. In addition, auralization is also desirable to subjectively assess the speech intelligibility of the M.P.A. system in acoustical design. Therefore, the authors are developing a simulation technique in which the sounds calculated by the geometrical simulation are reproduced through a 6-channel reproduction system, by which 3D information can be aurally realized. In this paper, the outline of this simulation method and some examples of its application are presented.

11:00

5aAAa7. Semi-anechoic music recording using multi-microphone technique to simulate orchestra directivity in room acoustic auralizations. Alban A. Bassuet, Terence Caulkins, Joseph Digerness (Acoust., Ove Arup and Partners, 77 Water St., New York, NY 10002, alban.bassuet@arup.com), and Glenn KnicKrehm (R&D, ConstellationCenter, Cambridge, MA)

Over the last two decades, computer generated auralizations have become increasingly relevant to the design process of new performance spaces. Auralizations are based on anechoic material, and simulation realism is often limited by a small repertoire of recordings and the challenges of simulating an orchestra's complex sound radiation characteristics. To address some of these issues, this paper describes a method to record—in semi-anechoic conditions—large musical ensembles for use in computer simulations. The method uses a multi-microphone hemispherical array distributed over an orchestra playing on a hard floor. A ray-tracing program models the orchestra as sectors of a hemisphere corresponding to each microphone location. Microphone signals are then mapped to their respective source sector. The method permits to record and simulate an entire orchestra with its directivity, including the movement of the musicians while playing. The paper describes the influence of the radius and angular resolution of the microphone array; it also investigates different source configurations in the computer model. The method has been implemented with renowned performers and orchestras and has been used for auralizations of several new concert hall design projects.

11:20

5aAAa8. Interoperability Building Information Modeling and acoustical analysis software—A demonstration of a performance hall design process. Sunyoung Kim, Robert C. Coffeen, and Paola Sanguinetti (Architecture, Univ. of Kansas, 1730 Bagley Dr. #6, Lawrence, KS 66044, sunyoungkim@ku.edu)

By sharing and managing the database for a building model, Building Information Modeling (BIM) facilitates the design process at less cost. Some of BIM software has capabilities for acoustical analysis, but it is limited to noise level demonstration of MEP system in a building. Sophisticated acoustical analysis programs, such as EASE and CATT, provide the useful parameters to determine the sound quality of a space with graphs, A/V systems, material database, and auralization. This research explores how BIM with discipline-specific data reinforced with acoustical information will be beneficial to both architects and acousticians, by allowing them to communicate efficiently. The design process for a performance hall is used to demonstrate how to interoperate between BIM system and acoustical analysis programs. Revit Architecture and EASE program are used as design and analysis tools for for the project. A

model is developed to improve the process of embedding acoustic information in BIM and sharing knowledge in an integrated delivery process.

11:40

5aAAa9. The system (software) for acoustic and lighting calculation (SCAL) program like support for acoustic and lighting conditioning in interior spaces. Beatriz S. Garzón (AAII, IAA, FAU-SeCyT, UNT-CONICET, Av. Roca 1900, Crisóstomo Alvarez 722, San Miguel de Tucumán, Tucumán 4000, Argentina, bgarzon@gmail.com), Carlos R. Coria, and Maria Josefina Manson (FAU-SeCyT, UNT, San Miguel de Tucumán, Tucumán, Argentina)

This work aims to show an alternative software tool for calculating the acoustic and lighting conditioning and comfort in interior spaces. This program is developed in a desktop version and it can run on different operating systems because it is multiplatform. It is easy to operate. Program tools allow a quick calculation because the system values are stored standards of acoustics and lighting through an intuitive interface. The calculations can be stored and open for future modification; also, the results can be summarized and printed. The activities for its development and validation were as follows: (1) Survey on the requirements and interest that the system should contain. (2) Implementation of the program: It was conducted jointly between subject architecture specialists and computer engineers. (3) Development of training materials for the understanding of the functioning of the application. 4) Testing and evaluation period: The functional and non-functional requirements were evaluated and verify if the obtained results would be the expected results.

FRIDAY MORNING, 7 JUNE 2013 — 513DEF, 9:00 A.M. TO 11:40 A.M.

Session 5aAAb

Architectural Acoustics: Effects of Room Boundaries on Diffusion and Reverberation

Isabelle Schmich-Yamane, Chair
CSTB, 24 rue Joseph Fourier, Grenoble 38000, France

Contributed Papers

9:00

5aAAb1. Energy decay analysis of non-diffuse sound fields in rectangular rooms. Tetsuya Sakuma and Kazushi Eda (The Univ. of Tokyo, 5-1-5 Kashiwanoha, Kashiwa 277-8563, Japan, sakuma@k.u-tokyo.ac.jp)

Rectangular rooms with irregular aspect ratio or nonuniform absorption distribution apt to have non-diffuse sound fields, where the curvature of energy decay curve occurs in the reverberation process. In general, this curvature leads to longer reverberation times than the estimates by the Sabine or Eyring formula; however, it can be suppressed to a certain extent with diffusive wall surfaces. Recently, the author has proposed a new approximate theory of reverberation in rectangular rooms with specular and diffuse reflections. In this paper, the validity of the theory is investigated by two case studies with geometrical and wave-based acoustic simulation. In the first study, energy decay in a variety of rectangular rooms with changing the aspect ratio, the absorption distribution and the scattering coefficient, is simulated with the image source method and the ray tracing method. In the second study, a two-dimensional FDTD analysis is performed to demonstrate the frequency dependence of energy decay in a variety of rectangular rooms with flat or corrugated walls. Finally, the simulated results are compared with the theoretical ones, and the validity and the limitation of the approximation are discussed.

9:20

5aAAb2. Mean-free-paths in concert and chamber-music halls and validation of the Sabine/Eyring equations for predicting their reverberation times. Leo L. Beranek (10 Longwood Dr., 265, Westwood, MA 02090, beranekleo@ieee.org) and Noriko Nishihara (Takenaka R & D Inst., Chiba, Japan)

The Eyring/Sabine equations assume that in a large irregular room a sound wave travels in straight lines from one surface to another. It is assumed that the surfaces have an average sound absorption coefficient αav and that the average distance between reflections, i.e., the mean-free-path MFP, is 4V/Stot where V is the volume of the room and Stot is the total area of all of its surfaces. No account is taken of diffusivity of the surfaces. The 4V/Stot relation is based on experimental determinations in 11 very differently shaped rooms made by Vern Knudsen [*Architectural Acoustics*, (Wiley, 1932), pp. 132–141]. This paper sets out to test this relation experimentally for a wide variety of unoccupied concert and chamber music halls with seating capacities ranging from 200 to 15,000. To determine the MFP's in them, the measured values of the sound strengths G and reverberation times RT were used. Computer simulations of the sound fields for several of these rooms were also made to determine the MFP's. For these rooms, 4V/Stot is found to be an acceptable relation for MFP in the Sabine/Eyring equations.

9:40

5aAAb3. Exploration of the differences between a pressure-velocity based in situ absorption measurement method and the standardized reverberant room method. Peter Cats (Fontis Hogeschool Eindhoven, Eindhoven, Netherlands), Emiel Tijs, and Daniel Fernandez Comesana (Microflown Technol., Tivolilaan 205, Arnhem 6824BV, Netherlands, tijs@microflown.com)

Several measurement techniques are available for the determination of the sound absorbing properties of material packages. The Kundt's method and the reverberant room method are the most commonly used techniques and they are standardized. However, both methods cannot be used *in situ*. In the past, it has been shown that the PU *in situ* method can be used in a broad frequency range (typically from 300 Hz up to 10 kHz), on small samples (typically 0.03 m^2 to 0.38 m^2 or larger), while hardly being affected by background noise and reflections. Several studies revealed that similar results can be obtained as with the Kundt's tube if the measurements are performed under certain circumstances. A thorough comparison with the reverberant room method has not been conducted yet. In this paper, preliminary results are presented of a comparison of the reverberant room method, the PU *in situ* method, and measurements with PU probes in a reverberant room. Several factors that may cause discrepancies amongst the methods are discussed. In addition, edge effects, which are experienced with the reverberant room method due to the finite size of the sample, are visualized with 3D intensity measurements that are performed in a reverberant room.

10:00

5aAAb4. The reverberation radius (rH) in an enclosure with asymmetrical absorption distribution. Higini Arau–Puchades (ArauAcustica, Barcelona, Spain) and Umberto Berardi (CEE, Worcester Polytechnic Inst., via Orabona 4, Bari 70125, Italy, u.berardi@poliba.it)

This paper reviews the concept of the reverberation radius (rH) from the viewpoint of the classic theories of Sabine and Eyring. These theories are only valid when the sound field is uniformly distributed around a room, or in other words, when the energy density throughout a room is constant. Nevertheless, these theories have been applied to any spatial sound diffusion situation, also where this is not diffuse. For example, they are currently used in rooms with asymmetric absorption distribution. The limits of this approach also characterize the revised theory of Barron and more recent investigations. This paper proposes a solution to calculate the reverberation radius in rooms with non-uniformly distributed sound absorption (rH-ND).

10:20–10:40 Break

10:40

5aAAb5. Optimal design of a sound reflector by particle swarm optimization. Yuuki Tachioka (College of Humanities and Sci., Nihon Univ., E612 3-6-13 Fujisawa, Fujisawa, Kanagawa 251-0052, Japan, yuuki_tachioka@yahoo.co.jp)

For room acoustic design, the shape of the room and the materials need to be well considered. Designing such rooms requires extensive experience. Numerical analyses can minimize designers' burdens by the automatic generation of many room candidates and the selection of an optimized one on the basis of objective measurements. Research on both numerical analysis methods and optimal design methods is required, but few studies are done regarding the latter. This paper proposes an optimal method by combining geometrical acoustic simulation with an optimization technique known as "particle swarm optimization." Although the final aim is to achieve a global acoustic design, this paper is limited to local shape design, which is still important, considering that local shapes affect the characteristics of the global sound field. The proposed method is used for designing sound reflectors, whose aim is to scatter first reflected sound waves across the entire field. Hence, the optimization variables are sound reflectors shapes and materials, and the objective function is to evaluate the uniformity of a sound field in terms of room acoustic indices (T20, C80, and Ts). Compared with the initial reflectors, the optimized sound reflectors exhibited a higher uniformity in the room acoustic indices.

11:00

5aAAb6. How absorptive can a diffuser be to accurately measure random incidence scattering coefficients? Isabelle Schmich-Yamane, Marina Malgrange, and Christophe Rougier (Département Acoustique et Eclairage, CSTB, 24 rue Joseph Fourier, Grenoble 38000, France, christophe.rougier@cstb.fr)

In auditorium design, absorptive surfaces are as important as diffusive ones. Diffuse surface elements are rarely completely reflective. Absorption and scattering coefficients are not independent from each other. A real-size diffuser designed to be installed in a new auditorium under construction in France has been measured according to the ISO 17497-1 standard. This diffuser shows higher absorptiveness than any previously tested sample in CSTB's laboratory. The calculation of the random incidence scattering coefficient in the ISO 17497-1 standard is based on the Sabine equation for elements with α_s less than 0.5. This paper presents how scattering coefficient calculations give incoherent results for more absorptive diffuse elements and proposes a new calculation method based on the Eyring equation. Measurement results obtained with both methods are presented and compared.

11:20

5aAAb7. Prediction and measurement of the random incidence scattering coefficient of periodic reflective rectangular diffuser profiles. Isabelle Schmich-Yamane (Département Acoustique et Eclairage, CSTB, 24 rue Joseph Fourier, Grenoble 38000, France, isabelle.schmich@cstb.fr), Jean-Jacques Embrechts (INTELSIG group - Département E.E.I., Université de Liège, Sart-Tilman (Liege 1), Belgium), Markus Müller-Trapet (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany), Christophe Rougier (Département Acoustique et Eclairage, CSTB, Saint MArtin d'Hères, France), and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany)

Periodic reflective rectangular diffuser profiles have been previously studied by Embrechts through an analytical method calculating their random incidence scattering coefficients. This paper presents the results of further investigations obtained by real-scale and 1:5th scale random incidence scattering coefficient measurements on these profiles. All measurements were performed according to the corresponding ISO 354 and ISO 17497-1 standards. The measurement results are compared with the calculation results derived by the analytical method. Although some periodic profiles could not be treated by the analytical model, fairly good agreements are obtained between the different approaches. The real scale and 1:5th scale measurement results are also presented and compared.

Session 5aBAa

Biomedical Acoustics: Acoustic Characterization of Biological Media

James C. Lacefield, Chair
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Contributed Papers

9:00

5aBAa1. Evaluation of tumor cell death response in locally advanced breast cancer patients to chemotherapy treatment by scattering property estimates using ultrasound backscatter. Lakshmanan Sannachi, Hadi Tadayyon, Ali Sadeghi-Nani, Omar Falou, Zahra Jahedmotlagh (Imaging Res., Radiation Oncology, Med. Biophys., SunnyBrook Health Sci. Ctr., Univ. of Toronto, 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, Lakshmanan.Sannachi@sunnybrook.ca), Michael Oelze (Bioacoustics Res. Lab., Dept. of Electron. and Comput. Eng., Univ. of Illinois, Urbana, IL), and Gregory Czarnota (Imaging Res., Radiation Oncology, Med. Biophys., SunnyBrook Health Sci. Ctr., Univ. of Toronto, Toronto, ON, Canada)

Breast cancer tumor response to chemotherapy in 29 patients was examined using quantitative ultrasound. Backscatter parameters, such as the average scatterer diameter (ASD) and average acoustic concentration (ACC), were estimated from regions-of-interest in tumors prior to treatment onset and at four times during neoadjuvant chemotherapy treatment (weeks 1, 4, 8, and prior to surgery). Gaussian and Anderson form factor models were used over an analysis bandwidth of 4 to 9 MHz to obtain ASD and ACC estimates. The Gaussian model did not fit with the measured data as well as Anderson model. Both ASD and ACC estimates yielded significant differences with therapy times in clinically treatment responded patients. Data indicated increases of approximately 3.5 μm in scatterer size and 6.6 dB/cm in acoustic concentration obtained maximum at week 8 in treatment responding tumors. Non-responding tumors did not show any significant difference in both the parameters. This study demonstrates that the scattering parameters have the potential to being used in quantifying the changes in tumors during treatment noninvasively and distinguishing treatment responders and non-responders.

9:20

5aBAa2. Characterizing tumor heterogeneous response to chemotherapy using low-frequency ultrasonic spectroscopy. Ali Sadeghi-Naini, Omar Falou, and Gregory J. Czarnota (Imaging Res., Radiation Oncology, Med. Biophys., Sunnybrook Health Sci. Ctr., Univ. of Toronto, Odette Cancer Ctr., TB064, 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, a.sadeghi.naini@utoronto.ca)

Textural analysis techniques in conjunction with low-frequency ultrasonic spectroscopy have been proposed for characterization of heterogeneous responses developed within tumors undergoing chemotherapy. Such characterization can be beneficial for early appraisal of tumor responses to the cancer therapy. Breast xenograft tumor-bearing animals were treated with chemotherapy. Animals were assessed with low-frequency ultrasound data acquired at different times after chemotherapy exposure. Following imaging, tumors underwent standard histological analysis for detecting cell-death effects. Several average and texture-based parameters were derived from normalized ultrasonic spectral parametric maps. Statistically significant differences were revealed between parameters extracted from treated and untreated tumors 12–24 h after exposure. Regression analyses were also performed in order to assess levels of correlation between non-invasive ultrasonic surrogates of therapy response and standard histological findings, where strong correlations were obtained with a maximum r^2 value of 0.98. Obtained results demonstrate that ultrasound-based textural properties of tissue can be used for characterizing the micro-structure alterations and heterogeneous responses within tumors early on during treatment. This is an important observation which can be applied clinically for treatment response monitoring and predicting patient responses to therapy early after therapy initiation.

9:40

5aBAa3. Ultrasonic assessment of the *in vitro* biomechanical stability of a dental implant. Romain Vayron (CNRS, Laboratoire MSME, 61, Ave. du Général de Gaulle, bât P2, Créteil 94010, France, romain.vayron@u-pec.fr), Patrick Karasinski (Université Paris Est, Laboratoire LISSI, Créteil, France), Domitille Loriot, and Guillaume Haiat (CNRS, Laboratoire MSME, Créteil, France)

Dental implants are widely used for oral rehabilitation. However, there remain risks of failure, which are difficult to anticipate. The objective is to investigate the potentiality of a quantitative ultrasound method to assess the biomechanical stability of a dental implant *in vitro*. Two experimental configurations were considered using a 10 MHz contact transducer located at the implant extremity. For each ultrasound measurement, a quantitative indicator I is derived based on the time variation of the amplitude of the rf signal. First, seven implants were embedded in tricalcium silicate-based cement. One implant was left without any mechanical solicitation and six implants were subjected to mechanical stresses during 24 h. The ultrasonic response of each implant was measured during 24 h. The results show no variation of I without mechanical solicitation, while I significantly increases as a function of fatigue time. Second, 10 implants were unscrewed from bone tissue and their ultrasonic response was measured after each turn. Analysis of variance tests revealed a significant effect of the amount of bone in contact with the implant on the distribution of I. The results show the feasibility of our QUS device to assess the biomechanical quality of the interface surrounding the implant.

10:00

5aBAa4. Axial transmission measurements of guided modes on bone phantoms and *in vitro* human bone specimen. Jean-Gabriel Minonzio, Josquin Foiret, Maryline Talmant, and Pascal Laugier (LIP (Laboratoire Imagerie Parametrique), UPMC - CNRS, 15 rue ecole de medecine, Centre des Cordeliers, Paris 75006, France, jean-gabriel.minonzio@upmc.fr)

Cortical bone porosity has been evidenced as being a major if not the major "footprint" of bone loss and fragility. Several studies report that cortical bone behaves like a waveguide. Measurements of guided mode wavenumbers together with appropriate waveguide modeling have therefore the potential for providing estimations of effective stiffness coefficients (which are largely determined by cortical porosity) and also cortical thickness. However, data interpretation is challenging due to the heterogeneous, dissipative, and irregular nature of the wave guide. Moreover, surrounding and internal tissues modify the guided modes. This paper presents measurement of the wavenumbers of guided wave modes, using a multi-transmitter multi-receiver axial transmission probe. The method has been previously validated on isotropic elastic and visco-elastic plates covered by a soft tissue-mimicking layer. As an extension of our previous works, we have conducted *in vitro* experiments on (i) bone-mimicking tubes without and with soft-tissue mimicking layers and on (ii) human radius and tibia specimens. Despite challenging measurement conditions owing to low signal-to-noise ratio,

guided mode wavenumbers have been successfully measured. The results were consistent with a free plate model. Data inversion led to an reasonably accurate estimate of the shell thickness and bulk velocities for all cases.

10:20

5aBAa5. Determination of bone properties from Lamb type of waves. Jean-Gabriel Minonzio, Josquin Foiret, Pascal Laugier, and Maryline Talmant (Laboratoire d'Imagerie Parametrique, UPMC Univ Paris 06, UMR 7623, 15 rue de l'Ecole de Medecine, Paris F-75006, France, maryline.talmant@upmc.fr)

Our domain of interest is cortical bone characterization, i.e., determination of structural and material properties by means of ultrasound waves. The frequency spectrum of Lamb type of waves are the input data of the method of parameters identification, based on a least mean square algorithm. Specificities of clinical measurements of long bones require first to address the issue of measurements with limited access [Minonzio *et al*., J. Acoust. Soc. Am. (2010)] and second, to develop methods for a combined determination of structural and elastic properties. The first physical model was the free plate under plane strain assumption for transverse isotropic materials with four elastic parameters and one structural (thickness). Experiments on flat plates and tubes of circular cross section are used as test case and the method was validated by comparison to independent techniques dedicated to material properties [Bernard *et al*. (2012)]. The same method was applied on *in vitro* experiments on cortical bone samples. Here, the bone thickness was measured independently by x-ray technique. This study opens the perspective of updating the physical model to determine additional relevant parameters, the tube diameter and/or the soft tissue properties in case of *in vivo* measurements.

10:40

5aBAa6. Development and validation of resonant ultrasound spectroscopy for the measurement of cortical bone elasticity on small cylindrical samples. Simon Bernard, Quentin Grimal, and Pascal Laugier (Laboratoire d'Imagerie Paramétrique, CNRS - UPMC Univ Paris 6, UMR 7623, 15 ru de l'école de médecine, escalier A, 3ème étage, Paris 75006, France, simon.bernard@upmc.Fr)

Documentation of cortical bone elastic properties is important for orthopedic applications and fracture risk prediction. Cortical bone is heterogeneous and anisotropic. Ideally, measurements should be performed on cylindrical samples of characteristic size of a few millimeters, which are adapted to the native geometry and size of bones. Our objective was to measure bone with resonant ultrasound spectroscopy (RUS), which is a powerful method to determine the elastic constants of a sample from a set of its resonant frequencies. Application of RUS to bone is difficult due to viscoelasticity, which causes resonance peaks to overlap. Some of the resonances in the investigated frequency band cannot be observed. The formulation of the inverse problem, which requires pairing measured and predicted frequencies, must be adapted in the case of bone. We developed dedicated signal processing methods to retrieve resonant frequencies from overlapping peaks. A probabilistic approach was used, which allows an automated pairing based on a Bayesian criterion. The method was validated on isotropic (PMMA) and anisotropic (bone phantom) calibrated materials. The adapted RUS method allows the automated assessment of bone elasticity with a precision of a few percents, which is a significant improvement over traditional methods based on velocity measurements.

11:00

5aBAa7. Ultrasound propagation through bone fractures with reamed intramedullary nailing: Results from numerical simulations. Fernanda Catelani (Biomed. Eng. Program - COPPE, Federal Univ. of Rio de Janeiro, Nova Friburgo, Brazil), Ana Paula M. Ribeiro (Biomed. Ultrasound Lab., Estácio de Sá Univ., Nova Friburgo, Brazil), Carlos Alberto V. Melo (Physical Therapy Dept., Estácio de Sá Univ., Nova Friburgo, Brazil), Wagner C. Pereira (Biomed. Eng. Program - COPPE, Federal Univ. of Rio de Janeiro, Rio de Janeiro, Brazil), and Christiano B. Machado (Biomed. Ultrasound Lab., Estácio de Sá Univ., Av. Ariosto Bento de Mello, 35/101, Nova Friburgo, Rio de Janeiro 28610100, Brazil, cbmfisio@gmail.com)

Low-intensity pulsed ultrasound stimulation (LIPUS) accelerates fracture healing, enhancing the release of inflammatory mediators and subsequent bone formation. The reamed intramedullary nailing (with the same diameter of the medullary cavity) is a surgical procedure widely used in Medicine. The aim of this work was to study ultrasound propagation inside fractures with and without reamed intramedullary nailing using 2D simulations. It was used a custom-made simulation code applied to numerical models (a 4-mm thick cortical plate, a medullary cavity with radius 4 mm with and without reamed nailing, and fracture gaps varying from 1 to 3 mm). A 1-MHz emitter was positioned above fracture center, and 14 receptor transducers were uniformly placed inside fracture gap. The acquired signals were used to estimate the time-of-flight of the first arriving signal (TOF) and the energy amplitude by means of the root mean square (RMS). TOF was slightly influenced by fracture gap variations. It was observed an increase in RMS values with the presence of metal nailing, due to the reflection in the interface water-metal. The receptors placed near cortical plates received more energy (constructive interference between the direct and lateral waves). For the case of reamed nailing, ultrasound stimulation may be intensified.

11:20

5aBAa8. Computational simulations of time of flight and attenuation of first arriving signal from healing process of diaphyseal femur fractures. Paulo Tadeu C. Rosa, Aldo José F. Pereira, and Wagner Coelho A. Pereira (Biomed. Eng., UFRJ, P.O. Box 68510, Univ. City, Rio de Janeiro, Rio de Janeiro 21941-972, Brazil, ptcardozo@gmail.com)

Quantitative ultrasound (QUS) has been proposed to evaluate structural conditions from bone tissue in a non-invasive way. These techniques are based on the fact that the ultrasound propagation depends directly of tissues structures, so they carry information about them. Based on these arguments, the aim of this work was to estimate the time-of-flight (TOF) and attenuation of the first arriving signal (FAS) using computational simulations (Wave2000® CyberLogic, Inc., NY, EUA) in an axial transmission model in different kinds of fracture, during the healing process. In this work, we used QUS techniques to analyze the fracture healing process. The FAS has been chosen because it does not suffer interference of other waves since it is the first signal to arrive at the receiver. TOF increases immediately after bone fracture. When bone tissue starts to consolidate, TOF decreases and stabilizes with the same value of the intact bone. Attenuation is bigger in oblique and spiculate fracture than transversal ones for the same stage, which suggests that attenuation is sensitive to the kind of fracture. Other studies are being conducted to clarify this point.

11:40

5aBAa9. Ultrasonic attenuation and speed in phantoms made of polyvinyl chloride-plastisol and graphite powder. Guillermo A. Cortela (Ultrasonic Lab., U de la República, Montevideo, Uruguay), Luis E. Maggi (Escola Superior de Educacao Fisica, Universidade Estadual de Goias, Goiania, Brazil), Marco Antonio v. Kruger (Programa de Engenharia Biomédica - COPPE, Universidade Federal do Rio de Janeiro, Rio de Janeiro, Brazil), Carlos A. Negreira (Ultrasonic Lab., Universidad de la República, Montevideo, Montevideo, Uruguay), and Wagner C A. Pereira (Programa de Engenharia Biomédica - COPPE, Universidade Federal do Rio de Janeiro, P.O. BOX 68510, Rio de Janeiro, Rio de Janeiro 21941972, Brazil, wagner.coelho@ufrj.br)

Biological phantoms are very useful for controlled experiments on biomedical ultrasound. Nevertheless they are normally made of organic materials with short time-duration. We have studied the ultrasonic properties of test-blocks made of polyvinyl chloride-plastisol (PVCP) that are very stable in time. In this work, we analyzed ultrasonic (US) attenuation and speed at 1 MHz, as a function of temperature (15–45°C) of five phantoms made with PVCP and different concentrations of graphite powder (0, 0.5, 1, 2, and 5%) using the classical transmission method. US speed diminishes almost linearly (from 1408 to 1333 $m.s^{-1}$) as temperature increases. In general attenuation lied between 0.73 and 0.09 $dB.cm^{-1}$, but presenting a more complex behavior. For graphite concentrations of 0.5 and 1%, attenuation was lower than for 0% and for the other two phantoms (2 and 5% concentrations) attenuation was higher. This behavior can be perhaps due to the fact that the fabrication temperature for 0.5 and 1% was 140°C and for the other was 170°C. Although the standard recipe is 170°C, we observed that smaller temperatures may add in adjusting the attenuation values and it is a very useful property to mimic different biological tissues. We are now working on multilayer phantoms of PVCP.

Session 5aBAb

Biomedical Acoustics: Imaging, Therapy, and Bubbles (Again)

Michel Versluis, Chair
Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands

Contributed Papers

9:00

5aBAb1. The effect of boundary proximity on the fundamental and subharmonic emissions from individual microbubbles at higher frequencies. Brandon Helfield (Med. Biophys., Univ. of Toronto, 2075 Bayview, Toronto, ON M4N 3M5, Canada, brandonhelfield@gmail.com), Ben Leung (Sunnybrook Res. Inst., Toronto, ON, Canada), and David E. Goertz (Med. Biophys., Univ. of Toronto, Toronto, ON, Canada)

It is recognized that the proximity of a boundary can influence the dynamic behavior of acoustically stimulated microbubbles. In a biomedical ultrasound context, this is relevant to molecular imaging with targeted microbubbles, and when microbubbles are near vessel walls or contained within microvessels. Theoretical models have recently been developed to examine these effects, but experimental work has been more limited and primarily focused on the assessment of resonant frequency effects rather than its impact on nonlinear behavior, which is perhaps more relevant to imaging applications. Understanding this behavior is important to improving microbubble detection and for the quantitative interpretation of contrast images. With the use of an optical trap, this study experimentally investigates the effect of boundary proximity (0 to 200 μm) and boundary stiffness (Opticell and agarose) on fundamental and subharmonic emissions from individual Definity and MicroMarker bubbles at 11 MHz. The scattered pressure dependence on proximity from an Opticell boundary resulted in an oscillatory dependence, while from an agarose boundary resulted in a decreasing fundamental and an increasing subharmonic response with increasing distance from the boundary. These experimental findings are not entirely captured by basic analytical simulations, likely suggesting that more complex numerical models may be required.

9:20

5aBAb2. Bifurcation structure of the ultrasonically excited microbubbles undergoing buckling and rupture. Amin Jafari Sojahrood, Raffi Karshafian, and Michael C. Kolios (Dept. of Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B2K3, Canada, amin.jafarisojahrood@ryerson.ca)

Bubbles exposed to ultrasound are long known to exhibit highly nonlinear and chaotic dynamics. Bubbles stabilized by a shell material (MBs) are widely used as contrast agents in diagnostic ultrasound. However, the nonlinear behavior of the shell significantly increases the complexity of the dynamics. In order to realize the full potential of the MBs, better understanding of the MB behavior is necessary. In this study, the bifurcation structure of the MB with nonlinear shell behavior is investigated for the first time. The Marmottant model was numerically solved, and the bifurcation diagrams of the radial oscillations of the MB were plotted versus the control parameters (e.g., buckling radius). In agreement with recent experimental observations, results predict the generation of subharmonics at very low acoustic pressures. In addition, the numerical simulations predict the generation of higher order subharmonics (e.g., period 3) at very low acoustic pressures (<300 kPa and 25 MHz), which contradicts the predictions by free bubble and viscoelastic shell MB models. Results revealed the strong influence of the buckling and rupture radius on the order of the subharmonics. The numerical results were verified by experimental observations of higher order subharmonics in the oscillations of Definity at 25 and 55 MHz.

9:40

5aBAb3. Ultrafast dynamics of the acoustic vaporization of phase-change microdroplets. Oleksandr Shpak, Laura Stricker (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands), Tom Kokhuis, Ying Luan (Biomed. Eng., Erasmus MC, Rotterdam, Netherlands), Brian Fowlkes, Mario Fabiilli (Dept. of Radiology, Univ. of Michigan, Ann Arbor, MI), Detlef Lohse (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands), Nico de Jong (Biomed. Eng., Erasmus MC, Rotterdam, Netherlands), and Michel Versluis (Phys. of Fluids Group, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, m.versluis@utwente.nl)

Superheated emulsion droplets are a promising tool for localized drug delivery. The physical mechanisms underlying the ultrasound-triggered vaporization of phase-change emulsions are largely unexplored. Here we study the acoustic vaporization of individual micron-sized perfluoropentane droplets at a nanoseconds timescale. The nucleation and growth of the vapor bubbles was imaged at frame rates up to 20 Mfps. The droplet vaporization dynamics was observed to have three distinct regimes: (1) prior to nucleation, a regime of droplet deformation and oscillatory translations; (2) a rapid growth of a vapor bubble enhanced by ultrasound-driven rectified heat transfer; and (3) a final phase characterized by a relatively slow expansion that is fully dominated by heat transfer. A method to measure the moment of inception of the nucleation event with respect to the phase of the ultrasound wave is proposed. A simple physical model captures quantitatively all of the features of the subsequent vapor bubble growth. In addition, we study the role of gas through a model for a vapor-gas bubble, including thermal diffusion and gas diffusion inside the liquid and we find good agreement with the experimental data. We underline the fundamental role of gas diffusion to prevent total recondensation of the bubble at collapse.

10:00

5aBAb4. Dynamics of a constrained bubble. Alexey Maksimov and Yuri Polovinka (Phys. of the Ocean, Pacific Oceanological Inst. Far Eastern Branch of the Russian Acad. of Sci., 43, Baltic St., Vladivostok 690041, Russian Federation, maksimov@poi.dvo.ru)

In recent studies [Bostwick and Steen (2009), Prosperetti (2012), Ramalingam *et al*. (2012)], the shape oscillations of constrained drops and bubbles have been analyzed and the difficulty of possible singularity in the pressure and the curvature of the free surface at the location of the constraint has been identified. This result indicates the need for a small cutoff length scale, as well as some more input from the physics at smaller length scales. The method of restraining the bubble against rising by attaching it to a wire is a common procedure in conducting precision acoustic measurements. The dynamics of the tethered bubble differs from those of free bubble due to variation in inertial mass. The objective of this study is to obtain a closed-form, leading order solution for the dynamics of the constrained bubble. It was shown that, by using the invariance of the Laplace equation to conformal transformations and the geometry of the problem, the toroidal coordinates provide separation of variables and are most suitable for analysis of this problem. Thus, the dynamics of the constrained bubble in toroidal coordinates can be investigated by using analytical approach and by analogy to the dynamics of a free spherical bubble.

10:20

5aBAb5. Temporal and spatial characteristics of nonlinear acoustic field generated by an extracorporeal shockwave therapy device: Modeling and measurements. Maria Karzova (Dept. of Acoust., Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation and Lab. de Mécanique des Fluides et d'Acoustique, Ecole Centrale de Lyon, Leninskie Gory 1/2, Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Vera Khokhlova (Dept. of Acoust., Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation and Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Camilo Perez, and Thomas Matula (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Extracorporeal shock wave therapy (ESWT) refers to the use of focused shock pulses to treat certain musculoskeletal disorders. Although the technology is used clinically, acoustic characteristics of ESWT fields and their relation to the bioeffects induced are not fully understood. In the present work, the acoustic field of a clinical ESWT device (Duolith SD1) was characterized in water using a combined measurement and modeling approach. Simulation model was based on the nonlinear KZK equation; the boundary condition was based on the pressure waveforms measured in a plane 5 mm away from the therapy head. The model was used to simulate and to analyze pressure waveforms along the axis of the therapy head, 2D spatial distributions of peak pressures, and the shock structure. The modeling results were found in a good agreement with experimental data. Simulations performed for different initial source pressure amplitudes showed that a shock is not formed at the focus, even at the maximum operational level of the device. Predictions from the modeling at higher output settings suggest that a true shock would develop if the initial pressure output were doubled. [Work supported in parts by RFBR 12-02-31830 and 12-02-31418 grants and student stipend from the French Government.]

10:40

5aBAb6. Dynamic time reversal acoustic focusing of ultrasound for biomedical applications. Yegor D. Sinelnikov (SUNY at Stony Brook, 126 Liberty Ave., Port Jefferson, NY 11777, yegorasha@yahoo.com), Alexander M. Sutin (Stevens Inst. of Technol., Hoboken, NJ), Sergey Y. Tsyuryupa, and Armen P. Sarvazyan (Artann Labs., Ewing Township, NJ)

Time Reversal Acoustic (TRA) system provides effective focusing in inhomogeneous media that can be used in various biomedical applications including high intensity ultrasound treatment, ultrasound-assisted drug delivery, ultrasonic battery charging of implants, etc. In many cases, the TRA focusing is conducted in tissues with time varied properties and variation in propagating media degrades the TRA focused signal. Dynamic focusing is required to maintain the same acoustic intensity in the focus. We suggest adjustment to the radiated signal in order to maintain the maximal amplitude in the focus using the inverse filtering technique with Tikhonov regularization. The suggested algorithm enables to determine the radiated signal based on the changes in the focused signal received by a beacon. Developed algorithm of TRA refocusing was tested in the experiments with a TRA system with an aluminum reverberator comprising ten piezotransducers with resonance around 1 MHz, DAC control system and a multichannel binary amplifier. This system was built in Artann Laboratories for investigation of time reversal acoustic focusing for hyperthermia therapy and drug delivery. The dynamic TRA refocusing is shown to be effective even in the case of significant lateral shifts in the media and strong focused signal variation.

11:00

5aBAb7. Superresolution imaging in ultrasound B-scan imaging. Kevin J. Parker (Dept. of Elec. & Comput. Eng., Univ. of Rochester, Hopeman Eng. Bldg. 203, P.O. Box 270126, Rochester, NY 14627-0126, kevin.parker@rochester.edu)

A number of imaging systems exhibit speckle, which is caused by the interaction of a coherent pulse reflecting off of random reflectors. The limitations of these systems are quite serious since the speckle phenomenon creates a pattern of nulls and peaks from subresolvable scatterers or targets that are difficult to interpret. Another limitation of these pulse-echo imaging systems is that their resolution is dependent on the full spatial extent of the propagating pulse, usually several wavelengths in the axial or propagating dimension and typically longer in the transverse direction. This limits the spatial resolution to many multiples of the wavelength. This paper focuses on the particular case of ultrasound B-scan imaging and develop an inverse filter solution that eliminates both the speckle phenomenon, and the poor resolution dependency on the pulse length and width, to produce SURUS (super-resolution ultrasound) images. The key to the inverse filter is the creation of pulse shapes that have stable inverses. This is derived by use of the standard Z-transform and related properties. Although the focus of this paper is on examples from ultrasound imaging systems, the results are applicable to other pulse-echo imaging systems that also can exhibit speckle statistics.

11:20

5aBAb8. Wearable long duration ultrasound therapy in rotator cuff tendinopathy. George Lewis (ZetrOZ, 421 N. Aurrora St., Ithaca, NY 14850, george@zetroz.com), Lyndon Hernanez (Med. College of Wisconsin, Milwaukee, WI), George Lewis (Transducer Eng., Andover, MA), and Ralph Ortiz (Cayuga Med. Ctr., Ithaca, NY)

Approximately one-third of the westernized adult population will experience some type of shoulder pain. The purpose of this pilot study was to evaluate a novel self-applied wearable therapeutic ultrasound device in the management of shoulder pain from rotator cuff tendinopathy. The Institutional Review Board of Cayuga Medical Center (CMC) approved this study and informed consent for the study was obtained from all subjects. The wearable ultrasound device provides 90 mW/cm^2, 2.95 MHz, continuous-wave ultrasound for 5.5 h on a single charge. Four subjects meeting the studies inclusion criteria, presenting with rotator cuff tendinopathy, and demonstrating cognitive and functional ability to apply the pager-size device were enrolled at the outpatient physical therapy center of CMC. Subjects were instructed to wear the device for 3–4 h/day for 12 consecutive treatment sessions, and record their daily pain score on the visual analog scale (1 to 10) and global rate of health improvement scale (-7 to 7). Across the 12 treatments, subjects reported a 30% reduction in pain and 52% improvement in health compared to baseline scores ($p < 0.05$). The results of the pilot study indicate the device may be applied successfully and provides supportive evidence for a placebo controlled study.

11:40

5aBAb9. Ultrasound-equipped colonoscope for point-of-procedure colorectal preparation and examination. Lyndon Hernanez (Med. College of Wisconsin, 8701 Watertown Plank Rd., Milwaukee, WI 14850, lvhernan@hotmail.com), Martin Ton (Cornell Univ., Ithaca, NY), Shane Fleshman, and George Lewis (ZetrOZ, Ithaca, NY)

Colorectal cancer is the second leading cancer killer in the United States. Poor colon preparation occurs in 20–40% of colonoscopies in the community, which increases the duration of the colonoscopy by at least 10% and the cost of the procedure by up to 22% due to repeat visits. The goal of this research was to develop and preliminarily evaluate the first ultrasound-cavitation equipped colonoscope as an innovative approach to liquefy fecal matter with water/cavitation and improve colonoscope utility. Two ultrasound-equipped colonoscopes were developed. The first consisted of a 30 element 235 kHz array that was mounted as a cap on the tip of a commercial colonoscope (Olympus). The second consisted of a Time-Reversal Acoustic extroporeal 32 channel 100 kHz array that was electrically steered to the commercial colonoscope using PVDF detectors to acquire and monitor the TRA focusing routines. Both systems were evaluated in a series of bench tests for fecal liquification, as well as in the porcine cadaver. Results show that ultrasound exposure assists the liquification of fecal matter and 50 kPa exposure to ultrasound increases liquification by greater than 50 times. Blinded histological reports on excised tissues showed no significant different findings between control and ultrasound experiments.

Session 5aEA

Engineering Acoustics: Sound Emission from Vehicle and Rotating Machinery

Stephen Elliott, Chair
Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom

Contributed Papers

9:00

5aEA1. Effects of median barriers on highway noise levels. Jonghoon Kim (Civil Eng., Arizona State Univ., 472W. Carob Dr., Chandler, AZ 85248, ghim96@gmail.com), Louis Cohn (Civil and Environ. Eng., Univ. of Louisvile, Louisville, KY), and Ning Shu (Environ., AZTEC Eng., Phoenix, AZ)

Median barriers are widely used on roadways in the United States. The main purpose of this paper was to evaluate a median barrier performance in reducing traffic noise using the latest FHWA Traffic Noise Model, TNM version 2.5. For this study, median barriers were modeled on three different roadway configurations—at grade, depressed, and elevated. The analysis results indicated that the range of insertion loss for a median barrier at grade was less than 1.5 dBA with a barrier height of 2.5 to 10 ft. The range of insertion loss for a median barrier on a depressed roadway (5, 10, and 20 ft below grade) was 0 to 2.8 dBA with a barrier height of 2.5 to 4.5 ft and insertion loss increased up to 4.3 dBA with a taller barrier height of 6 to 10 ft. On an elevated roadway (5, 10, and 20 ft above grade), the range of insertion loss for a median barrier was 0 to 1.7 dBA with a barrier height of 2.5 to 10 ft. Given the results of this research, it is reasonable to conclude that a standard median barrier would not provide a significant level of noise reduction. Furthermore, taller median barriers do not alter this conclusion. When considering the construction of very tall median barriers for noise reduction purposes, the costs would far outweigh the relatively minimal benefits of this approach.

9:20

5aEA2. Multichannel feedback control of interior road noise. Jordan Cheer and Stephen J. Elliott (Inst. of Sound and Vib. Res., Univ. of Southampton, Univ. Rd., Highfield, Southampton, Hampshire SO17 2LG, United Kingdom, j.cheer@soton.ac.uk)

Active noise control systems offer a potential method of reducing the weight of passive acoustic treatments and, therefore, increasing a vehicles' fuel efficiency. The active control of engine noise can be implemented cost-effectively by using the car audio loudspeakers as control sources and an array of low-cost microphones as error sensors. Such systems have been commercially implemented, but without also controlling road noise their subjective benefits may be limited. The active control of road noise using a feedforward control strategy has also been practically demonstrated, but these systems require a number of accelerometers to be mounted to the vehicle's structure to obtain a coherent reference signal and, therefore, lead to a significant implementation cost. This paper proposes a multichannel feedback system for the active control of road noise, which uses an array of microphones and car audio loudspeakers, which is common to a feedforward engine noise control system. The design of the multichannel feedback controller is described and its performance is validated using offline simulations employing data measured in a small city car.

9:40

5aEA3. Determination of the transfer matrix of an automotive compressor under realistic flow conditions. Benoit Rousselet (Lab. PHASE, Université Paul Sabatier - Toulouse III, FR CTL L16 1 29, 1 allée Cornuel, Lardy 91510, France, rousseletbenoit@hotmail.com), Vincent Gibiat (Lab. PHASE, Université Paul Sabatier - Toulouse III, Toulouse, France), Alain Lefebvre, and Stephane Guilain (Powertrain Eng. Div., Renault S.A., Lardy, France)

Waves propagating into air intake pipes of automotive engines have been widely studied and are commonly used to obtain high volumetric efficiency. Nowadays, most of modern engines have a centrifugal compressor in their intake line. If the acoustic of intakes lines for naturally aspirated engines is well known, it is not the case for lines comprising a compressor, where both the geometry and the flow effects are of primary interest. As it is quite difficult to experimentally determine the acoustical behavior of a complete line comprising a compressor, a new system has been developed for determining acoustical transfer matrices of each separated element of the line with flow. This new method is based on the two loads method; a derivative of the Two Measurements Three Calibrations calibration method has been introduced. The basics of the determination of transfer matrices with flow with our method are presented. Experimental results on cylindrical tubes and parts of intake lines are then discussed. Finally, an example of a transfer matrix of centrifugal compressor is presented.

10:00

5aEA4. Acoustical absorption by materials in a nacelle of turbojet. Ibanez R. Carlos (Mech., Benemérita Universidad Autónoma de Puebla, Boulevard del Niño Poblano 2901, Puebla, Puebla 72197, Mexico, carlos.ibanez@iberopuebla.mx) and Panneton Raymond (Génie Mécanique, Université de Sherbrooke, Sherbrooke, QC, Canada)

The paper discusses the absorption of certain materials with different acoustic properties used in a nacelle of aircraft is investigated using numerical model. A model 2-D with the configuration of axisymmetrical and the differential equations of the Continuity and Helmholtz equation is realized, it couples a finite element description with a boundary conditions a rigid wall, impedance and a frequency range (low–high) in order to evaluate three treatments (resonator, foam, and the combination of both of materials) and his frequency behavior. Experimental results are presented to validate the model in the special case of insertion loss and the coefficient of sound absorption.

10:20

5aEA5. Investigation of hysteresis effect in the impinging jet using acoustic data. Abhijit Dhamanekar and K. Srinivasan (Mech. Eng., Indian Inst. of Technol., Madras, TDCE Lab., Mech. Dept. IITM, Chennai, Tamilnadu 600036, India, abhikd11@gmail.com)

Experimental acoustic investigation of underexpanded free and impinging jet is carried out for various nozzle-plate spacing. The reservoir pressure is slowly increased from atmospheric pressure to 6 bar and than decreased from 6 bar to atmospheric pressure. The free jet acoustic radiation remains same for both paths, but it is observed that for impinging jet the acoustic

radiations differ in some regions. The hysteresis effect observed in acoustic characteristics may be due to the presence of hysteresis effects in the recirculation zone of the impinging jet. This variation is significant for nozzle-plate spacing of 2 to 4 times jet diameter. It is also seen that the acoustic staging occurs for low pressure and high pressure for small and large nozzle-plate spacing, respectively.

10:40

5aEA6. Effects of scalloping depth on the sound generated by turbofan engine lobed mixers. Hao Gong, Kaveh Habibi, and Luc Mongeau (Dept. of Mech. Eng., McGill Univ., Rm. 364, McDonald Eng. Bldg., 817 Sherbrooke St. West, Montreal, QC H3A 0C3, Canada, hao.gong@mail.mcgill.ca)

The growing stringency of community noise regulations for commercial turbo-fan engine requires the development of effective jet noise suppression configurations. The large number of geometrical design parameters for lobed mixers precludes trial and error experimental studies. In this study, a robust computational tool was used to investigate the effects of the scalloping depth on sound radiated from a lobed mixer. The near field sound and flow were simulated using a flow solver based on the Lattice Boltzmann Method (LBM). The far field radiated sound was predicted using the Ffwocs William-Hawkings (FWH) surface integral method. One baseline confluent nozzle and three nozzles with varying scalloping depth were considered. Low Mach number flow was assumed, with operating conditions selected to best approach conditions for actual engines. The effects of an outer mean flow to simulate forward flight were not included. The increased scalloping depth was found to enhance mixing and to reduce noise levels in the mid-to-high frequency range, as anticipated. Results were in qualitative agreement with available experimental results.

11:00

5aEA7. Analysis of simulated flyover Contra Rotating Open Rotors noise data by using beamforming techniques for moving sources. Vincent Fleury and Alain Chélius (Onera, 29 Ave. de la Division Leclerc, Chatillon 92322, France, vincent.fleury@onera.fr)

The relevance of beamforming techniques to analyze conventional airframe noise sources from flyover noise measurements is now well-known. With the development of Contra Rotating Open Rotors (CROR), the performance of such techniques needs to be assessed. To this aim, realistic ground CROR noise data are simulated. First, the flow is simulated by a U-Rans approach in the CROR frame. The front rotor consists of 11 blades and the rear rotor is composed of 9 blades. In addition, the incoming flow Mach number is $M = 0.2$. Then, the hydrodynamic blade pressure is propagated toward a ground microphone array by using the Ffowcs-Williams Hawkings' equation. The CROR follows an horizontal trajectory at constant speed ($M = 0.2$) and 150 m altitude. Finally, the microphone array data are analyzed by a conventional beamforming-based deconvolution technique for moving sources, DAMAS-MS. The results show that the amplitude of the harmonics of each blade passage frequency is correctly recovered. However, the amplitude of the interaction tones is badly estimated. To overcome this difficulty, the DAMAS-MS methodology, based on an acoustic source model constituted of uncorrelated monopoles, should be modified in order to introduce the correlation between the monopoles.

11:20

5aEA8. An experimental investigation on the near-field turbulence for an airfoil with trailing-edge serrations and an owl specimen. Kunbo Xu, Weiyang Qiao, Liang Ji, and Weijie Chen (School of Power and Energy, Northwestern Polytechnical Univ., No.127 Youyi Rd. Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

The ability to fly silently of most owl species has long been a source of inspiration for finding solutions for quieter aircraft and turbomachinery. This study concerns the mechanisms of the turbulence for an airfoil with trailing-edge serrations and a real owl specimen . the turbulence spatio-temporal information are measured with 3D hot-wire. The experiment is carried out in the Northwestern Polytechnical University low speed open jet wind tunnel on the SD2030 airfoil. Mach numbers range up to 0.3, with the Reynolds numbers from 1.9e+4 to 2.6e+5, the angle of attack α at -50, 00, and +50, the sawtooth of $\lambda/h = 0.2$. The individual trailing-edge serrations tips and valleys could be seen in the wake region. It is showed the spreading rate of the wake and the decay rate of the wake centerline velocity deficit increased with serrated edge compared to the straight edge. It is also found that the turbulence peak occurs further from the airfoil surface in the presence of the serrations, and the serrations generate additional horseshoe vortices shed in the near wake. However, the boundary layer statistics slightly upstream from the TE are not influenced by the serrations.

11:40

5aEA9. Acoustic analysis of turbine generator to predict overall noise using statistical energy analysis methodology. Vasudev S. Nilajkar (India Technol. Ctr., GE, JFWTC, #122, EPIP, Phase 2, Hoodi Village, Whitefield Rd., Bangalore, Karnataka 560066, India, vasudev.nilajkar@ge.com), Sadeo Ramtahal, and Anil K. Tolpadi (Generator and Elec. Technol., GE, Schenectady, NY)

In the growing regions across the globe, products are being differentiated not only by their performance but also on characteristics like noise, aesthetics, ergonomics, etc. Noise norms are getting more and more stringent and customers are looking for quieter products. Engineers and technicians now face the challenge of developing effective products for lower noise emissions in addition to enhanced performance. As generators are a major noise source in a power plant, it is essential to contain generator noise well within the limit to meet statutory requirements. The major contributors to generator noise include cooling fan, rotor jet, electromagnetic excitation, and vibrations. These sources should be configured properly for reducing the overall noise. Therefore, it is essential to establish a robust process to predict the generator noise accurately. This paper describes the process for predicting generator noise by using Statistical Energy Analysis methodology. Correlation of the predictions with test data is also discussed.

Session 5aMUa

Musical Acoustics: Acoustic Analysis of Musical Instruments

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Vincent Gibiat, Cochair
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Contributed Papers

9:00

5aMUa1. Real-time magnetic resonance imaging fo the upper airways during harmonica pitch bends. Peter R. Egbert, Lewis K. Shin, David Barrett (School of Med., Stanford Univ., Stanford, CA), Thomas D. Rossing (School of the Blues, Los Altos Hills, CA), and Andrew Holbrook (Ctr. for Res. in Comput. Music and Acoust., Stanford Univ., Stanford, CA 94305, aholbrook@stanford.edu)

Skilled harmonica players learn to bend the pitch of certain notes by a semitone or more, especially in blues playing, by adjusting the shape of their vocal tract [Bahnson *et al*., J. Acoust. Soc. **103**, 2134 (2008)]. The changes of the vocal tract have been partially viewed with endoscopy and ultrasound but are still incompletely understood. While in a magnetic resonance imaging (MRI) scanner, a professional harmonica player using nonmagnetic, MRI-compatible diatonic harmonicas played draw and blow notes in both unbent and bent positions. Three-dimensional static and two-dimensional real-time magnetic resonance images of the upper airway were recorded in the sagittal and coronal planes. We identified and characterized the static and dynamic changes that facilitated pitch bends for low and high notes with specific attention to tongue positioning, tongue morphology, and airway shape. Deliberate changes in the tongue shape are often accompanied by changes in other parts of the vocal tract such as the pharynx.

9:20

5aMUa2. An acoustic study of ceramic traditional whistles. Vincent Gibiat (PHASE-UPS, Toulouse Univ., 118 route de Narbonne, Toulouse 31400, France, vincent.gibiat@univ-tlse3.fr) and Marie-Barbara Le Gonidec (Music Dept., MuCEM, Paris, France)

Ceramic whistles are very common all along the old European traditions. Gifts, jars, or simple decorative objects these whistles may have extremely various shapes: animals, jars, pitchers and sometimes flutes. Even with this extremely variability, their interior shapes are very similar: a channel where to blow, an edge, and a small cavity. Some of them produce a single sound when others present one or more tone holes, the latter corresponding mainly to flute shaped whistles. Their acoustic behavior appears to be very simple: Helmholtz resonators or flutes driven by the common non linear excitation system. Nevertheless some are known as water whistles and sometimes as "nightingales." The acoustic study presented will first verify that the simplest whistles are really working as Helmholtz resonator with the noticeable exception of the flute shaped ones that remain whistles for the organologist but are a different object for the acoustician. Then, a particular attention will be given to the "nightingales" ones. Their sound production will be related to the level of water they contain and the modulation of the acoustic signal analyzed in terms of coupling between the oscillation of the water and the Helmholtz acoustic resonance.

9:40

5aMUa3. Sound analysis and synthesis of Marquis Yi of Zeng's chime-bell set. Chih-Wei Wu (Master Program of Sound and Music Innovative Technol., National Chiao Tung Univ., 1001 Ta Hsueh Rd., Hsinchu 30010, Taiwan, g9611504@alumni.nthu.edu.tw), Chih-Fang Huang (Dept. of Information Commun., Kainan Univ., Luzhu Shiang, Taoyuan, Taiwan), and Yi-Wen Liu (Dept. of Elec. Eng., National Tsing Hua Univ., Hsinchu, Taiwan)

In this paper, the analysis and synthesis results from a complete set of Chinese chime-bells (also known as Chinese two tone bells) are presented. Consisting of 65 bells with different sizes and tones, Marquis Yi of Zeng's set is an ancient musical instrument with fascinating acoustical features but scarcely appears in current music performances for being huge and inaccessible. To preserve this cultural legacy in digital form, sounds of a complete set of replicated Marquis Yi of Zeng's chime-bells were recorded and analyzed, and the pitch discrimination of fundamental frequencies between this replica and the original set has been evaluated. Sound synthesis models of chime-bells were constructed using multiple inharmonic digital waveguides, creating chime-bell like sounds. Quality of synthetic sounds was evaluated using both objective and subjective measures. Objectively, the similarity between synthetic and recorded sounds was compared in both the spectral and the temporal domains. The subjective measure was achieved through listening tests. Results show that sounds from different bells on the rack could be successfully generated from the proposed models, leading toward the realization of virtual chime-bell set in the future.

10:00

5aMUa4. Acoustic analysis from pentatonic Angklung. Anugrah S. Sudarsono and I Gde Nyoman Merthayasa (Eng. Phys., Institut Teknologi Bandung, Jalan Ganesha 10, Bandung, West Java 40151, Indonesia, anugrahsabdono@gmail.com)

Angklung is traditional music instruments from Indonesia made from bamboo. Pentatonic angklungs are angklung that only have five notes. The notes are from the traditional Sundanese scale. Experiment was done to analyze spectral, temporal, and spatial parameters from pentatonic angklung. Spectral analysis was done by analyze the pitch and timbre. Temporal analysis was done by analyze the sound envelope and from τe parameter from the music. Spatial analysis was done from measurement of the sound energy by multiple microphones in semi anechoic room. Pitch and timbre were analyzed with Fast Fourier Transform from 29 angklungs. It was found from spectral analysis that angklung has overtone 1.44 f0, 2 f0, 3.47 f0, 6.31 f0, and 7 f0 with f0 as the fundamental frequency. The scales of the notes are very different from western scale and pentatonic angklung has four different scale intervals. Sound envelope analysis shows that angklung has attack time 192–179 ms and release time 346–902 ms. τe is a parameter that correlated with bandwidth frequency, lower value of τe correlated with wider bandwidth frequency. The value of τe from pentatonic angklung music is between 17 and 50 ms. The value was measured from six songs played by pentatonic angklung. Spatial analysis shows that the direction of sound headed to the left front side of angklung.

Session 5aMUb

Musical Acoustics: Digital Libraries for Speech and Singing

Annabel J. Cohen, Cochair
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Steven R. Livingstone, Cochair
Psych., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada

Contributed Papers

10:40

5aMUb1. A digital library to advance interdisciplinary research in singing. Annabel J. Cohen (Psych., Univ. of Prince Edward Island, 550 Univ. Ave., Charlottetown, PE C1A 4P3, Canada, acohen@upei.ca), Ichiro Fujinaga (Music, McGill Univ., Montreal, QC, Canada), Nyssim Lefford (Sound Music & Digital Media, Lulea Tech. Univ., Stockholm, Sweden), Theresa Leonard (Audio Recording, Banff Ctr., Banff, AB, Canada), George Tzanatakis (Comput. Sci., Univ. of Victoria, Victoria, BC, Canada), and Coralie Vincent (Developmental Phonet., CNRS, Paris, France)

In 2008, at the ASA/EAA symposium honouring pioneering scientist of singing, Johan Sundberg, the Advancing Interdisciplinary Research in Singing (AIRS) project was introduced as a major collaborative research initiative on singing [Cohen, Acoustics 08, Paris (2008), 3177–3182]. Over 70 collaborators around the world were to investigate singing from perspectives of development, education, and well-being. A digital library was to facilitate distant team members' work on the same data, such as examples from voice studios around the world, performance stages, playgrounds, public places, solos, groups, classrooms, intergenerational or multicultural choirs, therapeutic settings or new tests of singing skills [Vincent *et al.* PEVOC9 (2011)]. Plans also included tools for annotation and analysis along with relevant documents and images. The present progress report on this endeavor describes preliminary prototypes, stages of development, and the current functional implementation. It is noted that although singing is primarily an acoustic and auditory phenomenon, video records of the singer are highly valuable. Their benefit, however, must be weighed against challenges arising from ethical considerations and storage requirements. Issues of ownership and data sharing are also raised as are practical matters of choice of platform, storage, formats, backup, human resources, and long-term preservation.

11:00

5aMUb2. Acoustic differences in the speaking and singing voice. Steven R. Livingstone, Katlyn Peck, and Frank A. Russo (Psychology, Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, steven.livingstone@ryerson.ca)

Speech and song are universal forms of vocal expression that reflect distinct channels of communication. While these two forms of expression share a common means of sound production, differences in the acoustic properties of speech and song have not received significant attention. Here, we present evidence of acoustic differences in the speaking and singing voice. Twenty-four actors were recorded while speaking and singing different statements with five emotions, two emotional intensities, and two repetitions. Acoustic differences of speech and song were reported in several acoustic parameters, including vocal loudness, spectral properties, and vocal quality. Interestingly, emotion was conveyed similarly in many acoustic features across speech and song. These results demonstrate the entwined nature of speech and song, and provide evidence in support of the shared emergence of speech and song as a form of early proto-language. These recordings form part of our new Ryerson Audio-Visual Database of Emotional Speech and Song (RAVDESS) that will be freely released in 2013.

11:20

5aMUb3. The diverse environments multi-channel acoustic noise database: A database of multichannel environmental noise recordings. Joachim Thiemann (METISS, INRIA/IRISA, Campus Beaulieu, Rennes 35042, France, joachim.thiemann@irisa.fr), Nobutaka Ito (NTT Commun. Sci. Labs., Rennes, France), and Emmanuel Vincent (METISS, INRIA/IRISA, Rennes, France)

Multi-microphone arrays allow for the use of spatial filtering techniques that can greatly improve noise reduction and source separation. However, for speech and audio data, work on noise reduction or separation has focused primarily on one- or two-channel systems. Because of this, databases of multichannel environmental noise are not widely available. DEMAND (Diverse Environments Multi-channel Acoustic Noise Database) addresses this problem by providing a set of 16-channel noise files recorded in a variety of indoor and outdoor settings. The data was recorded using a planar microphone array consisting of four staggered rows, with the smallest distance between microphones being 5 cm and the largest being 21.8 cm. DEMAND is freely available under a Creative Commons license to encourage research into algorithms beyond the stereo setup.

11:40

5aMUb4. Variation of rhythm metrics in regional varieties of Acadian French. Wladyslaw Cichocki (French, Univ. of New Brunswick, P.O. Box 4400, Fredericton, NB E3B5A3, Canada, cicho@unb.ca), Sid-Ahmed Selouani, Alaidine B. Ayed, Catherine Paulin, and Yves Perreault (Gestion de l'information, Université de Moncton, Shippagan, NB, Canada)

This preliminary investigation studies variation of rhythm metrics in dialects of Acadian French spoken in New Brunswick (Canada). The aim is to determine whether regional and social factors are significant sources of this variation. Data are recordings of 140 speakers who represent five geographic regions, both genders and two age groups (20–30 and 40–55 years of age). Sound files were segmented manually; eight interval-based metrics were calculated. Mean metric scores indicate that regional varieties of Acadian French pattern with other dialects of French; these scores are similar to those of other "syllable-timed" languages. Regional differences are found for several metrics (%V, Delta C, VarcoC, nPVI-C, rPVI-C), although the five geographic regions are not all clearly distinguished by these metrics. A major pattern that emerges is regional variation in high-vowel devoicing and/or deletion. Analyses also show that social factors are significant sources of interspeaker variability: gender (VarcoV, nPVI-C, rPVI-C) and age (DeltaV, VarcoV). These results suggest a certain amount of complementarity between regional and social factors in their effects on rhythm metrics.

Session 5aNSa

Noise: Urban Noise and Modeling

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

Contributed Papers

9:00

5aNSa1. The Internet of sound observatories. Dick Botteldooren, Timothy Van Renterghem, Damiano Oldoni, Samuel Dauwe, Luc Dekoninck, Pieter Thomas, Weigang Wei, Michiel Boes, Ramanan Muthuraman, Bert De Coensel, Bernard De Baets, and Bart Dhoedt (Information Technol., Ghent Univ., St. Pietersnieuwstraat 41, Gent 9000, Belgium, dick.botteldooren@intec.ugent.be)

With the advance of electronics, sound level meters have become more powerful when it comes to analyzing and storing huge amount of measurements. In recent years, these devices have been hooked up to the internet and stream life data. In the IDEA project, the whole concept of a sound observatory is turned upside down by stripping the sensor nodes to their bare essential, and by migrating all logic and data storage to computing centers. This opens new opportunities in particular for long-term environmental sound monitoring and analysis. As unlimited computing power is available, more advanced analysis such as auditory scene analysis can be incorporated. In addition, new analysis methods and indicators can be deployed on the whole network of sound observatories using up-to-date software agent technology. As each observatory is a cheap plug-and-measure device without any buttons or display, participatory sensing becomes easy: citizens plug in their device and data streams to central servers and is displayed on a website of choice for the community. During the presentation, application cases in urban tranquil area, building site noise, wind turbine noise, and train noise monitoring, as well as noise mapping validation will be shown.

9:20

5aNSa2. Permanent noise and vibration monitoring as a valuable tool to the construction industry. Daniel Vaucher de la Croix and Christian Freneat (ACOEM, 200 Chemin des Ormeaux, Limonest 69578, France, daniel.vaucherdelacroix@acoemgroup.com)

The construction of large infrastructures in dense urban areas comes along with a number of environmental challenges. Roads, railways, subways, and large building constructions necessarily have a significant impact on residents as well as surrounding buildings. This is especially true when it comes to consider large projects duration, which generally counts in months and even years. In this context, noise and vibration induced by the construction activities are major sources of annoyance to the community and may also induce potential damages to the immediate surroundings. Both issues have thus to be properly monitored in order to reduce adverse effects on residents, help mitigating risks and prevent potential interruption of the construction site's activity which would increase the overall project cost. The proposed paper will focus on how available communication technologies as an essential added value to noise and vibrations measurements. Operational conditions and project managers' requirements for system deployment will be reviewed. Then, benefits to the different parties will be highlighted on the basis of recent practical situations where adequate measures could be taken in the right timing and kept the project running while minimizing its noise and vibration environmental impact.

9:40

5aNSa3. Modeling urban noise exposure and contribution of noise reflection against façades of buildings: Does correction matter? Quentin M. Tenailleau, Nadine Bernard, Sophie Pujol (Laboratoire chrono-environnement (UMR6249), Univ. of Franche-comté, CMC - Hopital St. Jacques, 2 place St. Jacques, Besançon 25030, France, quentin.tenailleau@univ-fcomte.fr), Daniel Joly, Hélène Houot (Laboratoire ThéMA (UMR6049), Univ. of Franche-Comté, Besançon, France), and Frédéric Mauny (Ctr. de Methodologie Clinique, Ctr. Hospitalier Universitaire, Besançon, France)

European noise directives advise to apply corrections when measuring and modeling noise levels close to a building in the aim of excluding the contribution of noise reflection against the façade. The advised +3 dB correction is still subject to discussion. In order to investigate the needed correction for an household exposure studies, a high definition noise model was used to estimate noise levels at 10,394 inhabitable buildings. Three buffers were used to sample area surrounding façades of buildings. The surfaces were defined between the following distances : (i) 0 and 2 m, (ii) 0 and 6 m, (iii) 2 and 6 m. No differences between the distribution structures were observed. Mean noise levels do not differ significantly between the buffers methods [respectively (i) 48.8$\pm$6.5 dB, (ii) 48.9$\pm$6.4 dB, (iii) 49.0$\pm$6.5 dB; $p > 0.05$]. Maximum noise levels differ significantly between the methods [respectively (i) 52.0$\pm$7.2 dB, (ii) 52.5$\pm$7.2 dB, (iii) 52.4$\pm$7.2 dB; $p < 0.05$]. These results show no or light differences between indices computed by the three sampling methods. They are in favour of no or at least a low correction value to deal with the contribution of noise reflection against the façade of a building.

10:00

5aNSa4. Urban traffic noise assessment by combining measurement and model results. Frits Van der Eerden, Freek Graafland, Peter Wessels, and Tom Basten (Acoust. and Sonar, TNO, Oude Waalsdorperweg 63, The Hague 2597 AK, Netherlands, frits.vandereerden@tno.nl)

A model based monitoring system is applied on a local scale in an urban area to obtain a better understanding of the traffic noise situation. The system consists of a scalable sensor network and an engineering model. A better understanding is needed to take appropriate and cost efficient measures, especially when changes to the local infrastructure are proposed. The monitoring system provides information about the sound level distribution in the area in time and place. This can be used to create dynamic noise maps or to characterize the soundscape in the area. Results of a field test of two weeks in an urban area of 400 by 200 m are used. Three different areas are considered: (1) the main road which is the major source for traffic noise, (2) a quiet street, and (3) a quiet courtyard. The noise level measurements near the main road are compared with the engineering model results. Next, with the use of actual source levels from the measurements, the sound levels in the quiet street and the quiet courtyard are calculated. By comparing the model results with measurements in these areas, the parameters in the model are updated to better reflect the actual situation.

10:20

5aNSa5. Outdoor sound propagation for high-speed moving sources. Didier Dragna, Philippe Blanc-Benon (Laboratoire de Mécanique des Fluides et d'Acoustique, Ctr. Acoustique, 36 Ave. Guy de Collongue, Ecully, France, philippe.blanc-benon@ec-lyon.fr), Estelle Bongini, and Franck Poisson (SNCF, Direction de l'Innovation et de la Recherche, Paris, France)

This paper deals with modeling of sources in motion in time-domain solvers. In the context of transportation noise, acoustic sources are complex. Indeed, they are in motion, and they are generally not compact. Equivalent point sources are often used to simplify the problem. Heuristic methods are then applied to handle acoustic propagation over complex sites. Besides, time-domain solutions of the linearized Euler equations have proved to be an attractive approach to study outdoor sound propagation, and can then be used to validate these models. However, point sources in arbitrary motion are difficult to account for in these approaches. Distributed volume sources can be used instead. First, influence of the spatial support of the source on the acoustic field is investigated. The case of a harmonic source moving at a constant speed is studied. Directivity of a non-compact source is shown to be dramatically different to the one of a point source. Then, simulations of a broadband moving source above an impedance ground surface in a three-dimensional geometry are presented, and ground effect is highlighted.

10:40

5aNSa6. The modeling and calculation of sound radiation from facilities with gas flowed pipes. Fabian Probst (Res. & Development, DataKustik GmbH, Gewerbering 5, Greifenberg 86926, Germany, info@datakustik.com)

Computer modeling of industrial facilities like chemical plants, refineries, or other production areas is the first and most important step in the calculation of sound exposure in the environment. The pipework with gas flows is often contributing relevant to the sound radiation of the complete facility. This radiation can be determined applying the methods described in technical papers like VDI 3733 and ISO 15664. On the basis of these descriptions a software tool was developed that allows to create pipework in 3D models with line sources and to calculate the sound propagation with methods like ISO 9613-2. The line sources are linked with the technical parameters like pipe cross section, flow rate, pressure, density, and temperature of the gas and material parameters of the pipe wall. The sound power emission from the pipe to the environment and the internal flow of sound power—linked to the next section of piping—is calculated on the basis of these parameters. The same technique is used to calculate the sound emission of cooling towers, electric and fuel driven motors, gears pumps, and other devices. This powerful technique allows creating sustainable models that can be adapted to different operation conditions with minimum time and effort.

11:00

5aNSa7. Reproduction of sound source directionality in reduced scale model. Aline Lisot (FEC - Engenharia Civil, UNICAMP, Rua Colombo, 5790, Bloco C67, Maringá, Paraná 87020900, Brazil, alinelisot@gmail.com) and Stelamaris R. Bertoli (FEC - Engenharia Civil, UNICAMP, Campinas, Brazil)

The study of acoustic phenomena through reduced scale models is a useful tool for predicting the acoustic performance of closed and outdoor environments. When using reduced scale models, begins a process of constructive adequacy details of the study environment and sound source characteristics, reproducing its level and directionality. It is presented in this paper, a study on the reproduction of sound source directionality with application in outdoor reduced scale model. The sound source is an electric power converting substation. The directionality of this source was calculated using sound pressure levels measured, in 1/3 octave frequency bands between 50 Hz and 12.5 kHz, in 24 points positioned on a circle located around of the sound source. The sound was recorded in a point outside the circle for later playback. For playback of the sound signal, the noise was emitted in each monitoring direction, considering in each one the directionality through the signal equalization. The reduced scale used in the model was 1:5 and, based on this, it was transposed the frequencies of the signal. It was concluded that the sound source used in the reduced scale can be an efficient tool for reproduction of the directionality characteristics of the source.

FRIDAY MORNING, 7 JUNE 2013 — 511CF, 9:00 A.M. TO 10:40 A.M.

Session 5aNSb

Noise and ASA Committee on Standards: Current U.S. and Canadian Noise Standards

Richard L. McKinley, Chair
Air Force Res. Lab., Wright-Patterson AFB, OH 45433-7901

Contributed Papers

9:00

5aNSb1. Evaluation of proposed ASTM standard to measure the normalized insertion loss of doors. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

There is currently no ASTM method for field measurement of the acoustical noise isolation specific to doors. Measurement of the overall noise reduction of the composite wall/door assembly or the Apparent Sound Transmission Class of the door can be attempted using the methods in ASTM E336, but these methods are not well suited to measuring doors. Doors typically have poorly defined source and receiving rooms, such as long and narrow corridors, stairwells, or outdoor spaces, which often do not meet the room requirements of ASTM E336. The sound fields are rarely diffuse, and the measurement locations are not well defined in the ASTM standard. An alternative insertion loss method (i.e., a comparison of the sound pressure levels with the door open and closed) was developed by MJM Acoustical Consultants (Michel Morin, "Research project on the noise isolation provided by access doors in multi-dwelling buildings," 1993), and a draft ASTM standard has been developed based on this method. A laboratory testing program has been designed evaluate the proposed method and investigate variations in the test method. The results of the laboratory testing program are presented.

9:20

5aNSb2. Validation of the CSA Z107,56 standard method for the measurement of noise exposure from headsets. Alberto Behar, Gabe Nespoli, Tristan Defrancesco-Loria, and Frank Russo (Psych., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, albehar31@gmail.com)

The last section of the CSA Standard CAN/CSA-Z107.56-06 (R2011)—Procedures for the Measurement of Occupational Noise Exposure—deals with communication headsets. This section is typically applied in retail, call centers, airport control towers, and other settings where an operator is exposed to background noise while communicating with a headset. Because the user adjusts the volume to hear speech comfortably, the total equivalent sound level is a function of the level of the background noise minus the attenuation provided by the headset, plus a comfortable signal/noise ratio, estimated to be 13.5 dBA. Despite ample research demonstrating the role of spectral and temporal factors in speech masking, the standard is applied consistently regardless of the type of background.

9:40

5aNSb3. Comparison of direct measurement methods for headset noise exposure in the workplace. Flora Nassrallah (Population Health, Univ. of Ottawa, 1 Stewart St., Ottawa, ON K1N 6N5, Canada, fnass039@uottawa.ca), Christian Giguère (Audiol./SLP Program, Univ. of Ottawa, Ottawa, ON, Canada), and Hilmi R. Dajani (School of Elec. Eng. and Comput. Sci., Univ. of Ottawa, Ottawa, ON, Canada)

Specialized equipment and techniques are required to carry out direct sound measurements under occluded ears for the purpose of assessing the noise exposure from communication headsets. Standard ISO 11904 describes two procedures: (1) the microphone in a real ear (MIRE) and (2) the acoustic manikin technique using an occluded ear simulator IEC 60318-4. Methods using simpler artificial ears, such as IEC 60318-1, have also been proposed in occupational noise measurement standards such as CAN/CSA Z107.56. Such devices are more practical to use and more easily accessible. However, they have not been designed specifically for noise measurements under communication headsets and there is little comparative data to the manikin technique, which is considered the gold standard for simulated in-situ acoustic measurements. Furthermore, little is known about measurement reliability for the purpose of standardization. Fit-refit measurements were obtained under laboratory conditions with four different types of artificial ears (type 1, type 2, type 3.3, manikin), three communication headset types (circum-aural, supra-aural, intra-aural) and six different communication signals. Data was transformed into equivalent-diffuse sound levels using third-octave procedures as well as single number corrections. Results illustrate that methods using single number corrections as well as fit-refit standard deviations vary according to measurement conditions.

10:00

5aNSb4. Evaluating harmful acoustic impulses. Gerald Fleischer (Justus-Liebig-Univ., Hoehenstr. 18, Giessen 35466, Germany, gerald.fleischer@gmx.net)

Noise-induced auditory impairment is widespread, despite many successful efforts to reduce sound emitted by technical equipment. Using accumulated acoustic energy as an indicator for health hazard, as in ISO1999, is insufficient to describe the hazard for impulses. While stimulating a system with many vibrating parts, like the ear, it is more important how the energy is fed into the various components than how much is applied. The Human Ear Model (AHAAH) uses the pressure-time-history of impulses, which is much better. It is based on theoretical assumptions, but it has serious deficits that lead to results that can be extremely wrong. Within another long-term approach the effects on hearing of accidents caused by impulses have been collected and documented. The harmful situations were reenacted thoroughly, to obtain acoustic measurements that can be correlated with the auditory damage, if present. All told, impulses cause three types of notches, at different frequency ranges. Hence, harmful impulses show characteristic footprints in carefully recorded audiograms, and the type of notch depends upon details of the causing impulse. This EPI-approach (Effects of Powerful Impulses) applies to most cases of auditory damage, and it is useful to prevent and understand such injuries.

10:20

5aNSb5. Experience with developing a room to meet the requirements of the joint Australian/New Zealand standard 1270. Ben Scott and John Pearse (Mech. Eng., Univ. of Canterbury, Ilam, Christchurch 8041, New Zealand, ben.scott@pg.canterbury.ac.nz)

Hearing protectors are used as a “last line of defence” to protect people in high noise exposure environments from noise induced hearing loss. The assessment of hearing protector performance has been standardized to ensure a minimum level of performance for the end user. In New Zealand, the standard AS/NZS 1270 specifies a REAT based test method to rate a hearing protector’s performance. The rating is used to calculate appropriate exposure times for end users. This paper describes the modification of an existing audiology booth, to meet the sound field requirements specified in AS/NZS 1270. Current status of the facility will be reported; followed by a general discussion on future uses of the facility and related projects at the University of Canterbury.

FRIDAY MORNING, 7 JUNE 2013 — 519B, 8:55 A.M. TO 11:40 A.M.

Session 5aPA

Physical Acoustics: Chemical and Non-Medical Biological Effects of Ultrasound

Kenneth Suslick, Cochair
Chemistry, Univ. of Illinois, 600 S. Mathews Av., Urbana, IL 61801

Hao Feng, Cochair
Univ. of Illinois at Urbana-Champaign, 1304 W Pennsylvania Ave., Urbana, IL 61801

Chair’s Introduction—8:55

Invited Papers

9:00

5aPA1. Sonofragmentation of molecular crystals. Kenneth Suslick and Brad W. Zieger (Chemistry, Univ. of Illinois, 600 S. Mathews Av., Urbana, IL 61801, ksuslick@uiuc.edu)

Developing processes for the production of active pharmaceutical ingredients (APIs) with a specific crystal size or polymorph distribution is critical for improved drug delivery by aerosolization, injection or ingestion, for control of bioavailability, and for economy of preparation. The use of ultrasound for the crystallization of APIs has attracted substantial recent attention due to (1) its influence on particle size and size distribution, (2) reduction of metastable zone-width, induction time, and supersaturation levels required for nucleation,

(3) improved reproducibility of crystallization, (4) control of polymorphism, and (5) reduction or elimination of the need for seed crystals or other foreign materials. Possible mechanisms for the breakage of molecular crystals under high-intensity ultrasound were investigated, using acetylsalicylic acid (aspirin) crystals as a model compound for active pharmaceutical ingredients (APIs). Surprisingly, kinetics experiments rule out particle-particle collisions as a viable mechanism for sonofragmentation. Two other possible mechanisms — particle-horn or particle-wall collisions — were dismissed based on decoupling experiments. Direct particleshockwave interactions are therefore indicated as the primary mechanism of sonofragmentation of molecular crystals.

9:20

5aPA2. Sonochemical synthesis of nano-cocrystals. Leonard R. MacGillivray, Dejan-Kresimir Bucar, John R. Sander, and Elizabeth Elacqua (Chemistry, Univ. of Iowa, E555 Chem. Bldg., Iowa CIty, IA 52242, len-macgillivray@uiowa.edu)

Cocrystals are multicomponent solids with organic molecules assembled in combination to form a crystalline solid with properties different than the individual components. A cocrystal typically consists of a target molecule crystallized with a second molecule, or cocrystal former, employed to influence properties of the target (e.g., solubility). The conformer interacts with the target via intermolecular forces (e.g., hydrogen bonds) that hold the components together. The modularity of a cocrystal makes such solids attractive for applications where fine-tuning of properties is important (e.g., optical). In this presentation, we describe the use of sonochemistry to form cocrystals of nanoscale dimensions. In contrast to single-component solids, cocrystals present a fundamentally different challenge with respect to those reprecipitation methods used to form nanocrystals since the components of a cocrystal will tend to exhibit different solubilities. We show that sonochemistry affords nano-cocrystals with properties (e.g., reactivity) that contrast solids of macroscale dimensions. Related applications of sonochemistry to afford single-component nanocrystals will also be presented.

9:40

5aPA3. Enhancing biofuel production by ultrasonics. David Grewell, Melissa Montalbo-Lomboy, and Priyanka Chand (Iowa State Univ., 100 Davidson Hall, Ames, IA 50011, dgrewell@iastate.edu)

This work evaluated the use of high-powered ultrasonics to enhance biofuel production in terms of efficiency and costs. A wide range of feed stocks, including switch grass, corn stover, and soft wood, were studied. The effect of ultrasonic pretreatment on the removal of lignin for hydrolysis of starches and cellulose to fermentable sugars was studied. It was found that many of the pretreatments were very successful in enhancing lignin removal. For example, time of dissolution of lingo-cellulosic biomass in ionic liquids was reduced from hours to minutes accompanied by a significant decrease in energy consumption compared to mechanical stirring. In addition, it was found that hydrolysis of corn starch could be greatly accelerated utilizing ultrasonics. Economic models showed that the technology, once implemented, would have a payback period of less than one year. The work also focused on biodiesel production. It was seen that ultrasonics accelerated the transesterification process so that soy bean oil could be converted to biodiesel in less than a minute, compared to 45 min using traditional methods. It was shown that yeast grown from glycerin, a co-product of biodiesel production, could be extracted and simultaneously converted to biodiesel with ultrasonics in less than a minute, compared to traditional techniques that require multiple processes and relatively long cycle times (+1 h).

10:00

5aPA4. The influence of ultrasound on the structure, rheological properties, and degradation path of citrus pectin. Donghong Liu (Fuli Inst. of Food Sci., Zhejiang Univ., No. 866 Yuhangtang Rd., Hangzhou 310016, China, dhliu@zju.edu.cn) and Lifeng Zhang (Food Sci. and Nutrition, Zhejiang Univ., Hangzhou, China)

The effects of ultrasound on the molecular weight, structure, and rheological properties of citrus pectin were investigated. The degradation path of citrus pectin by ultrasound was also studied. The structure and rheological properties of the degradation products were identified by high performance liquid chromatography-photodiode array detector (HPLC-PAD), Fourier transform infrared spectroscopy (FTIR), nuclear magnetic resonance spectroscopy (NMR), atomic force microscope (AFM), and rheometer. The results indicated that the average molecular weight of citrus pectin decreased rapidly after ultrasound treatment and reduced to one third of the initial pectin after treated for 90 min. The polydispersity reduced from 2.30 to 1.59. The degradation products had a uniform and narrow distribution of molecular weight. The main chain composition and monosaccharide constituents of citrus pectin remained unchanged after ultrasound treatment. The reduction ratio of (Gal+Ara): Rha suggested a decrease in neutral sugar side chain size of citrus pectin after ultrasonication. FT-IR and NMR results approved that the main chain of citrus pectin was not changed by ultrasound treatment. Together with the AFM results indicated that ultrasound could reduce the branched structure of citrus pectin. The viscosity of citrus pectin decreased after ultrasound treatment. Meanwhile, the ultrasound-treated citrus pectin showed predominantly viscous responses ($G' < G''$) over the same frequency range.

10:20

5aPA5. Free radical formation and scavenging by solutes in the sonolysis of aqueous solutions. Franz Grieser (Chemistry, Univ. of Melbourne, Grattan st, Parkville, VIC 3010, Australia, franz@unimelb.edu.au)

It has long been known that the primary radicals generated in water (H and OH), on the collapse of acoustic bubbles, largely recombine. It has been estimated that as much as 90% react within the bubble to produce molecular hydrogen, hydrogen peroxide and water [Henglein, Ultrason. Sonochem.**2**, S115–S121 (1995)]. This high recombination efficiency has been likened to radicals reacting within "spurs" produced by ionizing radiation in water. Several studies have shown that by using high concentrations (100s of mM) of primary radical scavengers, e.g., aliphatic alcohols, iodide, etc., a large number of the primary radicals are able to be captured in acoustically produced hot spot spurs. What is less well examined is the effect the scavengers themselves have on the production of the primary radicals and hence on the radical yields measured. The talk will consider the effect that typical radical scavengers have on active bubble populations in aqueous solutions, and on the production of primary radicals in the presence of added solutes.

10:40

5aPA6. Microbial inactivation by ultrasound for enhanced food safety. Hao Feng, Bin Zhou, Hyoungill Lee, Yanfang Li, Hee Kyung Park, Sindy Palma, and Arne Pearlstein (Univ. of Illinois at Urbana-Champaign, 1304 W Pennsylvania Ave., Urbana, IL 61801, haofeng@illinois.edu)

Inactivation of foodborne pathogens by power ultrasound provides an alternative to traditional thermal processing modalities, with potential for minimizing food-quality degradation. To enhance efficacy, ultrasonic treatment is often combined with other physical or chemical lethal factors, which serve to shorten treatment time and improve quality retention. The inactivation mechanisms, thermodynamic aspects, and kinetic modeling of ultrasonic microbial inactivation will be discussed. The critical issue of how to achieve a relatively uniform acoustic field distribution during treatment will be investigated by computer simulation and verified with microbial inactivation tests. Inactivation of foodborne pathogens in liquid foods, and surface decontamination of fresh produce and nuts, will be used as examples demonstrating the potential of ultrasound-assisted processes. Lastly, the effect of sonication treatment on food product quality and quality retention will be examined.

Contributed Papers

11:00

5aPA7. Effects of frequency and initial concentration of methylene blue on rate constants of ultrasonic degradation. Chiemi Honma (Dept. of Chemical Sci. and Technol., Tokyo Univ. of Sci., Shinjyukuku, Tokyo 162-8601, Japan, jb112864@ed.tus.ac.jp), Daisuke Kobayashi (Dept. of Industrial Chemistry, Tokyo Univ. of Sci., Tokyo, Japan), Hideyuki Matsumoto (Dept. of Chemical Eng., Tokyo Inst. of Technol., Tokyo, Japan), Tomoki Takahashi (Dept. of Industrial Chemistry, Tokyo Univ. of Sci., Tokyo, Japan), Chiaki Kuroda (Dept. of Chemical Eng., Tokyo Inst. of Technol., Tokyo, Japan), Katsuto Otake, and Atsushi Shono (Dept. of Industrial Chemistry, Tokyo Univ. of Sci., Tokyo, Japan)

Ultrasound has been found to be an attractive advanced technology for the degradation of hazardous organic compounds in water. The ultrasonic degradation of dyes has been investigated by many researches, but the effects of ultrasonic frequency on degradation rate were not investigated quantitatively. In our previous study, we proposed a simple model for estimating the apparent degradation rate constant of methylene blue based on ultrasonic power and the SE value in the range of frequency between 20 and 500 kHz. However, we have not investigated the ultrasonic degradation of methylene blue in high frequency region around 1 MHz. The effects of initial concentration of methylene blue have not been investigated yet, either. In this study, the degradation process using methylene blue as model hazardous organic compounds by ultrasonic irradiation was investigated. The ultrasonic frequency was operated in the range from 20 kHz to 1.6 MHz, and the initial concentration of methylene blue was in the range from 0.005 to 0.04 mM. Our proposed model can apply to this study which extended the frequency range. The effects of initial concentration can also be estimated in the range of 0.01 to 0.04 mM using this model.

11:20

5aPA8. Effect of gases on radical production rates during single-bubble cavitation. Shin-ichi Hatanaka (Dept. of Eng. Sci., The Univ. of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan, hatanaka@pc.uec.ac.jp)

The yields of hydroxyl radicals and nitrite ions produced from single-bubble cavitation were quantified while the corresponding dynamics of the single bubble was observed by stroboscopic and laser-light scattering methods. The numbers of both hydroxyl radicals and nitrite ions per cycle were proportional to pressure amplitude at 25 kHz under the conditions of stable sonoluminescing single-bubbles. Under the dancing single-bubbles without sonoluminescence below sonoluminescence threshold for air bubbles, however, the number of hydroxyl radicals was larger than that for the stable single-bubble and nitrite ions were not detected in contrast to hydroxyl radicals. For argon bubbles, the numbers of hydroxyl radicals were larger than those for air bubbles and the number of hydroxyl radical was significantly increased with the dancing bubble by argon bubbling. The results imply that the shape instability may promote the dissolution and the diffusion of hydroxyl radicals from the bubble into the liquid and the part of hydroxyl radicals may convert into nitrite ions in the case of the air bubble above the threshold of sonoluminescence.

Session 5aPP

Psychological and Physiological Acoustics: Recent Trends in Psychoacoustics I

Hugo Fastl, Cochair
AG Technische Akustik, TU München, Arcisstr.21, München 80333, Germany

Sonoko Kuwano, Cochair
Osaka Univ., 2-24-1-1107 Shinsenri-Nishimachi, Toyonaka, Osaka 560-0083, Japan

Chair's Introduction—8:55

Invited Papers

9:00

5aPP1. Evaluation of the loudness of stationary and non-stationary complex sounds. Sonoko Kuwano (Osaka Univ., 2-24-1-1107 Shinsenri-Nishimachi, Toyonaka, Osaka 560-0083, Japan, kuwano@see.eng.osaka-u.ac.jp), Tadasu Hatoh (Toho Gakuen, College of Drama and Music, Yokohama, Japan), Tohru Kato (Otemon Gakuen Univ., Osaka, Japan), and Seiichiro Namba (Osaka Univ., Osaka, Japan)

In our sound environment, there are various complex sounds including both stationary and non-stationary sounds. It is important to evaluate the loudness of these sounds, to clarify the relation between the loudness and the physical metrics and to estimate the loudness by the physical metrics in order to evaluate and control the sound environment. The temporal patterns of non-stationary sounds in daily life environment, such as road traffic noise, construction noise, speech and music, are different from each other. It would be desirable to evaluate the loudness of both stationary and non-stationary sounds with various temporal and spectral characteristics by the same metric in the evaluation of the loudness. It is needless to say that the metric should have a good relation with the subjective impression. From physical viewpoint, sounds with various temporal patterns including stationary sounds can be measured by a single scale on the basis of energy. In this paper, a single common metric for the evaluation of the loudness of both stationary and non-stationary sounds is examined by conducting psychological experiments.

9:20

5aPP2. Loudness of complex time-varying sounds – A challenge for current Loudness models. Jan Rennies (Hearing, Speech and Audio Technol., Fraunhofer IDMT, Marie-Curie-Str. 2, Oldenburg 26129, Germany, jan.rennies@idmt.fraunhofer.de), Jesko L. Verhey (Experimental Audiol., Otto von Guericke Universität, Magdeburg, Germany), Jens E. Appell (Hearing, Speech and Audio Technol., Fraunhofer IDMT, Oldenburg, Germany), and Birger Kollmeier (Med. Phys., Carl-von-Ossietzky Universität, Oldenburg, Germany)

The calculation of perceived loudness is an important factor in many applications such as the assessment of noise emissions. Generally, loudness of stationary sounds can be accurately predicted by existing models. For sounds with time-varying characteristics, however, there are still discrepancies between experimental data and model predictions, even with the most recent loudness models. This contribution presents a series of experiments in which loudness was measured in normal-hearing subjects with different types of realistic signals using an adaptive loudness matching procedure and categorical loudness scaling. The results of both methods indicate that loudness of speech-like signals is largely determined by the long-term spectrum, while other speech-related properties (particularly temporal modulations) play only a minor role. Loudness of speech appears to be quite robust towards even severe signal modifications, as long as the long-term spectrum is similar. In contrast, loudness of technical, strongly impulsive signals is considerably influenced by temporal modulations. For some of the signals, loudness could not be predicted by current models. Since the perceived loudness was underestimated by the models for some signals, but overestimated for other signals, a simple adjustment of the employed time constants in the temporal integration stage could not eliminate the discrepancies.

9:40

5aPP3. Differentiating between loudness and preference in the case of multi-tone stimuli. Stephan Toepken and Reinhard Weber (Acoust. Group, Oldenburg Univ., Carl-von-Ossietzky-Str.9-11, Oldenburg 26129, Germany, stephan.toepken@uni-oldenburg.de)

When exploring sound quality, often a high correlation between pleasantness and loudness can be observed. However, sometimes it is desirable to know to which extent other sound characteristics than loudness are responsible for a preference evaluation. In this respect multi-tone sounds with rich perceptual aspects are interesting test sounds. This talk will present a separate determination of preference and loudness by comparing a test sound of interest with a reference sound. Using an adaptive paired comparison the points of subjective equality (PSEs) for preference and loudness between test and reference sound are separately measured—"Which sound is louder?" and "Which sound do you prefer?"—by varying the test sound level in an adaptive staircase manner. The level changes affect both loudness and preference evaluation of the test sound. The results of these experiments are level differences ΔL between the test and the reference sound at which equal preference and equal loudness are reached between them. (Similar procedures have been employed to determine equal loudness contours.) It will be shown, with multi-tone sounds as examples, how this method reliably differentiates between loudness and preference.

10:00–10:20 Break

10:20

5aPP4. Suprathreshold perception under a masking release condition using categorical scaling. Jesko L. Verhey and Wiebke Heeren (Dept. of Experimental Audiol., Otto von Guericke Univ. Madgeburg, Leipziger Str. 44, Magdeburg 39120, Germany, jesko.verhey@med.ovgu.de)

Masking experiments provide important information on how the auditory system processes sounds. For example, a tone is less masked by a modulated sound than by an unmodulated sound with the same long-term spectrum, indicating the ability of the auditory system to use modulation as a cue. In general, studies on this modulated-unmodulated difference (MUD) focus on the thresholds, whereas little is known about suprathreshold perception under these conditions of masking release. In the present study, loudness growth functions of a masked 1000-Hz tone are measured for two different masker types: (i) amplitude modulated broadband noise with a square-wave modulator and (ii) an unmodulated noise with the same spectral content and level. A categorical loudness scaling procedure (ISO 16832) is used to measure loudness of the masked tone over a large level range. The accuracy of the procedure is quantified by comparing the scaling results with loudness matching data for the same masker types. It is investigated (i) up to which suprathreshold level a masking release is still observed and (ii) whether the effect of the reduced masking for the modulated masker is equivalent to a condition where the unmodulated masker is reduced in level by the magnitude of the MUD.

10:40

5aPP5. Comparison of detection threshold measurements and modeling for approaching electric cars and conventional cars presented in traffic and pink noise. Julian Grosse, Reinhard Weber, and Steven van de Par (Acoust. Group, Univ. of Oldenburg, Carl-von-Ossietzky-strasse 9-11, Oldenburg 26129, Germany, steven.van.de.par@uni-oldenburg.de)

This study investigates the difference in audibility of an approaching conventional car with internal combustion engine and an electric car at various velocities. The goal was to compare the risk that pedestrians do not hear the approaching car in time. Binaural recordings of each of these approaching cars were presented together with either a traffic noise masker or a pink noise masker. In the first detection experiment, the threshold level was determined for which the cars could just be detected. In a second reaction time experiment, the moment was determined at which the approaching car was first detectable. This measured reaction time should give an indication about how much time a person has to evade an impending collision. Results indicated that slowly approaching electric cars where less audible than cars with a conventional engine. The results also showed that the decrement of reaction times as a function of SNR was halved when pink noise was used instead of traffic noise. A psycho-acoustic masking model [Dau *et al*., J. Acoust. Soc. Am. **99**, 3615–3622 (1996)] was applied to predict detection thresholds and showed good correspondence with the subjective data.

11:00

5aPP6. Rating the dieselness of vehicle noise using different psychoacoustic methods. Jakob Putner and Hugo Fastl (AG Technische Akustik, MMK, Technische Universität München, Arcisstraße 21, Munich 80333, Germany, putner@tum.de)

Modern diesel engines meet the demand for high power-engines while strict emission regulations have to be fulfilled. Therefore, diesel engines entered vehicle segments where the expectations on the sound quality are exceptionally high. Sound quality and fuel efficiency are often conflicting goals during the development of a diesel engine. The typical sound character of diesel engines, the so called Dieselness, is an indicator for the overall sound quality of the vehicle noise. Hence, it is desirable to rate the Dieselness of engine sounds. Sounds emitted by gasoline- and diesel-powered vehicles in idle condition were rated in psychoacoustic experiments using different methods. First, the method of line length was used as direct scaling procedure to get ratio ratings of the relative Dieselness of the vehicle noises. Second, a direct ranking of the noises has been done with the Random Access method where subjects had to rank the sounds according to their Dieselness. Third, in a paired-comparison test the participants had to judge which of two sounds had more Dieselness, resulting in an indirect scaling. These methods are compared regarding the time the experiments took and the resulting ranking respectively scaling. In addition, a semantic differential test with general adjective pairs was conducted.

11:20

5aPP7. Fluctuation strength on real sound: Motorbike exhaust and marimba tremolo. Masanobu Miura (Dept. of Media Informatics, Ryukoku Univ., 1-5, Seta, Oe-cho, Yokotani, Otsu 5202194, Japan, miura@mail.ryukoku.ac.jp) and Nozomiko Yasui (Dept. of Information Eng., Matsue College of Technol., Matsue, Japan)

Psychoacoustics research has been contributed on evaluating the timbre of acoustic signals. The fluctuation strength (FS) has been respected as the important index which describes the sensory fluctuation of the acoustic signal based on its amplitude, frequency and the combination of them as well. Although FS has been calculated by tremor components on acoustic signals, here points out not only the inappropriateness of the calculation when simply based on literatures but also a method which focuses on the shape of the waveform in order to extract parameters from it, so that the FS is newly calculated by our original method with estimation results of the subjective scores of sensory fluctuation. Signals dealt with are motorbike exhaust sounds and tremolo played by marimba. The developed method is then applied to the design of an electric vehicle approaching sound for pedestrians in order to let pedestrians notice the approaching car without rising up the sensory loudness, on which designed sounds have a fluctuation with irregular pulses proven to give us the sensory fluctuation so that it is expected to be noticeable for pedestrians. This paper discusses the possibility to apply the prestigious psychoacoustic indexes to industrial and artistic sounds.

11:40

5aPP8. Perception of roughness of time-variant sounds. Roland Sottek and Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

Besides loudness, other psychoacoustic parameters like sharpness and roughness, are important for sound quality evaluation. Sharpness considers the relationship between the loudness of high frequency components to total loudness, and roughness evaluates modulation characteristics. While loudness of stationary sounds has been standardized for decades, standards for sharpness of stationary sounds and for loudness of time-varying sounds have been published in 2009 (DIN 45692:2009-08) and 2010 (DIN 45631/A1:2010-03),

respectively. In addition, there are several roughness models available, performing more or less well for synthetic and selected technical signals. Currently, a roughness standard is under discussion in a DIN working group. In recent listening tests, the subjects showed consistent overall roughness evaluation of synthetic signals but heterogeneous judgments concerning the more complex technical sounds containing different rough components. The results of the listening test will be discussed and compared with the evaluation by several models like the roughness calculation using the hearing model of Sottek and another approach based on a psychoacoustically weighted modulation spectrum.

FRIDAY MORNING, 7 JUNE 2013 | 512CG, 9:00 A.M. TO 12:00 NOON

Session 5aSA

Structural Acoustics and Vibration: Applications in Structural Acoustics and Vibration IV

Kuangcheng Wu, Cochair
Ship Survivability, Newport News Shipbuild, 4101 Washington Ave., Newport News, VA

Vyacheslav Ryaboy, Cochair
Newport Corp., 1791 Deere Ave., Irvine, CA 92606

Contributed Papers

9:00

5aSA1. Splitting up resonances of elastic elliptical disc. Emmanuelle Bazzali, Stéphane Ancey, Paul Gabrielli, and Michaël Mercier-Finidori (Physics, UMR CNRS 6134 SPE Université de Corse, Campus Grimaldi, Bâtiment Alfonsi, Corte, Corsica 20250, France, ebazzali@gmail.com)

The elastodynamic resonances of two-dimensional elliptical objects are studied from a modal formalism by emphasizing the role of the symmetries as the circumference is deformed from circular to elliptical geometry. More precisely, as the symmetry is broken in the transition from the circular disc to the ellipse, resonance splittings and level crossings are observed. This observation can be mathematically explained by the broken invariance of the continuous O(2) symmetry group of the circular disc. However, the ellipse remains invariant under the finite C2v group. The main difficulty comes from the application of the group theory in elastodynamics since the vectorial formalism is used to express the physical quantities involved in the boundary conditions. This method significantly simplifies numerical computations and provides a full classification of the resonances. The vibrational normal modes are computed. We focus on the resonance splittings in the transition from the circular disc to the elliptical one. Then, the resonances are tagged and tracked as the eccentricity of the ellipse increases. A series of experiment on three-dimensional objects are also carried out to emphasize the physical effects described above, although no quantitative comparison can be done between theory and experiment. We expect that those effects in 2D appear also in 3D when the sphere is deformed to the spheroid.

9:20

5aSA2. Extension of SmEdA to non-resonant transmission. Laurent Maxit, Kerem Ege, Nicolas Totaro, and Jean-Louis Guyader (INSA LYON, 25 bis av. Jean Capelle, Villeurbanne 69621, France, laurent.maxit@insa-lyon.fr)

Statistical modal Energy distribution Analysis (SmEdA) may be used as alternative to Statistical Energy Analysis for describing subsystems with low modal overlap. In its original form, SmEdA predicts the power flow exchanged between the resonant modes of different subsystems. In the case of the sound transmission through a thin light structure, it is well-known than the non-resonant response of the structure may have a significant role on the transmission below the critical frequency. In this paper, one presents an extension of SmEdA taking into account the contributions of the non resonant modes of the thin structure. The dual modal formulation (DMF) is used for describing the behavior of two acoustic cavities separated by a thin structure knowing their subsystem modes. A condensation in the DMF equations is achieved on the amplitudes of the non-resonant modes. Using some simplifications, a new coupling scheme between the resonant modes of the three subsystems is obtained. It shows direct couplings of the cavity modes through stiffness elements characterized by the modes shapes of the cavities and the structure, both. Comparisons with reference results show the ability and the interest of the present approach for representing the non-resonant contributions of the structure.

9:40

5aSA3. On the energy finite element method for the acoustic design of ships. Bernd Stritzelberger, Martin Abele, and Otto von Estorff (Inst. of Modeling and Computation, Hamburg Univ. of Technol., Denickestr. 17, Hamburg 21073, Germany, bernd.stritzelberger@tuhh.de)

To ensure dynamic requirements of technical systems, methods like the finite element method (FEM) are successfully applied. For large structures as ship geometries, such analyses in the acoustic-relevant frequency range are usually not used productively. Highly time consuming investigations are incompatible to the generally single-unit production and short conception phases in ship design. The energy finite element method (EFEM) is a grid-based approach, which has the potential to provide a technique for the evaluation of acoustic characteristics even for major and complex structures at high frequencies. The less time consuming calculations generally result from a smaller number of degrees of freedom at the nodes and, in particular, it is feasible to use coarser grids than in the FEM. The governing equations are similar to that of the static heat conduction. State variables are the time and locally space averaged energy densities of the different wave types. The main focus is on the coupling— not only between the structure and the fluid, but also at junctions within the structure. Preliminary investigations on the reliability of EFEM results will be presented, questioning if the approach is applicable to operative ship design. [This work was done within the collaborative research project EPES.]

10:00

5aSA4. Simulating sound radiation using the energy-finite-element-method. Marius Karger, Otto von Estorff, and Olgierd Zaleski (Novicos GmbH, Kasernenstrasse 12, Hamburg 21073, Germany, karger@novicos.de)

The most established numerical methods for calculation of sound radiation are the boundary-element-method (BEM) and the finite-element-method (FEM). For large-scale geometries and high-frequency ranges these methods are limited by enormous numerical costs. The applicability of the

energy-finite-element-method (EFEM) in these cases is analyzed within the research project EPES, sponsored by the Federal Ministry of Economy and Technology. Under certain assumptions, the equations describing structure-borne sound and sound radiation can be condensed to the static heat conduction equation, transforming the pressure and velocities in energy densities. Using EFEM, the structure geometry and acoustic cavities are separately modeled and coupled by transmission coefficients for energy flow interactions. An important value calculating the coefficients is the radiation efficiency. This paper focuses on the analysis of the radiation efficiency for EFEM calculations. This contribution presents the EFEM approach, calculations of radiation efficiency, transmission coefficients and energy densities of different fluid-structure interactions. Based on those calculations, the applicability of the EFEM is discussed.

10:20

5aSA5. Validity of transfer matrix method for prediction of the transmission loss of curved panels. Mejdi Abderrazak, Sgrad Franck (bruit et Vibration, Institue de recherche de robert sauvé en santé et sécurité de travail (IRSST), 505, boul. De Maisonneuve Ouest, Montréal, QC H3A 3C2, Canada, abderrazak.mejdi@usherbrooke.ca), and Atalla Noureddine (Mechanical, Université de Sherbrooke, Sherbrooke, QC, Canada)

This paper discusses the modeling of the transmission loss of curved panels with attached sound absorbing materials (foam or fiber) using both analytical and numerical methods. Special attention is devoted to the validity of modeling the problem using the transfer matrix method (TMM) and statistical energy analysis (SEA). Classically, in SEA models the sound package is unwrapped and the TMM is used to calculate its effects in terms of added damping, absorption and insertion loss. A systematic comparison with an efficient FEM/VBEM formulation of the problem is presented to examine validity of this practice and demonstrate its range of applicability and usefulness.

10:40

5aSA6. The analytical model of rheological fluid for vibration and noise control. Marek L. Szary (College of Eng., Southern Illinois Univ., Carbondale, IL 62901, szary@engr.siu.edu)

Rheological fluids (RF) also known as a controllable viscosity fluids (CVF) introduced in the area of vibration control of mechanical systems made possible a more efficient control of both: transient and continuous vibration. They are also used in design of sound barriers to control noise transmission loss and diaphragms for modification of noise absorption characteristics of sound absorbing materials. Their apparent viscosity is controllable by the use of external (electrical in electro-rheological ER or electromagnetic in magneto-rheological MR fluids) field. In the absence of applied external field, the RF exhibits Newtonian-like behavior. Applied external field changes this behavior and the RF shows in addition a yield shear stress which depends on strength of this field. In proposed model, the shear stress is expressed as a superposition of two components. One of them is proportional to the viscosity and relative velocity of a base fluid, and the second one, which depends on strength of applied external field. Modification of external field strength according to selected design parameter (velocity or time or distance—combination of them) allows to develop family of vibration or sound attenuating devices, which were not achievable before.

11:00

5aSA7. Multi-component power transmission from structure-borne sound sources into lightweight structures. Sebastian Mathiowetz and Hannes A. Bonhoff (Tech. Univ. of Berlin, Einsteinufer 25, Berlin 10587, Germany, s.mathiowetz@tu-berlin.de)

The power transmission between structure-borne sound sources and adjacent structures is generally of complex nature. For an accurate description, the interaction between multiple contact points and several directional components must be taken into account. While calculation methods to predict the transmitted power are generally available, the main problem is the acquisition of extensive source and receiver data. This is especially true with regard to lightweight structures where source and receiver mobilities exhibit matched conditions and when rotational components of motion are involved. Therefore, the description needs to be simplified while at the same time a sufficient accuracy has to be retained. This work investigates the power transmission of a fan unit source that is mounted to a rib-stiffened aluminium plate at several contact points. Full data sets of source and receiver have been measured using a finite difference technique, including translational motion perpendicular to the structure as well as moment excitation around the in-plane axes of the plate. Both a rigid connection as well as a connection using resilient mounts is considered. Contributions of different components of motion are discussed and possible simplifications are deduced.

11:20

5aSA8. A low density, high stiffness flat loudspeaker with combined feedback-feedforward response correction. Jen-Hsuan Ho (Signals and Systems Group, Faculty of EEMCS, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, j.ho@ewi.utwente.nl) and Arthur Berkhoff (Acoust. and Sonar, TNO Tech. Sci., Den Haag, Netherlands)

This paper presents a novel perforated flat loudspeaker with improved sound frequency response by applying velocity feedback control. Flat loudspeakers provide advantages of compact dimensions and high durability. Known flat loudspeaker technology is based on high model density. However, the resonances in the panel are complex and difficult to control, which often leads to complicated computations and insufficient low frequency response. The flat loudspeaker in this paper comprises a novel panel structure, which offers low density, high stiffness, and efficient space utilization. Direct velocity feedback control provides a simple and stable control loop. We apply the multiple direct velocity feedback control method to obtain flat sound frequency response. Furthermore, a Linkwitz filter is applied to our system to increase response at very low frequencies. Experimental results show that a multiple combined feedback-feedforward control method effectively improves the performance of the flat loudspeaker with extended low-frequency response.

11:40

5aSA9. Ultrasonic transducers with directional converters of vibration of longitudinal-longitudinal type and longitudinal-longitudinal-longitudinal type intended to work in gaseous media. Tadeusz Gudra, Lukasz Palasz, and Krzysztof J. Opielinski (Electronics, Wroclaw Univ. of Technol., Wybrzeze Wyspianskiego 27, Wroclaw 50-370, Poland, tadeusz.gudra@pwr.wroc.pl)

The study presents a realized concept of an ultrasonic transducer intended to work in a gaseous medium, which radiates a focused ultrasonic beam in many directions simultaneously. The use of a longitudinal-longitudinal (L-L) type vibration direction converter makes it possible to radiate ultrasonic energy in a required direction without changing the location of the activation source. The transducer consists of four resonant elements: an ultrasonic sandwich type transducer, vibration amplitude transformer, L-L type converter, and axisymmetrical radiating plates. It is also acceptable to use a longitudinal-longitudinal-longitudinal (L-L-L) type vibration direction converter. The benefit of such a solution is that it is possible to use one ultrasound source for simultaneous activation of several radiating plates. The study presents conductance and susceptance characteristics of a complex resonance system, amplitude frequency characteristics of the level of acoustic pressure and sample characteristics of the directivity of a transducer with two independent radiating plates. The suggested solution creates new possibilities for applications of this type of resonance systems.

Session 5aSCa

Speech Communication: Flow, Structure, and Acoustic Interactions During Voice Production I

Scott L. Thomson, Chair
Mech. Eng., Brigham Young Univ., 435 CTB, Provo, UT 84602

Chair's Introduction—8:55

Invited Papers

9:00

5aSCa1. The role of the thyroarytenoid muscle in the regulation of prephonatory glottal opening. Zhaoyan Zhang and Jun Yin (UCLA School of Med., 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Previous research has demonstrated a positive correlation between the thyroarytenoid muscle activity and the closed quotient, particularly for chest voice production. To understand the physical mechanism behind this correlation, the interaction between the thyroarytenoid and the cricothyroid muscles and the subglottal pressure was investigated using a muscularly controlled vocal fold model. The results showed that activation of the thyroarytenoid muscle stiffened the vocal folds and caused the vocal fold to bulge toward the glottal midline, which makes it possible to maintain intended degree of glottal closure, from fully open to fully closed to tightly compressed, against the varying subglottal pressure. For a constant flow rate, increasing contraction of the thyroarytenoid muscle also led to increased subglottal pressure, thus increasing voice intensity. This function of the thyroarytenoid muscle as a fine regulator of glottal closure is most effective at conditions of low levels of cricothyroid muscle contraction and low-to-middle range of subglottal pressure. At strong cricothyroid muscle contraction or extremely high subglottal pressures, a full-closed prephonatory glottis become impossible. Vibration at this condition is likely to have a small closed quotient and the effects of thyroarytenoid muscle contraction on closed quotient and subglottal pressure are minimal. [Work supported by NIH.]

9:20

5aSCa2. Relationship between divergence angle and skewing of the volumetric flow rate in an excised canine larynx model without a vocal tract. Sid M. Khosla, Liran Oren (Otolaryngology, Univ. of Cincinnati Academic Health Ctr., 231 Albert Sabin Way, 0528, Cincinnati, OH 45208, khoslasm@uc.edu), and Ephraim Gutmark (Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH)

Until now, skewing of the volumetric flow rate (Q) curve was thought to be due to the inertance effects produced by the vocal tract. The goal of these experiments is to measure the volumetric flow rate at the entrance and exit of the glottis during the closing phase in a 1 mm thick coronal section halfway between the anterior commissure and the vocal process. The velocity fields and the intraglottal geometry are measured using modified particle imagining velocimetry (PIV) methodology. In these experiments, five excised canine larynges were used, and in all of them it is shown that the flow rate is greater at the glottal exit than at the glottal entrance when the glottis is divergent and an intraglottal vortex is formed at the superior aspect of the fold. In addition, flow skewing is seen at the glottal exit but not at the glottal entrance. In this talk, we will show that this flow skewing without a vocal tract is due to the entrainment effects of the intraglottal vortex.

9:40

5aSCa3. Determination of the stresses and strain on the superior surface of excised porcine larynges during phonation using digital image correlation. Hani Backshaei, Chan Woo Yang, Amir K. Miri, and Luc Mongeau (Mech. Eng., McGill Univ., 817 Sherbrooke St. West, Montreal, QC H3A 0C3, Canada, luc.mongeau@mcgill.ca)

The stresses and strains acting over the superior surface of excised porcine vocal folds during self-excited, flow-induced vibrations were investigated. Digital image correlation analysis of stereoscopic high speed video data was used. The pre-strain resulting from the initial deformation of the tissue before phonation onset was estimated in four larynges. The kinematics of the vocal folds were also measured with a high temporal resolution using a laser Doppler vibrometer. Chaotic jumps between different vibration modes was observed. Impact stresses were estimated from a model based on Hertzian theory, and isotropic constitutive laws. The results yielded values of compressional stresses near the vocal fold edges that were larger than previously reported data. Comparisons were made with similar data previously obtained for a synthetic silicone replica of the human larynx.

10:00

5aSCa4. Modeling incomplete glottal closure due to a posterior glottal opening and its effects on the dynamics of the vocal folds. Matías Zañartu (Dept. of Electron. Eng., Universidad Técnica Federico Santa María, Av España 1680, Valparaíso, Valparaíso 2390123, Chile, matias.zanartu@usm.cl), Byron D. Erath (Dept. of Mech. & Aeronautical Eng., Clarkson Univ., Potsdam, NY), Sean D. Peterson (Dept. of Mech. and Mechatronics Eng., Univ. of Waterloo, Waterloo, ON, Canada), Robert E. Hillman (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, Boston, MA), and George R. Wodicka (Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN)

Even though incomplete glottal closure is present in normal and pathological voices, it has received little attention in self-sustained models of phonation. The effects of acoustic interaction due to a posterior glottal gap on the tissue dynamics, energy transfer, and glottal aerodynamics were numerically investigated. The domain was prescribed as flow through two separate orifices (posterior gap and

membranous vocal folds) that merge in the supraglottal tract, with the governing flow equations determined from a control volume analysis based on conservation of mass and linear momentum. The equations of motion remained unaffected, although the driving forces were indirectly altered through the acoustic interaction. The method was implemented using the body-cover model, wave-reflection-analog sound propagation, and a boundary-layer asymmetric flow solver. The inclusion of a gap area of 0.03 cm^2 reduced the RMS energy transfer from the fluid to the vocal folds by 20 % and radiated SPL by 5 dB. When compensating for this reduction with an increased subglottal pressure to match the same SPL, a significant increase in MFDR and AC flow was noted, thus mimicking vocal hyperfunction. In addition, larger gap areas yielded less glottal pulse skewing and a glottal airflow proportional to the transglottal pressure drop.

10:20

5aSCa5. Modeling flow through the posterior glottal gap. Ronald Scherer, Brittany Frazer, and Guangnian Zhai (Commun. Sci. and Disord., Bowling Green State Univ., 200 Health Center, Bowling Green, OH 43403, ronalds@bgsu.edu)

The BGSU three mass model of phonation incorporates a posterior glottal gap. The airflow through the gap is governed by a simple hydraulic formula. The current study compares predictions of the gap size to the DC flow of three human subjects over two pitches at normal loudness. This work also explores the importance of the posterior gap by comparing extensive human mean subglottal pressure—mean flow—vocal process adduction data (from one subject) with the behavior of the model. The relation between glottal gap size to the DC flow for the three subjects was exceptionally strong ($R^2 = 0.99$). The relatively linear pressure-flow relationships at specific adduction values for the human data are hypothesized to be mimicked by the model, with a reasonable relationship to the size of the posterior glottal gap. [Work support from NIH.]

10:40

5aSCa6. Acoustic coupling during incomplete glottal closure and its effect on the inverse filtering of oral airflow. Matías Zañartu (Dept. of Electron. Eng., Universidad Técnica Federico Santa María, Av España 1680, Valparaíso, Valparaíso 2390123, Chile, matias.zanartu@usm.cl), Julio C. Ho (Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN), Daryush D. Mehta, Robert E. Hillman (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, Boston, MA), and George R. Wodicka (Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN)

Inverse filtering of oral airflow using closed-phase linear prediction is expected to preserve the effects of source-filter interactions in the glottal airflow pulse. Under incomplete glottal closure, the glottal airflow estimation is more challenging due to a lowered glottal impedance, increased subglottal coupling, and violated all-pole assumption. To account for these effects, a model-based inverse filtering scheme allowing for coupling between glottis and upper and lower airways was developed. Acoustic transmission in the tracts used a frequency-domain transmission line. A linearized, time-varying expression was used for the glottal impedance, along with a dipole representation. Synthetic vowels sounds and actual recordings were used to evaluate the proposed scheme. Subject-specific model parameters were obtained from simultaneous aerodynamic, acoustic, and high-speed videoendoscopic recordings of normal subjects uttering vowels with various degrees of glottal closure. Results illustrated that, even under incomplete glottal closure, the airflow entering the vocal tract preserved source-filter interactions and was comparable to that obtained using closed-phase linear prediction. The scheme also yielded an uncoupled glottal airflow that exhibited a clear pulse de-skewing, making it proportional to the glottal area. Cases with larger glottal gaps exhibited lower mean impedances and less pulse skewing, with airflow estimates proportional to the transglottal pressure drop.

11:00

5aSCa7. Liquid dynamics in a self-oscillating poroelastic model of the vocal fold. Chao Tao (Inst. of Acoust., Nanjing Univ., 22 Hankou Rd., Nanjing, Jiangsu 210093, China, taochao@nju.edu.cn), Jack J. Jiang (Dept. of Surgery, Univ. of Wisconsin Med. School, Madison, WI), and Xiaojun Liu (Inst. of Acoust., Nanjing Univ., Nanjing, China)

The vocal fold tissue is considered as the composite of a porous elastic frame (composed of specialized proteins, carbohydrates, lipids, collagen fibers, and elastin fibers) filled with liquid. The properties of tissue are codetermined by the porous solid, the fluid, and their interaction. The poroelastic description of the vocal fold tissue could improve our knowledge about the mechanical properties and microstructures of these biological tissues. In this study, a self-oscillating poroelastic model is proposed to study the liquid dynamics in the vibrating vocal folds, where the vocal-fold tissue is treated as a transversally isotropic fluid-saturated porous material. Rich liquid dynamics have been found in this model. In the vertical direction, the liquid is transported from the inferior side to the superior side due to the propagation of the mucosal wave. In the longitudinal direction, the liquid is accumulated at the anterior-posterior midpoint. However, the strong collision between two vocal folds forces the accumulated liquid out there in a very short duration. These findings of the liquid dynamics could be helpful for exploring etiology of some laryngeal pathology, optimizing laryngeal disease treatment, understanding hemodynamics in the vocal folds, etc.

11:20

5aSCa8. Nonlinearities in block-type reduced-order vocal fold models with asymmetric tissue properties. Byron D. Erath (Mech. and Aeronautical Eng., Clarkson Univ., 8 Clarkson Ave., Box 5725, Potsdam, NY 13699, berath@clarkson.edu), David E. Sommer (Mech. and Mechatronics Eng., Univ. of Waterloo, Waterloo, ON, Canada), Matías Zañartu (Deparment of Electron. Eng., Universidad Técnica Federico Santa María, Valparaíso, Chile), and Sean D. Peterson (Mech. and Mechatronics Eng., Univ. of Waterloo, Waterloo, ON, Canada)

Modeling the vocal fold structure as a reduced-order system is an attractive approach for exploring the dynamics of both normal and pathological phonation. This approach has been used ubiquitously in scientific speech investigations due to its relatively high order of accuracy and low computational cost. In addition, good agreement can also be found between model and clinical data. In the case of pathological speech complex vocal fold dynamics may exist, exhibiting phenomenon such as bifurcation and chaos. The ability to capture these features in reduced-order vocal fold models is a much celebrated feature. However, the question has arisen whether these nonlinearities arise due to the physics, or if they are merely an artifact of the model and its sensitivity to initial and boundary conditions. We explore the sensitivity of commonly-employed reduced-order vocal fold models to both contact mechanics, and the geometric prescription of the vocal fold model. Nonlinearities arising from asymmetric vocal fold tensioning are investigated. Nonlinearity in the vocal fold dynamics is identified by determining the predictive capability of linear and nonlinear Volterra-Weiner-Korenberg series. Nonlinearities in the vocal fold oscillations are shown to be highly dependent upon model formulation and implementation, as opposed to physical features of speech.

11:40

5aSCa9. High-dimensional and low-dimensional models of tissue and fluid movement in airways: How will they converge? Ingo R. Titze (National Ctr. for Voice and Speech, 136 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@ncvs2.org), Ingo R. Titze (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, Utah), and Ingo R. Titze (National Ctr. for Voice and Speech, Univ. of Utah, Salt Lake City, UT)

Airways surrounded with soft tissues are used by many species to produce sound. The structure of the airway can become unstable and oscillate at constricted regions such as the vocal folds, the ventricular folds, the aryepiglottic folds, the tongue dorsum, the velum, the tongue tip, or the lips. In modeling these fluid-structure interactions, approaches ranging from simple analytical models to low- and high-dimensional computational models have been used. For voice and speech scientists and practitioners who use these models, major issues are (1) what phenomenon do they explain, (2) how valid are model predictions, (3) what level of accuracy is needed to make a prediction, and (4) how user-friendly are the models? Some criteria will be given for making choices on degrees of freedom, tissue and flow characterization, and benchmarking with measurement to converge on utility and complexity of models.

FRIDAY MORNING, 7 JUNE 2013 516, 9:00 A.M. TO 12:00 NOON

Session 5aSCb

Speech Communication: Production and Perception II: The Speech Segment (Poster Session)

Michael Kiefte, Chair

Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada

Contributed Papers

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

5aSCb1. Acoustic and articulatory evidence for the phonological status of liaison consonants. Marie-Josee L'Esperance and Sam Tilsen (Linguistics, Cornell Univ., 522, West Seneca St., Apt A, Ithaca, NY 14850, ml978@cornell.edu)

French liaison consonants (LC) are a special class of word-final segments whose realization depends on a combination of phonological, lexical, and syntactic factors. Most previous analyses viewed LCs as coda consonants realized only before vowel-initial words. Because of their special status as syntactically and lexically conditioned, an interesting question is whether LCs exhibit typical acoustic and articulatory characteristics of word-final consonants. This paper presents the results of an experimental investigation of LCs in adjective+noun pairings in Quebec French, using electromagnetic articulography to collect kinematic data. Compared to typical coda and onset consonants, liaison consonants were found to exhibit smaller magnitude release gestures, and in some cases, LC closure gestures were more similar to those of onsets than codas. Unlike coda consonants, LCs did not induce gestural shortening or laxing (F1 raising) of the preceding vowel. Hence our results indicate that, although LCs have been analyzed as word-final consonants, they exhibit neither typical coda- or onset-like acoustic and articulatory properties. These results are important because they show that syntactically conditioned lexical phonology can result in non-canonical articulatory patterns, and hence speak to the need for models of production to incorporate both lexical representations and syntactic context as factors.

5aSCb2. Comparison of native and non-native consonant articulation with real-time magnetic resonance imaging of the vocal tract. Sam Tilsen (Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, tilsen@cornell.edu), Bo Xu, Pascal Spincemaille (Weill Cornell Medical College, New York, NY), Madhur Srivastava, Peter Doershuck (Cornell Univ., Ithaca, NY), and Yi Wang (Weill Cornell Medical College, New York, NY)

This study examines the effects of vocalic context and stress on consonantal articulation using real-time magnetic resonance imaging (rtMRI) of the vocal tract. A native speaker of English and an L2 English speaker of Mandarin produced eight repetitions of a set of vowel-consonant-vowel sequences in which the target consonants—voiceless stops and nasals—occurred before or after a stressed vowel. Images of the mid-sagittal plane of the vocal tract were acquired with a sampling rate of 8.1 ms and were reconstructed using a variable density golden angle ordered spiral algorithm. Vocal tract variable time-series for each token were extracted from the images by taking the average pixel intensity for each frame in hand-labeled regions of interest. Analyses of variance were conducted on kinematic variables (movement range, velocity, and duration of consonantal closure and release movements) and relative timing of consonantal and vocalic gestural landmarks. The results showed greater effects of stress and vocalic context on articulatory kinematics and timing for the native speaker compared to the non-native speaker. This study demonstrates that rtMRI can be used to assess fine-grained differences in articulation that are likely attributable to language background.

5aSCb3. Effects of phonemic variability and language dominance on Canadian French-English bilinguals' perception of French vowels in various phonological contexts. Allison A. Johnson (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, 2121 Univ. Ave., apt 28, Madison, WI, aajohnson4@wisc.edu) and Franzo Law (Psychology, Univ. of Wisconsin-Madison, Madison, WI)

Phonemic variability can cause productions within a category of one language to be mapped onto multiple categories in a different language [Escudero (2009)]. For example, French productions of /ɛ/ are labeled both as /e/ and /ɛ/ by monolingual English listeners, likely due to spectral variability in production of French /ɛ/ [Strange *et al.* (2009)]. The present study examines how phonemic variability in Canadian French (CF) affects the perceptual tendencies of bilingual Canadian French-English listeners with varying levels of language dominance. Vowel productions by monolingual CF speakers were used in a modified identification task [Law (2011)]. The vowels were word-final in several phonological contexts (preceded by labial, coronal, and back consonants; followed by labial and coronal consonants) in real and nonwords embedded in carrier phrases. A subset of these

5a FRI. AM

vowels /e-ɛ-i-y/ was analyzed for duration, vocalic midpoint, and formant trajectories to examine the relationship between phonemic variation (as a function of phonological environment) and the perception results by bilinguals dominant in either CF or Canadian English (CE). We predict that the performance of CE-dominant listeners will vary in terms of speed and accuracy based on how similar each token is to CE vowel category expectations.

5aSCb4. Listeners' sensitivity to talker differences in voice-onset-time: Phonetic boundaries and internal category structure. Rachel M. Theodore, Emily B. Myers, and Janice Lomibao (Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu)

Recent findings indicate that listeners are sensitive to talker differences in phonetic properties of speech, including voice-onset-time (VOT) in word-initial stop consonants. The current work extends our earlier research by examining the degree to which listeners adjust the initial mapping from acoustic signal to segmental representation on a talker-specific basis. Two groups of listeners are exposed to a talker producing "cane." Word-initial VOTs are manipulated such that one group hears "cane" produced with short VOTs and the other group hears "cane" produced with relatively longer VOTs. Following training, listeners' voicing boundary for a /g/-/k/ continuum is tested. In addition, listeners are tested on phonetic category space by rating members of the continuum for typicality as /k/. If listeners adjust segmental mapping to accommodate talker differences in phonetic properties of speech, then we expect to observe a displacement in the voicing boundaries in line with earlier exposure. Moreover, if this adjustment entails a comprehensive reorganization of phonetic category space, then the /k/ exemplars rated most prototypical will also be displaced for the two listener groups. These data will be discussed in terms of potential constraints on talker-specificity in spoken language processing.

5aSCb5. The Nez Perce vowel system: A phonetic analysis. Katherine Nelson (Dept. of Linguist, MS 23 Rice Univ., P.O. Box 1892, Houston, TX 77251-1892, katienelson@rice.edu)

The Nez Perce language, a highly endangered American Indigenous language, has been of great interest in phonology over the years due to its unusual vowel system and vowel harmony process. Nez Perce has five monophthongs and seven diphthongs, all with phonemic length. This system is unusual because rather than /i, e, a, o, u/ as is common, the Nez Perce inventory is /i, æ, a, o, u/. This uncommon inventory leads to two seemingly unrelated dominant, /i, a, o/, and recessive, /i, æ, u/, vowel harmony groups. To date there has been no phonetic analysis of the vowel system. This paper provides an acoustic analysis of the vowels as well as the vowel harmony system. Five native speakers (two males and three females) were used to analyze the vowels and the three female native speakers were used for the vowel harmony study. Results support the current vowel system analysis for Nez Perce. The vowel harmony data lend support to the current advanced tongue root analysis; however, it also poses questions for future research.

5aSCb6. Perception of Canadian French rhotic vowels. Jeffrey Lamontagne (Linguistics, Univ. of Ottawa, Ottawa, ON, Canada) and Jeff Mielke (English, North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-8105, jimielke@ncsu.edu)

Some speakers of Canadian French produce words such as *pneu*, *un*, and *coeur* with rhotic-sounding vowels similar to English /ɹ/ (Dumas 1972). Articulatory imaging [Mielke (2011)] shows that they are produced with bunched and retroflex tongue postures and low F3, much like English /ɹ/. Nevertheless, native speakers typically are completely unaware of the difference, even when it is pointed out to them. We report a preliminary perception study of rhotic vowels. 7735 words with mid front round vowels were coded as "rhotic," "non-rhotic," or "ambiguous" by two listeners: a French-English bilingual from eastern Ontario and an American English speaker. The bilingual coded 0.3% as rhotic (vs. 10.0% for the anglophone) and 7.3% as ambiguous (vs. 8.9%). Logistic regressions show that the anglophone relied on F3 to distinguish rhotic+ambiguous tokens from non-rhotic tokens, while the bilingual weighted several cues about equally, including F1 cues to diphthongization, which can co-occur with rhoticity. Results will be presented from an ongoing AX discrimination task experiment involving rhotic, non-rhotic, and ambiguous vowel tokens, with francophone, bilingual, and anglophone listeners from the Ottawa-Gatineau region, Paris, France, and Raleigh, North Carolina.

5aSCb7. Describing alternative articulations of the Spanish trill /r/ by ultrasound technology. Ahmed Rivera-Campos and Suzanne E. Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, 3407 Clifton Ave., Apt 27, Cincinnati, OH 45220, riveraam@mail.uc.edu)

The Spanish trill /r/ is typically described as having a single realization—that is, as the result of aerodynamic forces operating on a tongue tip/blade constriction placed midsagittally in the dental, alveolar, or postalveolar place of articulation, in such a way that air channels along the sides of the tongue open and close for multiple cycles of vibration. As with American English /ɹ/, this sound is acquired late by typically developing children and is frequently an element in articulatory disorders. As with American English /ɹ/, perceptually equivalent "correct" trill /r/'s may be realized differently by different speakers. Knowledge of these alternate "correct" realizations would clearly be helpful to clinicians and learners of Spanish. In this preliminary study, we report data from ultrasound images of individuals who speak different dialects of Spanish. Preliminary data suggests there are at least two different articulatory postures used when producing the Spanish trill /r/, one of which involves lateralization. These articulatory differences do not affect what native listeners categorize as perceptually correct Spanish trills.

5aSCb8. Producing whole speech events: Anticipatory lip compression in bilabial stops. Chenhao Chiu and Bryan Gick (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, chenhao@alumni.ubc.ca)

Bilabial stops /b/, /p/, and /m/ ostensibly share a common lip constriction. Recent evidence shows that different bilabial stops involve distinct facial muscle activations, suggesting that oral speech movements anticipate aerodynamic conditions [Gick *et al*. Proc. Acoust. (2012) 2pSC1]. The present study investigates how the lips themselves behave in whole speech events. Existing models of speech production governing only articulatory motions predict that lip compression would respond to changes in aerodynamic conditions rather than anticipating such changes; a model that includes whole events predicts anticipatory activation of lip muscles with concomitant kinematic lip compression, but only in cases where a real increase in air pressure is expected. Lip kinematics were recorded using OptoTrak to trace lip movements of bilabial stops in response to imperative acoustic stimuli. Results show consistent anticipatory lip compression in spoken /b/ and /p/, but not in non-speech jaw opening movements and only sporadic compression in mouthed /b/ and /p/, where air pressure is not expected to increase. Biomechanical simulation using an orofacial model developed within the Artisynth simulation toolkit (www.artisynth.org) confirms anticipatory muscle activations. These findings support a model of speech tasks wherein coordinated body-level muscular systems govern whole speech events.

5aSCb9. Intrinsic variations in jaw deviations in English vowels. Caroline Menezes (Health and Human Sci., Univ. of Toledo, 2801 W. Bancroft St., Toledo, OH 43606, caroline.menezes@utoledo.edu) and Donna Erickson (Showa Music Univ., Kawasaki, Japan)

Research shows that jaw deviations follow lexical and phrasal stress patterns reflecting the rhythmic structure of English [Erickson (2002), (2010), Erickson *et al*. (in press), Menezes (2003), Menezes *et al*. (2003)]. Syllable strength thus can be determined by the amount of jaw displacement regardless of the target vowel. This study systematically analyzes jaw displacement based on the intrinsic variations in vowel height. EMA recordings were analyzed of one speaker producing a short phrase wherein 11 English vowels were spoken in a controlled phonetic environment. The phrase used was "Type X first" where, X was a closed monosyllabic word containing the target vowel surrounded by stop consonants. Stop consonants allow the jaw to start from the bite plane and return to the bite plane therefore all deviations of the jaw are attributed to the articulation of the vowel. The target word was produced in phrase initial, middle and final positions. Comparisons were made across vowel height, tongue root advancement and phrase position. Preliminary findings reveal that jaw displacement was significantly

different at the level of $p = 0.05$ for vowels based on all three parameters: Vowel height, vowel tense/lax, and phrasal positioning.

5aSCb10. Multidimensional scaling of English fricatives using the acoustic change complex of electroencephalogram recordings. Paul Iverson, Marta Mulyak, and Anita Wagner (Univ. College London, 2 Wakefield St, London WC1N 1PF, United Kingdom, p.iverson@ucl.ac.uk)

In electroencephalogram (EEG) recordings, there is a characteristic P1-N1-P2 complex after the onset of a sound, and a related complex, called the Acoustic Change Complex (ACC), when there is a change within a sound (e.g., a formant transition between two vowels). In the present study, the ACC was measured for all possible pairs of eight sustained voiced and voiceless English fricatives, in EEG recordings from native speakers of British English. The magnitude of the ACC was used as a similarity measure for multidimensional scaling (MDS), producing a two-dimensional perceptual space that related to both voicing and place of articulation. The results thus demonstrate that this combination of ACC and MDS can be effective for mapping multidimensional phonetic spaces at relatively early levels of auditory processing, which may be useful for evaluating the effects of language experience in adults and infants.

5aSCb11. An acoustic study of the Mary-merry-marry vowels in the Mid-Atlantic United States. Carina Bauman (Linguistics, New York Univ., 10 Washington Place, New York, NY 10003, cb1864@nyu.edu)

This paper examines the acoustic properties of a subset of American English vowels before /r/–specifically, the mid and low front vowels in the MARY, MERRY, and MARRY classes. While the dialectal variation associated with the MARY, MERRY, and MARRY classes is well known in the sociolinguistics literature, the precise phonetic nature of these vowels has not been well studied. Many scholars [e.g., Labov *et al*. (2006)] transcribe the vowels as /eɪ/, /ɛ/, and /æ/, respectively, but these labels are largely impressionistic. The present study samples five speakers from the Mid-Atlantic region of the United States who maintain a three-way distinction between MARY, MERRY, and MARRY. The speakers' productions of MARY, MERRY, and MARRY vowels in both sentence and word list tasks were analyzed in Praat, and the resultant formant values were submitted to one-way ANOVAs, followed by pairwise comparisons (Tukey's HSD). The primary finding of this study is that the MARY vowel is acoustically closest to a tense (raised and fronted) variant of /æ/, similar to that which appears before nasals in many American English dialects. Contrary to previous descriptions, the MARY vowel shows little overlap with /eɪ/, suggesting that the conventional transcription should be revised.

5aSCb12. Effects of variation on processing of word-medial consonants. Benjamin V. Tucker (Dept. of Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca), Kaitlin Mackie (Dept. of Speech Pathol. and Audiol., Univ. of AB, Edmonton, AB, Canada), and Tatiana Kryuchkova (Dept. of Linguist, Univ. of AB, Edmonton, AB, Canada)

The present study investigates processing of variation in word-medial stops and the role of this variation in phoneme recognition. Variation in word-medial stops has been shown to influence lexical access, but it is unclear whether this variation also affects recognition at the phoneme level [Tucker, J. Phonet., **39**, (2011), 312–318]. A phoneme monitoring task is used to investigate the role of production variation in the identification of word-medial stop consonants. Following Warner and Tucker [J. Acoust. Soc. Am. **130**, (2012), 1606–1617], the stimuli comprise English nonwords with target consonants [b], [p], [d], [t], [g], [k] in an intervocalic post-stress position produced with a range of production variation. This variation was grouped into three sets including groups at each extremes and a group in the middle of the range. The acoustic characteristics of the stops, which reflect this variation (e.g., consonant duration and intensity difference) were extracted and used to model the reaction times collected during the phoneme monitoring experiment. The reaction times were statistically modeled using a linear mixed-effects regression. Models of spoken word recognition and models of lexical storage are used to interpret the results and the contribution of the results to our understanding of these models is discussed.

5aSCb13. Assimilation of word-final nasals to following word-initial place of articulation in United Kingdom English. Margaret E. Renwick, Ladan Baghai-Ravary, Rosalind Temple, and John S. Coleman (Phonet. Lab., Univ. of Oxford, 41 Wellington Square, Oxford, United Kingdom, margaret.renwick@phon.ox.ac.uk)

Using very large speech corpora, we can study rare but systematic pronunciation patterns in spontaneous speech. Previous studies have established that word-final alveolar consonants in English (/t/, /d/, /n/, /s/ and /z/) vary their place of articulation to match a following word-initial consonant, e.g., "ran quickly" → "ra[ŋ] quickly." Assimilation of bilabial or velar nasals, e.g., "alar[ŋ] clock" for "alarm clock," is unexpected according to linguistic frameworks such as underspecification theory. The existence of systematic counterexamples would challenge that theory, but these might have been previously overlooked because they are infrequent. From the c. 8-million word Audio BNC (http://www.phon.ox.ac.uk/AudioBNC) we extracted more than 4,000 tokens of relevant word pairs, to determine whether non-alveolar assimilations occur and with what distribution. Word and segment boundaries were obtained by forced alignment, and F1–F3 formant frequencies were estimated using Praat. Formant frequencies in assimilation environments were compared to non-assimilating controls (e.g., the**m d**own vs. the**m b**ack/the**n d**own). We also examined patterns of variability in different contexts. We will present evidence that velar and bilabial nasals sometimes *do* assimilate, though less frequently than alveolars.

5aSCb14. Nasality effects in word-final nasal clusters. Emily Nguyen (Linguistics, New York Univ., 10 Washington Place, New York, NY 10003, emily.nguyen@nyu.edu)

This study presents an acoustic analysis of the proposed phonological process of nasal deletion in English [Cohn (1993), Bybee (2001)] in which vowel-nasal consonant-oral consonant (VNC) sequences differ in the realization of the nucleus based on the voicing feature of the final oral consonant. Previous work has proposed that the nasal consonant in VNC[-voice] sequences, e.g., tent, is deleted resulting in the realization of a fully nasalized vowel followed by a voiceless oral consonant, ṼC[-voice]. This phonological process of nasal deletion is said not to take place in VNC[+voice] sequences, e.g., tend, resulting in the full realization of all segments. Data from 15 speakers of American English were analyzed, and nasal deletion rates for VNC[-voice] sequences were low ($<12\%$). A further complication for this proposed phonological process is that vowel nasality differences between VNC[-voice] and VNC[+voice] sequences are small. A measure of vowel nasality, A1-P0, shows that while nasality is predicted to be constant from the onset of vowel production in VNC[-voice] sequences, statistical results show that vowels in both VNC[-voice] and VNC[+voice] sequences are progressively nasalized throughout production. Taken together, these findings suggest that nasal deletion is occasional and occurs due to phonetic implementation rather than a phonological process.

5aSCb15. Articulatory overlap in English syllables with postvocalic /ɹ/. Rachel Walker (Linguistics, Univ. of Southern California, GFS 301, 3601 Watt Way, USC, Los Angeles, CA 90089-1693, rwalker@usc.edu) and Michael Proctor (Linguist, Univ. of Western Sydney, Sydney, NSW, Australia)

In General American English (GAE), only two full vowels [ɑ̃, ɔ] occur in syllables ending in [ɹ] plus a non-coronal consonant, e.g., <harp>, <pork>. An articulatory study of rhotic production by three speakers of GAE was conducted using real-time structural magnetic resonance imaging (rtMRI) [Narayanan *et al*. (2004)]. Subjects produced /ɹ/ in simple and complex syllable codas in a range of vocalic environments. Results show that the tongue dorsum shows the least movement in [-ɑ̃ɹ-] and [-ɔɹ-] sequences. This dorsal stability sheds light on why [ɑ̃] and [ɔ] are the only full vowels occurring before codas with /ɹ/ and a non-coronal consonant. English syllable rimes have been analyzed as maximally three timing units in length [Hammond (1999)]. Long vowels occupy two units, and most coda consonants occupy one unit, rendering rimes with long [ɑ̃]/[ɔ] followed by [ɹp] or [ɹk] problematic. We hypothesize that the high degree of overlap in the dorsal posture in [ɑ̃ɹ] and [ɔɹ] sequences allows their gestures to be partially blended and function like a diphthong that occupies two units in the rime. This study supports a view of maximal constituency in rimes with [ɹ] that takes articulatory overlap into account.

5aSCb16. Acoustic correlates of flaps in North American English. Donald Derrick (NZILBB, NZILBB, U Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, donald.derrick@gmail.com) and Ben Schultz (MARCS Inst., Univ. of Western Sydney, Penrith, NSW, Australia)

Using Brightness and Motion mode ultrasound, Derrick and Gick (2011, CJL) identified four categorical variations of flap (rapid "d"-like) tongue movements produced in North American English (that is, up-flaps, down-flaps, alveolar taps, and postalveolar taps). These variants can be used to test hypotheses about constraints on speech articulation, such as local context, gravity and elasticity, speech rate, and longer distance anticipatory coarticulation. The present study examines acoustic correlates of flap and rhotic ("r"-like) vowel variations in order to facilitate the understanding of articulatory mechanisms that underlie acoustic outputs. Understanding the relationship between articulatory mechanisms and acoustic outputs may allow us to draw inferences about articulatory mechanisms from pre-existing and future acoustic databases. Preliminary results identify significant differences in f0, F1, F2, F3, F4, and F5 values between these flap variants. In addition, we used supervised hierarchical clustering to aid in identification of flap variants based on both vocalic context and acoustic parameters. Unsupervised hierarchical clusters will also be used to identify whether the four flap variants that were previously identified in our articulatory studies are enough to capture the actual categorical variation that occurs and, if not, to identify currently unknown categories of flap variation in North American English.

5aSCb17. Cortical hemodynamic response patterns to normal and whispered speech. Gerard B. Remijn (Int. Education Ctr., Kyushu Univ., Shiobaru 4-9-1, Minami-ku, Fukuoka 815-8545, Japan, remijn@design.kyushu-u.ac.jp), Mitsuru Kikuchi, Yuko Yoshimura (Res. Ctr. for Child Mental Development, Kanazawa Univ., Kanazawa, Japan), Sanae Ueno, Kiyomi Shitamichi, and Yoshio Minabe (Dept. of Psychiatry and Neurobiology, Kanazawa Univ., Kanazawa, Japan)

Whispered speech is often used in direct person-to-person communication as a means to confidentiality. Compared with normally vocalized speech, whispered speech is predominantly unvoiced, i.e., produced without vocal fold vibration, and has no clear fundamental frequency. By using near-infrared spectroscopy (NIRS), we assessed cortical hemodynamic response patterns to normally vocalized and whispered speech in adult listeners ($n = 13$). Stimuli consisted of 20-s strings of Japanese word associations spoken by a female voice. Average oxygenated hemoglobin values (oxy-Hb) were obtained over two regions of interest (ROIs). Results showed that oxy-Hb values during the perception of normally vocalized speech were highest over the left temporal ROI, but not significantly different from values measured over other ROIs. Oxy-Hb values during whispered speech were highest over the right temporal ROI and significantly higher ($p < 0.05$) than those obtained over the left temporal ROI. No significant differences, however, were found in oxy-Hb comparisons between normally vocalized and whispered speech, although the right temporal ROI comparison bordered on significance, with whisper inducing the higher value. Together, the results seem to suggest that whispered speech is a potent catalyst of cortical hemodynamic activity, especially over the right temporal cortex, in spite of its relatively modest sound level as compared to normal speech.

5aSCb18. Spectral discrimination of consonants for English and Chinese Listeners. Tatum M. Fritz and Chang Liu (Commun. Sci. and Disord., Univ. of Texas at Austin, 1 Univ. Station A1100, Austin, TX 78712, tatumfritz@utexas.edu)

Previous work in our laboratory showed that Chinese listeners had significantly higher thresholds for vowel formant discrimination than American English listeners, possibly due to a sparse vowel system in Chinese but a crowded vowel space in American English. Considering that the two languages have similar consonant densities (i.e., similar numbers of consonants), we hypothesized that English and Chinese listeners might have similar thresholds of spectral discrimination of consonant stimuli. Thresholds of spectral shift in the English consonant, /s/, were measured in an isolated form and in a VCV context for both English and Chinese listeners. Preliminary results showed that there was no significant difference in consonant discrimination thresholds between the two groups of listeners for either isolated or VCV consonants. These findings support that phonemic density may play an important role in spectral discrimination of speech sounds.

5aSCb19. Three-dimensional vocal tract modeling of fricatives /s/ and /sh/ for post-glossectomy speakers. Xinhui Zhou (Elec. Eng., Univ. of Maryland, College Park, 4325 Rowalt Dr., apt 101, College Park, MD 20740, zxinhui2001@gmail.com), Jonghye Woo (Dept. of Neural and Pain Sci. and Orthodontics, Univ. of Maryland Dental School, Baltimore, MD), Maureen Stone (Dept. of Neural and Pain Sci. and Orthodontics, Univ. of Maryland Dental School, College Park, MD), and Carol Espy-Wilson (Elec. Eng., Univ. of Maryland, College Park, College Park, MD)

Production of fricatives involves a narrow supraglottal constriction along the vocal tract. Air flows through the constriction, and generates turbulent noise source(s) by impinging on some obstacles downstream. In post-glossectomy speakers, the production of /s/ and /sh/ is often problematic. It is mainly caused by the tongue surgery, which changes tongue properties such as volume, motility, and symmetry, preventing the tongue from creating proper constrictions. The purpose of this study was to gain some insights on how the vocal tracts of abnormal /s/ and /sh/ are shaped and what are their corresponding acoustic consequences. Based on cine magnetic resonance images, we built 3-D vocal tract models for /s/ and /sh/ from two post-glossectomy speakers (one with abnormal /s/ and the other with abnormal /sh/). Due to the missing part of the tongue, the reconstructed vocal tracts are asymmetric with either an air-flow bypass or a side branch formed near the constrictions. Two coupled physics submodels are included in the 3-D FEM acoustic simulation: incompressible potential flow for the mean air flow and aeroacoustics for the distributed noise sources. The resulting acoustic spectra and acoustic roles of air flow bypass or side branch will be discussed. [This study was supported by NIH R01CA133015.]

5aSCb20. A study of data normalization measured by an electro-magnetic articulograph. Seiya Funatsu (Sci. Information Ctr., Prefectural Univ. of Hiroshima, 1-1-71 Ujinahigashi Minami-ku, Hiroshima 734-8558, Japan, funatsu@pu-hiroshima.ac.jp) and Masako Fujimoto (Ctr. for Corpus Development, National Inst. for Japanese Lang. and Linguist., Tachikawa, Japan)

We investigated the normalization of the data measured using an electro-magnetic articulograph (EMA). The data normalization was needed because the size of the articulator of each subject is different from the sex and/or body size. Moreover, tongue movement range depends upon the speaking styles, clear or unclear. Our experiments were as follows: tongue tip movements during articulation of non-native consonant clusters were measured. Speakers were 2 Japanese and 2 Germans. Speech samples were four nonsense words, bnaht, pnaht, gnaht, knaht. Tongue tip displacement D (mm) and moving time T (ms) between first and second consonant in consonant clusters were measured. D_x (X component of D) was normalized by the difference between maximum value of X (X_{max}) and minimum value of X (X_{min}) in each utterance, i.e., $D_{nx} = D_x/(X_{max} - X_{min})$. Also, D_y (Y component of D) was normalized, i.e., $D_{ny} = D_y/(Y_{max} - Y_{min})$. Hence, D_n (normalized D) was $D_n = (D_{nx}^2 + D_{ny}^2)^{1/2}$. T was normalized by word length L (ms), i.e., $T_n = T/L$. Before the normalization, the measured data was localized by each speaker on the T-D plane, while the normalized data were not localized on the $T_n - D_n$ plane. Accordingly, it was suggested that this simple normalization method would be effective in this experiment.

5aSCb21. The fluctuating-masker benefit for normal-hearing and hearing-impaired listeners with equal audibility at a fixed signal-to-noise ratio. Kenneth K. Jensen and Joshua G. Bernstein (Audiol. and Speech Ctr., Walter Reed National Military Medical Ctr., 8901 Rockville Pike, Bethesda, MD 20889, kkj@jensenkk.net)

While normal-hearing (NH) listeners demonstrate better speech intelligibility for fluctuating-masker than for stationary-noise conditions, hearing-impaired (HI) listeners generally show little or no fluctuating-masker benefit (FMB). This result has been interpreted in terms of suprathreshold deficits (e.g., reduced spectral or temporal resolution or distorted stream-segregation cues) that limit "dip-listening." However, reduced FMB for HI listeners might instead be attributable to audibility limitations or to differences between the signal-to-noise ratios (SNRs) at which NH and HI listeners are tested. This study examined this issue by equalizing stationary-noise performance to allow measurements at a common SNR, equalizing audibility, and presenting identical signals to pairs of NH and HI listeners. Audibility was equalized using linear gain, low-pass filtering (4 kHz) and intensity filtering to remove speech-signal elements below the HI audiometric

threshold. Nonsense-syllable identification performance in stationary noise was equalized by adjusting the response set size. Stationary-noise trials (adapting set size) were interleaved with fluctuating-masker trials (adapting SNR), ensuring stable stationary-noise performance throughout the test. Fluctuating maskers included low- and high-rate modulated noise, speech-modulated noise, and an interfering-talker condition. Results were assessed to determine whether and under which conditions the HI listeners demonstrated reduced FMB not attributable to SNR or audibility effects.

5aSCb22. Phonetic properties of [v] in Russian, Serbian, and Greek. Christina Bjorndahl (Linguistics, Cornell Univ., 407 Lake St., B24, Ithaca, NY 14850, cjm295@cornell.edu)

This study examines the phonetic properties of the segment [v] in Greek, Serbian, and Russian. [v] patterns phonologically like an obstruent in Greek, but like a sonorant in Serbian; in Russian, it patterns with both obstruents and sonorants. We test the hypothesis that cross-linguistic differences in the phonological status of [v] correlate with phonetic differences. We report on spectral and durational measures of [v] in four environments: word-initial stressed, word-initial unstressed, word-medial stressed, and word-medial unstressed. Our results show an association between phonological patterning and phonetic realization. Greek tokens of [v] are produced with significantly more high-frequency spectral energy than those in Serbian, suggesting a relation between phonological status and phonetic realization in these two languages. Tokens of Russian [v] exhibit the same relationship to tokens of Serbian [v] in word-initial stressed position; elsewhere, they are produced with relatively little high-frequency spectral energy. Furthermore, the effects of word position and syllable stress are found to be additive in Russian. These results are important because they support the notion that there exist interactions between the phonological status of a segment and its phonetic realization.

5aSCb23. Phonological structure, non-native phoneme discrimination, working memory, and word learning. Noah H. Silbert, Benjamin K. Smith, and Scott R. Jackson (Ctr. for Adv. Study of Lang., Univ. of Maryland, 7005 52nd Ave., College Park, MD 20742, nsilbert@umd.edu)

It is well known that perception of non-native speech sounds is influenced by exposure and the mapping between non-native and native phonological categories. However, very little is known about the relationships between phonological structure, individual differences in non-native phoneme discrimination ability, and non-native word learning. These relationships are important in the design of tests for personnel selection for second language training. Two experiments were conducted to probe the generality of phoneme discrimination ability and the role of phonological structure and discrimination ability in word learning. In one experiment, 169 participants discriminated non-native contrasts from nine languages—three voicing/laryngeal contrasts, three place contrasts, and three tone/intonation contrasts. Confirmatory factor analysis model comparisons show that correlations between discrimination accuracies across contrasts are driven by low-level phonological structure (featural and segmental/super-segmental properties). In a second experiment, phonological working memory and voicing, place, and tone discrimination were measured for 167 participants and used to predict learning of pairs of non-native words differing in voicing, place, and tone. Consistent with the results from the first experiment, discrimination ability predicts accuracy in word learning above and beyond the ability of phonological working memory and according to feature-specific differences.

5aSCb24. Neutralizing differences in jaw displacement for English vowels. J. C. Williams (Independent, Kamuela, HI), Donna Erickson (Showa Univ. of Music, 1-11-1 Kamiasao, Asao-ku, Kawasaki 215-8558, Japan, EricksonDonna2000@gmail.com), Yusuke Ozaki (Univ. of Tokyo, Tokyo, Japan), Atsuo Suemitsu (JAIST, Ishikawa prefecture, Japan), Nobuaki Minematsu (Univ. of Tokyo, Tokyo, Japan), and Osamu Fujimura (The Ohio State Univ., Columbus, OH)

Maximum jaw displacement in the syllable varies primarily by vowel quality, syllable position in the phrase, lexical and phrasal stress, prosodic conditions, and the syllable consonantal periphery. EMA recordings were made of CVC syllables in 3-word phrases uttered by an American English speaker, where three target CVC words occurred in phrase initial, middle, and final positions, in order to ascertain the effect of vowel quality and syllable phrase position on jaw displacement, independent of other factors. Eleven English vowels, omitting diphthongs, formed the syllable nuclei, voiceless obstruents /p, t, k/ formed the syllable periphery, and the intonation pattern was kept constant for each phrase. Jaw displacement was measured by coil placement at the midline of the base of the lower incisors. The maximum vertical mandibular displacement on the vertical axis (z-axis for 3D EMA) was measured for each target CVC word. For each of the 11 vowels, an algorithm was developed to neutralize differences in the contribution of the mandibular vertical excursion in each of the three phrasal positions, i.e., 33 neutralization measures. These results indicate that this method may neutralize the mandibular contribution to differences in phonological vowel quality and phrasal position.

5aSCb25. Effect of syllable onset, coda, and nucleus on degree of skin stretching over the mandible. Ian L. Wilson (CLR Phonet. Lab., Univ. of Aizu, Tsuruga, Ikki-machi, Aizuwakamatsu, Fukushima 965-8580, Japan, wilson@u-aizu.ac.jp) and Donna Erickson (Showa Music Univ., Kawasaki, Japan)

Movements of the mandible have been shown to correlate with English speech rhythm, and significant differences have been found between native speakers' mandible movements and those of second-language speakers. A simple, inexpensive method of inferring movements of the mandible is to use video tracking of a chin marker during speech. However, since the skin is free to stretch over the mandible, inferences using the chin marker may not always be accurate. This study examines the degree of skin stretching during onset stop consonant, coda stop consonant, and vowel in CVC syllables spoken as the middle word in a 3-word utterance. We made electromagnetic articulometer (EMA) recordings of two North American English speakers (1 male, 1 female). Measurements were made from coils placed on the lower incisor (LI) and on the skin of the mental protruberance (chin). Preliminary results show that both speakers have significant differences during the syllable nucleus between the LI and chin coils due to onset consonant, but not coda consonant. These results need to be taken into account as we continue to develop a method for video recording jaw displacement patterns in running speech.

5aSCb26. Anatomical considerations on the extrinsic tongue muscles for articulatory modeling. Kiyoshi Honda (School of Comput. Sci. and Technol., Tianjin Univ., 92, Weijin, Nankai, Tianjin 300072, China, khonda@sannet.ne.jp), Emi Z. Murano (Dept. of Otolaryngol., Johns Hopkins Univ., Baltimore, MD), Sayoko Takano (Dept. of Physiol., Univ. of Arizona, Tucson, AZ), Shinobu Masaki (ATR-Promotion, Inc., Kyoto, Japan), and Jianwu Dang (School of Comput. Sci. and Technol., Tianjin Univ., Tianjin, China)

Physiological articulatory models have evolved from simpler forms to complex ones, while recent models preserve traits of oversimplification and anatomical unreality. This work combines MRI observations at ATR-BAIC and Johns Hopkins University to point to the issue for advancing extrinsic tongue muscle modeling. The genioglossus, previously thought to arise from the genial tubercle of the mandible, has direct fiber attachments on the short tendon of the tongue. The posterior genioglossus, being regarded as a functional division of the muscle, can be given anatomical definition to the horizontal bundle arising from the inferior aspect of the short tendon. The styloglossus has been modeled as linear strings traveling "free in air" before inserting into the tongue, but the extralingual part is actually restrained by the surrounding soft tissues to lack mobility. The intralingual styloglossus forms anterior and posterior slings in the tongue tissue, possibly with the distal fibers of the hyoglossus. Combined styloglossus and hyoglossus shortening via the slings may be a factor shaping the tongue into various forms. [Work supported by WQ20111200010, 2013CB329301, and NIDCD K99/R00-DC009279.]

5aSCb27. Pharyngeal constriction in English diphthong production. Fang-Ying Hsieh, Louis Goldstein, Dani Byrd (Linguistics, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, fangyinh@usc.edu), and Shrikanth Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

This study tests the hypothesis that the acoustic difference between [a] in English diphthongs (e.g., [a] in "pie'd") and its corresponding monophthong (e.g., [a] in "pod") results from the same pharyngeal gesture being truncated by the following palatal glide in the diphthongal environment. Production data were collected with real-time MRI and have been analyzed using the direct image analysis (DIA) technique, which infers tissue

movement by tracking pixel intensity change over time in regions of interest. Preliminary results show that (1) DIA is capable of capturing the timing and magnitude of the pharyngeal constriction gesture that produces [a], and (2) the proposed hypothesis is supported; the formation time of the pharyngeal constriction in diphthongs is generally shorter than that in like monophthongs. Further, the pharyngeal component of the diphthong is shortened in phrase-medial as opposed to phrase-final position or when followed by a voiceless as opposed to a voiced coda consonant, and the duration of this interval strongly correlates with the resulting constriction degree as predicted by the truncation analysis. [Work supported by NIH.]

5aSCb28. Does articulatory setting provide some mechanical advantage for speech motor action? Vikram Ramanarayanan, Adam Lammert (Elec. Eng., Univ. of Southern California, 3740 McClintock Ave., EEB421, Los Angeles, CA 90089-2564, vramanar@usc.edu), Louis Goldstein (Linguistics, Univ. of Southern California, Los Angeles, CA), and Shrikanth Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Articulatory setting postures adopted during speech production are examined with the goal of determining whether setting postures are more mechanically advantageous than rest positions in facilitating motion of vocal tract articulators toward task goals. Articulatory simulations using the Task Dynamics Application (TADA) suggest that setting postures afford large changes with respect to speech tasks for relatively small changes in low-level speech articulators, thus affording greater mechanical advantage as compared to absolute rest postures. This study investigates this hypothesis using real-time Magnetic Resonance Imaging (rtMRI) data of read and spontaneous speech elicited from five healthy speakers of American English. Frames corresponding to inter-speech pauses, speech-ready intervals, and absolute rest intervals were identified and image features were automatically extracted to quantify the vocal tract postures in terms of both task-level constriction variables and articulatory variables. Locally Weighted Regression is then used to estimate the ratio of task velocities to articulator velocities (i.e., the lever or speed ratio) at postures corresponding to the different intervals of interest. Results show substantially higher speed ratios at inter-speech and ready postures as compared to absolute rest postures. [Work supported by NIH.]

5aSCb29. An electropalatography study of nasal-trill/lateral sequences in Spanish. Laura Colantoni (Spanish and Portuguese, Univ. of Toronto, Toronto, ON, Canada) and Alexei Kochetov (Dept. of Linguist, Univ. of Toronto, 100 St. George St., Rm. 4076, Toronto, ON M5S 3G3, Canada, al.kochetov@utoronto.ca)

Trills and laterals require relatively precise articulatory and aerodynamic settings that are at least partly incompatible with setting necessary to produce nasal stops. Historically, this incompatibility has often been resolved through assimilation, deletion, or epenthesis in within-word [n+r] and [n+l] clusters (e.g., Romance [nr] > [rr] or [ndr]). It is expected that similar, yet gradient effects would be observed in across-word or hetero-morphemic sequences of nasals and liquids. This study examines the production of Spanish nasal-liquid sequences using electropalatography (EPG). Linguopalatal contact data were collected from nine native speakers of Spanish (representing three dialects) producing various utterances with nasals before /r/ and /l/ (as well as before /n/). The analysis of C1 and C2 using standard EPG indices of constriction location and degree showed that nasals had a more retracted and partly deocclusivized constriction before /r/, and a lowered tongue dorsum before both /r/ and /l/. These differences, clearly reflecting anticipatory coarticulatory effects, were similar across speakers and the three dialects, which differ in the default realization of the final nasal (alveolar or velar). The findings provide evidence for articulatory incompatibility as a source of historical development of combinations of nasals and liquids.

5aSCb30. Perception of non-native consonant length in naïve English listeners. Vincent Porretta and Benjamin V. Tucker (Univ. of Alberta, 2-40 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, porretta@ualberta.ca)

English speakers are sensitive to phonetic length [Pickett and Decker (1960)] but do not maintain a phonemic length contrast [Hayes (2002)]. This study examines English speakers' ability to discriminate and identify intervocalic consonant length in Finnish non-words. The consonants were manipulated for length and half of the participants were given brief written instruction regarding the Finnish length contrast. In an AX discrimination task, participants responded to increasing contrast ratio gradiently; however, the instructed group performed significantly better than the uninstructed group. Proficiency in any second language aids contrast detection in those receiving no instruction. In a forced-choice identification task, participants showed no evidence of boundary effects; however, the instructed group performed significantly more like native Finnish controls. Again, second language proficiency aids in consonant length detection. The present results indicate that information about and attention to a novel contrast, along with second language experience, aid processing in novice listeners. Given that learners can eventually maintain native-like contrasts, these factors may be influential in the initial formation of L2 phonological representations and support phonetic level of processing [Werker and Tees (1984)], intermediate to non-linguistic acoustic processing and phonemic processing, at which acoustic duration becomes a phonetically relevant cue.

5aSCb31. A following sibilant increases the ambiguity of a sibilant continuum. David Fleischer (Linguistics, McGill Univ., 1085 Ave. Dr. Penfield, Montreal, QC, Canada, david.fleischer@mail.mcgill.ca), Meghan Clayards (School of Commun. Sci. and Disord. & Linguist, McGill Univ., Montreal, QC, Canada), and Michael Wagner (Linguistics, McGill Univ., Montreal, QC, Canada)

We examined the effect of three following contexts: /s/, /ʃ/, and vowel, on the categorization of a /s/-/ʃ/ continuum. Unlike previous findings of a shift in category boundary due to context [Mann and Repp (1980)], we found that in the context of a following sibilant, listeners found the target sibilant to be more ambiguous (shallower categorization slopes and responses closer to chance) than when followed by a vowel ($p < 0.001$). There was also a tendency for the /ʃ/ context (which affects pronunciation) to create more ambiguity than the /s/ context (which does not) ($p = 0.057$). In a second experiment, half the participants heard the following context as part of the same syntactic phrase as the target (e.g., "Whenever they fra? Shelly gets upset") and half heard it as part of a different phrase (e.g., "Whenever they fra? Shelly, John gets upset"). Pronunciation usually is more affected when target and context are in the same phrase (Holst and Nolan (1995). Although our stimuli were acoustically identical, listeners tended to perceive target sibilants as more ambiguous when the following /ʃ/ was part of the same phrase ($p = 0.059$), suggesting a role for top-down knowledge in interpreting segmental information.

5aSCb32. Exploring vowel mergers in northeast Ohio. Anna M. Schmidt and Kristin M. Weaver (Speech Pathol. & Audiol., Kent State Univ., A104 MSP, Kent, OH 444242, aschmidt@kent.edu)

The northeast corner of Ohio, as listed on dialect area maps, contains the boundaries of several dialect regions. This study compared perception and production for speakers from northeast Ohio and surrounding areas who merge and do not merge back vowels before /l/ in words such as "pool, pull, pole, dull." Acoustic formant analysis indicated a lowered and fronted vowel, as expected, for those who merge some or all of these types of words but different patterns of merging were seen, especially when a final syllable was added as in "pulling." Formant values were compared with values in "Luke" and "look." Perceptually, preliminary results indicate that the merged vowel is often labeled as /u/ unlike findings for other US dialect regions such as Utah. Productions from over 100 speakers were situated on detailed dialect maps.

5aSCb33. Automating phonetic measurement: The case of voice onset time. Neville Ryant, Jiahong Yuan, and Mark Liberman (Linguist. Data Consortium, Univ. of Pennsylvania, 3600 Market St., Ste. 810, Philadelphia, PA 19104, nryant@gmail.com)

Of 58 papers published so far this year in Journal of Phonetics, 16 (28%) feature Voice Onset Time (VOT) or related measurements, confirming that VOT remains a central concern in the field. However, phoneticians' VOT measurements generally continue to rely on human judgment, which requires significant labor, makes even large laboratory experiments onerous, and prevents the field from taking full advantage of the millions of hours of digital speech now becoming available. We present an algorithm for accurate automatic measurement of VOT, combining HMM forced alignment for determining approximate stop boundaries with paired burst and voicing

onset detectors. Each detector is a frame-level max margin classifier operating on the scale-space projection of a small number of relevant acoustic features. On a large set of clean lab speech, this system has a mean absolute error (relative to human annotation) of only 2.8 ms, with 98% of errors <10 ms. On a subcorpus independently annotated by two of the authors, the system agreed with the two human annotators as well as they agreed with one another (1.49 vs 1.50 ms). Promising results on other datasets will be reported. The system will be released as open-source software.

5aSCb34. An electromagnetic articulography investigation of the Czech trill-fricative. Phil Howson (The Dept. of Linguist, The Univ. of Toronto, Sidney Smith Hall, 4th Fl. 100 St. George St., Toronto, ON M5S 3G3, Canada, phil.howson@mail.utoronto.ca), Chris Neufeld (The Dept. of Speech-Language Pathol., The Univ. of Toronto, Toronto, BC, Canada), and Alexei Kochetov (The Dept. of Linguist, The Univ. of Toronto, Toronto, British Columbia, Canada)

Previous studies have found that the degree of tongue grooving during production of fricatives correlates with their place of articulation. Czech has a cross-linguistically rare alveolar trill-fricative <ř>, which would be expected to pattern with alveolar fricatives in terms of the tongue grooving. The current study employs electromagnetic articulography (EMA) to investigate differences in tongue grooving between the trill-fricative and alveolar/post-alveolar fricatives. Czech native speakers produced words with target consonants in word-initial, intervocalic, and word-final positions, with sensors being attached to both the midline and the sides of the tongue. An angle between these sensors was calculated, and taken as a measure of the degree of grooving. The results (currently based on the data obtained from one speaker) showed that, contrary to the prediction, the degree of grooving for the trill-fricative was closer to the post-alveolar fricatives than to the alveolar fricatives, yet unique in its distribution. While the degree of grooving remained relatively stable temporally for the fricatives, it tended to gradually decrease for the trill-fricative. Furthermore, <ř> differed from the post-alveolar fricatives in having a significantly lower tongue body. The results thus suggest a unique articulatory configuration for the Czech trill-fricative. [Work supported by SSHRC.]

5aSCb35. Perception of vowel-inherent spectral change. Kathleen Chiddenton and Michael Kiefte (School of Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, kathleen.chiddenton@dal.ca)

One family of theories regarding vowel perception suggests that onset and offset formant-frequencies are important for identification but that the shape of the transitions themselves is not perceptually important [e.g., Morrison and Nearey, J. Acoust. Soc. Am. **122**, EL15–EL22 (2007)]. The present study determined just-noticeable-differences in deviations for linear formant trajectories. Diphthong-like stimuli were manipulated by inserting a point of inflection into the otherwise linear transition. Several parameters were manipulated including vowel duration, location of the inflection point in time, direction of formant change, and fundamental frequency. Data from the first experiment indicate that listeners are largely insensitive to deviations from linearity of formant trajectory but that deviations that fell outside the range of frequencies spanned by the onset and offset were detectable. However, a second experiment in which only the first half of stimuli were presented gave different results. Results from these experiments along with several hypotheses are presented. [Work supported by SSHRC.]

5aSCb36. Modeling of vowel-inherent spectral change in spontaneous and elicited speech. Michael Kiefte (School of Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, mkiefte@dal.ca) and Terrance M. Nearey (Dept. of Linguist, Univ. of Alberta, Edmonton, AB, Canada)

A database of speech samples was collected to examining vowel formant patterns in spontaneous and read speech. Recordings were made from 220 long-time residents of Nova Scotia and Prince Edward Island in two separate tasks: Participants read a series of sentences in several phonetic contexts and also provided a monologue of at least 10 min duration resulting in approximately 36 h of spontaneous speech. Vowel formant frequencies were measured for all stressed vowels in non-function words surrounded by obstruents. Speakers were divided among eight geographic regions. Consonant contexts were modelled via nonlinear regression based on a model inspired by Broad and Clermont (J. Acoust. Soc. Am. **81**,155). Spontaneous speech was characterized by retraction of front vowels, fronting of /u/, backing of /æ/, and greater diphthongization of /ɒ/. Older speakers were less likely to show a merger between /ɒ/ and /ɔ/. There was also some evidence of a partial Canadian Shift whereby /æ/ and /ɛ/ are lowered following merger of /ɒ/ and /ɔ/. Results from alternate models currently under development, which may better accommodate vowel-inherent formant movement, will be discussed. [Work supported by SSHRC.]

5aSCb37. Development of a vocal tract design tool based on a growth curve of the vocal tract length. Daisuke Ito and Kohichi Ogata (Ogata Lab., Graduate School of Sci. and Technol., Kumamoto Univ., 2-39-1 Kurokami, Chuo-ku, Kumamoto 860-8555, Japan, ito@st.cs.kumamoto-u.ac.jp)

One of the advantages of speech synthesis based on the vocal tract shapes is flexibility in speech sounds depending on the articulatory parameters. In order to take advantage of the flexibility, preparing various vocal tract shapes is important. To achieve this, this paper describes the development of a vocal tract design tool based on a growth curve of the vocal tract length reported by literature. In this tool, the growth curve is used to calculate the size of the vocal tract. A user can obtain a vocal tract shape depending on the age and sex using a slide button on the interface window. The parameters that describe vocal tract shape and vocal folds are used to produce vowel sounds and their formant frequencies are obtained. According to literature, there is a relationship between the fundamental frequency and formant frequencies through the age. We compared the formant frequencies calculated from the tool with those estimated from the literature. The comparison showed that the calculated formant frequencies were in relatively good agreement with the estimated ones. In addition, the distribution of the first and second formant frequencies showed a typical pentagonal distribution.

5aSCb38. Formant-based articulatory normalization and its application to vowel restoration. Yuichi Ueda, Kosuke Tominaga, and Tadashi Sakata (Graduate School of Sci. and Technol., Kumamoto Univ., 2-39-1 kurokami, chuoh-ku, kumamoto-shi, kumamoto 860-8555, Japan, ueda@cs.kumamoto-u.ac.jp)

Visual feedback of spontaneous speech is effective for articulatory training of deaf children and for speech rehabilitation of dysarthric patients. Especially, the visual representations of vowel formant frequencies have been used directly or indirectly for those purposes, because those acoustical parameters reflect the articulatory behavior. However, since not only the shape of the vocal tract but its size also affect the formant frequencies, minimization of the effect due to the differences in size is required. In such a speaker normalization, we defined a color space consisting of three circular ratios of formant frequencies and applied it to the color visualization of vowel sound. In this paper, we proposed a normalized articulation space as an expansion of the color space, where we assumed that neutral vowels of any speaker are mapped into a unique point. In addition, since the proposed articulatory space was regarded as the speaker independent representation of vocal tract shape, we also proposed a method to convert the modified articulation shape into the real formant space and applied it to the vowel restorations of disarthric speech.

5aSCb39. On distinguishing articulatory configurations and articulatory tasks: Tamil retroflex consonants. Caitlin Smith (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, smithcm@usc.edu), Michael Proctor (Linguist, Univ. of Western Sydney, Sydney, NSW, Australia), Khalil Iskarous, Louis Goldstein (Linguistics, Univ. of Southern California, Los Angeles, CA), and Shrikanth Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Speech production can be described in multiple coordinate frames: articulatory configurations, gestural tasks, and acoustic patterns. Examination of the achievement of retroflex stops and liquids in Tamil suggests that we must consider separately the gestural task of apical pre-palatal constriction and the articulatory maneuver to achieve the task. The maneuver of the tongue during retroflex consonants varies across vowel contexts. Specifically, in the symmetrical intervocalic contexts between back vowels /a/ and /u/, an apical pre-palatal constriction is achieved by curling back the tongue. In the context of high front vowel /i/, a laminal pre-palatal constriction is achieved by bunching the tongue. However, the location of retroflex consonant constriction within the vocal tract is consistent across all of these vowel contexts, suggesting that the constriction task remains the same. Variation in the articulatory configuration of the retroflex in the two contexts was quantified through Gaussian curvature functions at fourteen points along the tongue, sampled at evenly spaced points throughout the vocal tract, on every other gridline of a polar-rectangular grid in every frame in each utterance. The empirical results support the notion that the articulatory configuration coordinate frame and the gestural task frame provide separate, but related, descriptions of speech production.

5aSCb40. C-V coarticulation in velar plosives. Vicki L. Krebs (Linguistics, The Ohio State Univ., 301 W. Euclid Ave., Springfield, OH 45506, krebs.86@buckeyemail.osu.edu), Yourdanis E. Sedarous (Linguistics, The Ohio State Univ., Grove City, OH), and Amanda L. Miller (Linguistics, The Ohio State Univ., Columbus, OH)

We present 114 fps lingual ultrasound data of three speakers' productions of words containing the initial velar plosive, and the following [a] and [i] vowels, in Mangetti Dune !Xung (N = 95 for [k] in the [a] context, N = 36 for [k] in the [i] context). We traced the midsaggital tongue edge at the frame just prior to the [k] release, and measured the tongue dorsum (TD) and tongue root (TR) constriction locations (CL's). The TRCL was measured 1 cm below the [k] peak. Results show that [k] in the [i] context has a 1.1 cm further forward TDCL than [k] in the [a] context for one speaker, and 1.2 cm further forward for the second speaker. The results are similar to those found for English by Stevens and House (1963). The TRCL is retracted 0.3 cm in [k] in the [a] context compared with [k] in the [i] context for the first speaker, and 0.6 cm more retracted for the second speaker. A third speaker had a retracted TDCL and TRCL in both vowel contexts. These results confirm that [k] is less resistant to coarticulation. Results show that the tongue root is involved in dorsal-front vowel coarticulation.

5aSCb41. The "panphonic" text of "The North Wind and the Sun" for the illustration of the International Phonetic Alphabet of Japanese consonants and its use in the phonetic analysis of Japanese speech. Shizuo Hiki (Waseda Univ., 5246-90-1-402 Yamaguchi, Tokorozawa 359-1145, Japan, hiki@waseda.jp) and Kuniko Kakita (Toyama Prefectural Univ., Izumi, Toyama, Japan)

A "panphonic" version of the text of "The North Wind and the Sun" for the illustration of the IPA of Japanese (Tokyo dialect) consonants has previously been devised by the present authors [Hiki *et al*., Proc. 17th Int'l. Cong. on Phonetic Sciences, Hong Kong, 871–873 (2011)]. The present paper describes the main characteristics of this panphonic text in relation to its use in the phonetic analysis of Japanese speech utterances, providing examples of the result of the analysis of sample recitations. The panphonic text embraces all 16 consonant phonemes, their 11 major positional allophones, and 5 free variants in a short, simple text that consists of 8 sentences (113 words, 294 syllables). The relation among consonant phonemes and their allophones is shown effectively by a new arrangement of rows and columns in the IPA consonant chart. Possible pause locations are systematically indicated using appropriate pause symbols. The text is useful in examining the phonetic properties of Japanese utterances, for example, the effect of consonants on vowel devoicing, and large-scale segmentation of utterances by pauses of different durations.

5aSCb42. Perception and production in non-native speech: Russian palatalization. Leandro Bolanos (Linguistics, Yale Univ., 188 1/2 Willow St., Apt B, New Haven, CT 06511, leandro.bolanos@yale.edu)

It is well known that adults struggle in perceiving and producing certain phonological contrasts not present in their native language. Adults also find difficulty in learning the specific timing of non-native articulatory gestures and contextual differences present in the language. The present study investigates English speakers' perception and production of Russian contrasts involving palatalized consonants in varying contexts. Of interest are the effects of syllable position and palatalization on speakers' performance in perception and production. The framework of Articulatory Phonology [e.g., Browman and Goldstein (1986), (1992)] and the Perceptual Assimilation Model [e.g., Best *et al*. (2001)] are adopted to account for differences in timing between English and Russian with respect to palatalization, and to subsequently make predictions on English speakers' perception as well as their production of the different timing property present in Russian palatalization. Speakers of American English lacking any previous exposure to Russian participated in a series of perception and production experiments involving Russian palatalized stops, which vary in place of articulation (labials, coronals) and syllable position (onset, coda). Preliminary results indicate diminished performance for some contrasts in syllable coda position as well as correlations between the perception and production of palatalized consonants by English speakers.

5aSCb43. Analysis of stop consonants in Devanagari alphabet. Kushagra Singh and Nachiketa Tiwari (Mech. Eng., Indian Inst. of Technol. Kanpur, C110/9, IIT Kanpur, Kanpur, Uttar Pradesh 208016, India, kushagrs@iitk.ac.in)

The Devanagari alphabet, which is used by several Indian languages including Sanskrit and Hindi, has vowels and consonants are placed in tabular format, which are arranged according to how they originate. A part of this table is a 5 x 5 matrix and comprises of stop consonants, where different rows corresponding to velar, palatal, retro-flex, dental and labial consonants. In this paper, we have explored patterns that exist between different consonant sounds belonging to different rows and columns of this table. Toward this end, four sound samples from individuals have been recorded, and analyzed. Our analysis shows the existence of many interesting relationships, which exist between sounds populating different rows and columns of 5 X 5 matrix. One interesting observation which has been made is that the fundamental differences between 1st and 2nd/ 3rd or 4th member of each row are essentially the same in all the rows, but a few exceptions in the 5th row. For example, in all rows the 3rd member is a combination of a short-duration signal and the 1st member. Similarly, the 2nd member is 1st member with a nonzero mean pressure line. Further, the 4th member is found to be the combination of the other two (1–2 and 1–3) variations. In this paper, we present several such interesting relationships. These relationships may be potentially useful in several sound processing algorithms.

5aSCb44. Estimated relative vocal tract lengths from vowel spectra based on fundamental frequency adaptive analyses and their relations to relevant physical data of speakers. Mayuko Kobayashi, Ryuichi Nisimura, Toshio Irino, and Hideki Kawahara (Design Information Sci., Wakayama Univ., 930 Sakaedani, Wakayama, Japan, s130043@center.wakayama-u.ac.jp)

A Japanese vowel database of males, females and children speakers (385 speakers in total) along with relevant physical data [Deguchi *et al*. (2011)] was analyzed using a set of F0 adaptive procedures, which were developed for a speech analysis, modification and synthesis framework TANDEM-STRAIGHT [Kawahara *et al*. (2008)] and its extensions. By restricting spectral region in estimating ratios for vocal tract length normalization (VTLN), cepstrally weighted distance measure yielded estimates with 2% standard error, by assuming relative vocal tract lengths estimated using regression analysis as the ground truth. A set of regression analyses of the estimated relative vocal tract lengths, average F0s, ages, gender, and physical data (speakers' height, weight) were conducted. The results suggest that the proposed analysis procedures applied to the Japanese five vowels may provide sufficient information for estimating speakers' physical data. Possible applications of the proposed estimation procedures will also be discussed.

5aSCb45. Perception of /ra/-/la/ contrast in different contexts: mono-syllable vs. sentence. Kanako Tomaru and Takayuki Arai (Sophia Univ., 7-1 Kioi-cho, Chiyodaku 1028554, Japan, himawari.kanako@gmail.com)

A strict assumption that underlies categorical perception hypotheses is that two speech sounds are discriminable only when they cross a categorical boundary emerging from an identification function. A number of researchers have attempted to show the categorical perception of the sounds in question (particularly consonants) through discrimination and identification tests. However, these tests are usually used for mono-syllables or mono-syllabic words. In this study, we first investigated whether the perception of /ra/-/la/ contrast indicate any categorical perception in a mono-syllabic CV context, using synthesized syllables. Next, we tested whether categorical perception can be also observed in a sentence through perceptual discrimination and identification experiments. Results showed that (1) a discrimination peak

predicted by the identification function was obtained only for the mono-syllabic context, and (2) discrimination accuracy in the sentence condition was consistently low. These results suggest that categorical perception in a strict sense may not be evident in the perception of a sentence.

5aSCb46. Pronunciation of German suffixes by Japanese native speakers of different proficiency levels. Marino Kasuya and Takayuki Arai (Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku Tokyo 102-8554, Japan, mar3576ino_7@hotmail.co.jp)

This study investigates an aspect of speech rhythm in German spoken by Japanese native speakers of different proficiency levels. Previous studies on the production of vowel reduction have indicated that this is an area of difficulty for non-native speakers. One study, working on the assumption that second language (L2) speech production is affected by first language (L1), suggested that Japanese native speakers tend to fail at producing the required vowel reductions in unstressed syllables. The present study further investigated this issue by dividing Japanese native speakers into two groups: advanced and elementary learners. The aim of the present study was to investigate acoustic properties of vowel quality (first and second formants) and quantity (durational ratio) of unstressed syllables in German suffixes on the basis of German proficiency levels and the influence of L1. From results, main effect was obtained for the proficiency levels; acoustic analysis showed significant differences between first two formants and durational ratio same to be the factors that caused the difference among the levels. This suggests that L2 learning process may accompany the acquisition of L2 sounds even when rhythmic structures differ between L1 and L2. [Work supported by JSPS.]

5aSCb47. A comparative cross-linguistic study of vocal tract shaping in sibilant fricatives in English, Serbian and Mandarin using real-time magnetic resonance imaging. Li Hsuan Lu, Adam Lammert, Vikram Ramanarayanan, and Shrikanth Narayanan (Univ. of Southern California, 3771 McClintock Ave., 4013A, Los Angeles, CA 90089, lihsuanl@usc.edu)

An articulatory study of sibilant fricatives is described, with the goal of describing variability in lingual articulation across languages. Real-time magnetic resonance imaging (rtMRI) data were collected from three speakers each of English and Mandarin and two speakers of Serbian and reconstructed at a rate of 22.4 frames per second. Parallel acoustic data were also collected and subsequently denoised. Subjects spoke the segments /s/ and /S/ in symmetrical vowel contexts (e.g., "pa sap" for English and "asa" for Mandarin). Articulation was analyzed using a semi-polar grid overlaid on the image plane and midsagittal distance functions were obtained by measuring cross distances at ~0.5 cm intervals from the glottis to the lips. Analysis shows that place of articulation for /s/ is more anterior compared to /S/ across languages. Apical articulation is observed for /s/ across languages, while /S/ is produced laminally in English and apically in Serbian and Mandarin. Patterns of tongue shaping variability differ, as well, across languages. For instance, higher standard deviation is observed anterior to the place of articulation for /S/ in Mandarin, compared to Serbian and English. [Work supported by NIH.]

5aSCb48. Acoustic characteristics of glottalized obstruents in Gitksan. Michael D. Schwan (Dept. of Linguist, The Univ. of British Columbia, 2329 West Mall, Vancouver, BC V6T 1Z4, Canada, michael.d.schwan@gmail.com)

Glottalized obstruents are a defining feature in the phonetic inventory of languages of the Pacific Northwest. Gitksan (Tsimshianic), an endangered and understudied language in this region, is no exception. However, these segments, which have typically been labelled as ejectives by fieldworkers, have also been variously described as implosives or even as voiced ejectives. Evidently, the ability of fieldworkers to perceive these segments has proven difficult, even by those who have worked on the language for many years. This project seeks to describe some of the salient acoustic cues associated with glottalized obstruents in Gitksan by comparing glottalized and plain stops. While previous work has examined these stops only in word-initial position, the present study compares stops across positions within the word and across stressed and non-stressed environments. Speech tokens were collected from three fluent native speakers of three dialects of Gitksan in order to describe the prominent acoustic cues which characterize glottalized obstruents in Gitksan.

5aSCb49. Simulation of neural mechanism for Chinese vowel perception with neural network model. Chao-Min Wu, Ming-Hong Li, and Tao-Wei Wang (National Central Univ., #300, Chung-Da Rd., Chung-Li 32001, Taiwan, wucm@ee.ncu.edu.tw)

Based on the results of psycholinguistic experiments, the perceptual magnet effect is the important factor in speech development. This effect produced a warped auditory space to the corresponding phoneme. The purpose of this study was to develop a neural network model in simulation of speech perception. The neural network model with unsupervised learning was used to determine the phonetic categories of phoneme according to the formant frequencies of the vowels. The modified self-organizing map (SOM) algorithm was proposed to produce the auditory perceptual space of English vowels. Simulated results were compared with findings from psycholinguistic experiments, such as categorization of English /r/ and /l/ and prototype and non-prototype vowels, to indicate the model's ability to produce auditory perception space. In addition, this speech perception model was combined with the neural network model (Directions into Velocities Articulator, DIVA) to simulate categorization of ten English vowels and their productions to show the learning capability of speech perception and production. We further extended this modified DIVA model to show its capability to categorize six Chinese vowels (/a/, /i/, /u/, /e/, /o/, /y/) and their productions. Finally, this study proposed further development and related discussions for this speech perception model and its clinical application.

5aSCb50. Vowel onset marker based objective evaluation of Japanese phonemic length contrast produced by non-native speakers. Mee Sonu (Faculty of Sci. and Technol., Sophia Univ., 7-1 Kioi-Cho, Chiyoda-ku, Tokyo 102-8554, Japan, sonumeephonetic@gma1l.com), Yanlong Zhang (Global Information and Telecommunication Inst., Waseda Univ., Tokyo, Japan), Hiroaki Kato (National Inst. of Information and Commun. Technol., Kyoto, Japan), and Yoshinori Sagisaka (GITI/LASS Lab., Waseda Univ., Tokyo, Japan)

Aiming at building up an objective evaluation of second language (L2) learner's Japanese timing control characteristics, this study propose an objective measure to simulate a subjective measure of L2 speakers' production given by Japanese native evaluators from the view point of goodness of production. The focus here is on phonemic length contrast, e.g., /kako/ "the past" versus /kakko/ "parentheses" and /kaze/ "wind" versus /kaze:/ "taxation" which is difficult for L2 learners particularly when incorporated with speaking rates. The proposed objective measure uses a vowel onset time marker as a key perceptual and psychoacoustic marker to normalize speaking rate variations. The proposed new measure reflects tempo normalization between L2 learners by dividing the proficiency of mora-timing control in production. Results show that both vowel length contrast and consonant length contrast have a significantly higher correlation with the subjective evaluation score in which the coefficient was stable than using a simple duration difference measure. These results suggest that applying the psychoacoustic parameters would be effective to build up an objective evaluation of L2 learners. [Work supported by JSPS.]

5aSCb51. Canadian oats and Canadian goats: Comparing distal cues to segmentation and segments. Christopher C. Heffner (Neurosci. and Cognit. Sci., Univ. of Maryland, 1401 Marie Mount Hall, College Park, MD 20742, heffner@umd.edu) and Rochelle S. Newman (Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

Distal timing cues (specifically, the speech rate of sentences temporally removed from a point of ambiguity in speech) have been shown to weakly modulate segmental perception [i.e., the perception of basic speech sounds, like "p" and "b"; Newman and Sawusch (1996)] but strongly affect the perception of word boundaries [i.e., the perception of separation between words in fluent speech; Dilley and Pitt (2010)]. However, no study as of yet has directly compared the two classes of percept using identical manipulations. In this study, we will examine the role of distal timing cues to segmental perception and word segmentation using the same distal contexts (e.g., "The merchant sold Canadian oats/notes," a word boundary distinction, or "The merchant sold Canadian coats/goats," a voicing-based segmental distinction). Distal speech rate will be artificially slowed to an identical extent for both types of contrast. We predict that categorical perception leads distal context effects to be much stronger on word segmentation ("…Canadian

oats/notes") than on segmental perception ("…Canadian coats/goats"). This research may help bridge speech perception theories that have been developed for each class of percept, and further clarify the role of distal cues in speech perception.

5aSCb52. The role of duration in regional U.S. vowel shifts. Valerie Fridland (English Dept., Univ. of Nevada, Reno, English Dept, UNR, MS 0098, Reno, NV 89557, fridland@unr.edu), Tyler Kendall, and Charlie Farrington (Dept. of Linguist, Univ. of Oregon, Eugene, OR)

Contemporary American dialect research has produced a rich body of literature on phonetic differences in vowel production. Many studies have focused on contrasting vowel shifts occurring in the speech of Northerners and Southerners. While such research clearly shows formant differences across dialects, the investigation of regionally variable phonetic cues beyond F1 and F2 has been under-addressed. The limited work that exists on sociolinguistic aspects of duration suggests that speakers do show durational differences across regions [Clopper *et al*. (2005), Jacewicz *et al*. (2007), Labov and Baranowski (2006), Tauberer and Evanini (2009)]. However, beyond these studies, there has been little research on the regional variability of vowel duration and on how phonemically non-distinctive durational differences found across regions may be related to the degree and type of spectral shift present in individuals' systems. In this paper we consider the extent to which duration corresponds to spectral differences in the major U.S. vowel shifts. In addition to looking at overall regional patterns, we examine whether durational differences co-vary with degree of individuals' participation in spectral shift. We discuss how duration, particularly of lax vowels, is related to the advancement of vowel shift features and whether such a relationship supports a physiological or grammatical explanation for duration.

5aSCb53. One small step for (a) man: Function word reduction and acoustic ambiguity. Laura Dilley, Melissa M. Baese-Berk, Stephanie Schmidt, Jesse Nagel (Dept. of Communicative Sci. and Disord., Michigan State Univ., Oyer Center, Michigan State Univ., East Lansing, MI 48824, mbaese@msu.edu), Tuuli Morrill (Dept. of Psych., Michigan State Univ., East Lansing, MI), and Mark Pitt (Dept. of Psych., The Ohio State Univ., Columbus, OH)

"That's one small step for man, one giant leap for mankind." Neil Armstrong insisted for years that his famous quote upon landing on the moon was misheard, and that he had said "one small step for a man." This controversy has continued, as examinations of the sound files of his transmission have yielded mixed opinions about whether he produced a. The disagreement stems partly from the fact that function words like a can be acoustically fleeting in casual speech, consist of just a few pitch periods, and be spectrally indistinguishable from the preceding context. As a result, they can be perceptually fragile, and easily disappear if the rate of surrounding speech varies [Dilley and Pitt, Psychological Science (2010)]. Here, we examine naturally produced, reduced tokens of for (spoken as "fer"), which were or were not followed by the word a from the Buckeye Speech Corpus, which consists of speakers from Mr. Armstrong's home state of Ohio. Comparison of the acoustic properties of the two sets of tokens will provide an indication of how similar they can be. Inclusion of Mr. Armstrong's production will assist in evaluating the likelihood of the function word being spoken. [Work supported by NSF Grant No. BCS-0847653.]

5aSCb54. Manipulating phonological boundaries using distributional cues. Eric Schreiber (Psychology, McGill Univ., 3701 Blackthorn Ct, Chevy Chase, Maryland 20815, eric.schreiber@mail.mcgill.ca), Kristine H. Onishi (Psychology, McGill Univ., Montreal, QC, Canada), and Meghan Clayards (School of Commun. Sci. and Disord. & Linguist, McGill Univ., Montreal, QC, Canada)

This study investigates the role of phonetic variability in category formation. The category boundary between /n/ and /m/ was manipulated in English-speaking adults. In Experiment 1 ($n = 24$), categories were indicated by lexical (novel word-object pairings) and distributional (within-category variability, category mean) information. During pre- and post-tests, listeners categorized stimuli from a 10-step re-synthesized continuum from "nado" to "mado." During training participants heard a bimodal distribution of tokens with one mode spanning 7 steps (wide) and one spanning 3 (narrow). For half the participants the wide category was "nado" and for the other half it was "mado." During training, tokens from each category were paired with different objects (socks or balls). A repeated-measures ANOVA showed that after training, categorization shifted toward the wide distribution [$F(1,22) = 10.259, p < 0.01$] and more so for ambiguous steps [$F(9,198) = 5.283$, $p < 0.001$]. In Experiment 2 ($n = 12$), auditory information was identical to experiment 1, but lexical information about the categories was removed (all training tokens referred to the same object). Listeners again shifted categorization toward the wide distribution [$F(1,10) = 7.686$, $p < 0.05$], indicating that distributional information alone was sufficient to change categorization behavior.

5aSCb55. C-V coarticulation in consonants with multiple lingual constrictions. Amanda L. Miller (Linguistics, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210-1298, amiller@ling.osu.edu)

C-V coarticulation in monosyllabic words containing initial click consonants and /i:/ vowels is investigated in Mangetti Dune !Xung with 114 fps lingual ultrasound and acoustic data collected using the CHAUSA method [Miller and Finch (2011)]. The 114 fps rate yields an image of the tongue every 9 ms (+/−4.5 ms). Vowels following clicks have three lingual gestures involving the tongue tip/blade (TT), tongue body (TB), and tongue root (TR). TT and TB constrictions carried over from the clicks merge into a single vowel constriction at consonant specific rates. The second formant (F2) distinguishes each word type through the vowel midpoint. In regression analyses, TBCL and TRCL best predict F2 for alveolar click initial words, while TTCL best predicts F2 for dental/palatal click initial words. The more open constriction is acoustically inert. In the palatal click initial word, both constrictions are equally close for some speakers, and the gestures undergo blending [Browman and Goldstein (1990)]. I argue that these patterns are prosodically controlled. Dental and palatal clicks have TB and TR gestures associated with the syllable onset. In alveolar clicks, the right edge of TB and TR gestures are aligned to the right edge of the first mora.

5aSCb56. Examining the extent of anticipatory coronal coarticulation: A long-term average spectrum analysis. Alexei Kochetov (Linguistics, Univ. of Toronto, 100 St George St., 4th Fl., Toronto, ON M5S 3G3, Canada, al.kochetov@utoronto.ca) and Chris Neufeld (Speech-Language Pathol., Univ. of Toronto, Toronto, ON, Canada)

Phonetic studies of English liquids /r/ and /l/ have shown these consonants can exert strong coarticulatory effects on both adjacent and non-adjacent vowels. The studies of long-range coarticulation, however, have so far been limited to British English and have not explicitly compared liquids to other coronal consonants. The current study investigated local and long-range effects of coronals /l/, /r/, and /d/ in Canadian English. 14 speakers were recorded reading the sentences "We thought it might be(a) ram/lamb/dam/ham" repeatedly. Long-term average spectra of five vowels preceding the target consonants were calculated and compared to baseline values. The results revealed significant differences between the coronal consonants and the control (/h/) in up to five preceding syllables. Significant differences among the three coronals, however, were limited to the immediately preceding vowel. The other vowels showed some differences between /r/ and /d/, but not between /l/ and the other coronals. The results show that long-range coarticulation can spread for up to five syllables and involve both liquid and non-liquid coronals. The spectral differences between the two liquids in Canadian English, however, may not be as robust as have been reported for British English.

5aSCb57. A preliminary ultrasound study of Nepali lingual articulations. Alexei Kochetov (Linguistics, Univ. of Toronto, 100 St. George St., Sidney Smith 4076, Toronto, ON M5S 3G3, Canada, al.kochetov@utoronto.ca), Marianne Pouplier (Institut für Phonetik und Sprachverarbeitung, Ludwig-Maximilians-Universität, Munich, Germany), and Sarah Truong (Linguistics, Univ. of Toronto, Toronto, ON, Canada)

Previous descriptive and phonetic works on Nepali provided conflicting accounts of place contrasts in coronal consonants. Specifically, apical stops were characterized as either retroflex or alveolar, while laminal affricates were described as either alveolar or palatal. Some of these works used static palatography, which shows the contact between the tongue and the palate,

but provides no information about the tongue shape for a given consonant or its dynamic properties. In this study we used ultrasound to image tongue shapes for various Nepali lingual consonants produced by a single native speaker of Brahmin dialect. The results showed that the speaker's apical stops were produced with a substantially raised tongue front and retracted tongue tip, as would be expected of retroflex articulations. Laminal affricates had the tongue shape similar to dental stops, yet with a somewhat retracted tongue tip, indicative of the alveolar constriction. Apicals that differed in laryngeal features (voiceless, voiced, aspirated, breathy) did not show systematic differences in the tongue shape, except for the voiced stop, which was somewhat less retracted. While limited to the single speaker, the results confirm and extend some previous observations about Nepali coronals as showing a 3-way place contrast among dentals, alveolars, and retroflexes.

FRIDAY MORNING, 7 JUNE 2013 — 511AD, 9:00 A.M. TO 12:20 P.M.

Session 5aUW

Underwater Acoustics and Signal Processing in Acoustics: Using Graphic Processing Units for Computationally Intensive Applications in Acoustic Modeling and Signal Processing

Paul Hursky, Cochair
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Stan E. Dosso, Cochair
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Invited Papers

9:00

5aUW1. Large-scale virtual acoustics simulation at audio rates using three dimensional finite difference time domain and multiple graphics processing units. Craig Webb and Alan Gray (Univ. of Edinburgh, 1/3 Flat 1, Drummond St., Edinburgh EH8 9TT, United Kingdom, C.J.Webb-2@sms.ed.ac.uk)

The computation of large-scale virtual acoustics using the 3D finite difference time domain (FDTD) is prohibitively computationally expensive, especially at high audio sample rates, when using traditional CPUs. In recent years the computer gaming industry has driven the development of extremely powerful graphics processing units (GPUs). Through specialized development and tuning we can exploit the highly parallel GPU architecture to make such FDTD computations feasible.This paper describes the simultaneous use of multiple NVIDIA GPUs to compute schemes containing over a billion grid points. We examine the use of asynchronous halo transfers between cards, to hide the latency involved in transferring data, and overall computation time is considered with respect to variation in the size of the partition layers. As hardware memory poses limitations on the size of the room to be rendered, we also investigate the use of single precision arithmetic.This allows twice the domain space, compared with double precision, but results in phase shifting of the output with possible audible artifact. Using these techniques, large-scale spaces of several thousand cubic meters can be computed at 44.1 kHz in a usable time frame, making their use in room acoustics rendering and auralization applications possible in the near future.

9:20

5aUW2. Adapting the minimum variance beamformer to a graphics processing unit for active sonar imaging systems. Jo Inge Buskenes (Dept. of Informatics, Univ. of Oslo, Søndre Mohagen 17, Frogner 2016, Norway, joibu@ifi.uio.no), Jon Petter Åsen (MI-Lab, Norwegian Univ. of Sci. and Technol., Oslo, Norway), Carl-Inge C. Nilsen, and Andreas Austeng (Dept. of Informatics, Univ. of Oslo, Oslo, Norway)

The MVDR beamformer has been shown to improve active sonar image quality compared to conventional methods. Unfortunately, it is also significantly more computationally expensive because a spatial covariance matrix must be estimated and inverted for each image pixel. We target this challenge by altering and mapping the MVDR beamformer to a GPU, and suggest three different solutions depending on the system size. For systems with relatively few channels, we suggest arithmetic optimizations for the estimation step, and show how a GPU can be used to yield image creation rates of more than 1 Mpx/s. For larger systems we show that frequency domain processing is preferable, as this promotes high processing rates at a negligible reduction in image quality. These GPU implementations consistently reduced the runtime by 2–3 orders of magnitude compared to our reference C and Matlab implementations. For even larger systems we suggest employing the LCA beamformer. It does not calculate a weightset, but merely computes the beamformer output for each of a predefined set of weights, and selects the one that best fulfils the MVDR criterion. The LCA creates images with a quality comparable to MVDR, and it is perfectly suited for a GPU.

9:40

5aUW3. Exploiting data parallelism and population Monte Carlo on massively-parallel architectures for geoacoustic inversion. Jan Dettmer, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8W 3P6, Canada, jand@uvic.ca), and Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA)

Bayesian inference algorithms in geoacoustic inversion have high computational requirements on multiple computational scales. Predicting (modeling) data to match observations represents fine-grained computations which often cannot be implemented efficiently on CPU clusters since high latency and communication overhead outweigh parallelization gains. However, GPUs, which operate efficiently on 100,000s of parallel threads with low latency and high bandwidth, can provide significant performance gains. Bayesian sampling is generally coarse-grained, and can be implemented efficiently in parallel on multi-core/cluster architectures. For example, population Monte Carlo simulates many Markov chains in parallel, with chains running independently between interactions (at predefined intervals) which exchange information throughout the population, substantially increasing sampling efficiency. This paper combines fine- and coarse-grained parallelization to profoundly improve the efficiency of geoacoustic inversion of seabed reflection data. Spherical-wave reflection-coefficient predictions, which require solving the Sommerfeld integral for a large number of grazing angles and frequencies, constitute fine-grained, data-parallel computations which are implemented efficiently on a GPU. Sampling is based on population Monte Carlo simulation with chain interactions as exchange and crossover moves.The algorithm is applied to Malta Plateau data to study frequency dependence of velocity and attenuation in marine sediments. [Work supported by ONR.]

Contributed Papers

10:00

5aUW4. Efficient Bayesian multi-source localization using a graphics processing unit. Stan E. Dosso and Jan Dettmer (School of Earth & Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

This paper presents a highly efficient approach to matched-field localization of an unknown number of ocean acoustic sources employing a graphics processing unit (GPU) for massively parallel computations. A Bayesian formulation is developed in which the number, locations, and complex spectra (amplitudes and phases) of multiple sources, as well as noise variance at each frequency, are considered unknown random variables constrained by acoustic data and prior information. The number of sources is determined during an initial burn-in stage by minimizing the Bayesian information criterion using an efficient birth/death scheme. Marginal posterior probability distributions for source locations are then computed using Gibbs sampling. Source and noise spectra are sampled implicitly by applying analytic maximum-likelihood solutions in terms of the source locations (explicit parameters). This greatly reduces the dimensionality of the inversion, but requires solving a very large number (order 10^5) of complex matrix inversions for each sample of the explicit parameters. These inversions can be solved in parallel on a GPU, increasing efficiency by a factor of ~100. Examples are given of localizing a large number of sources (up to 10) in near real time.

10:20

5aUW5. Computing the singular value decomposition in parallel on graphics processing units using a one-sided Jacobi method. Michael V. Romer (Appl. Res. Labs. at the Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, romer@arlut.utexas.edu)

The singular value decomposition (SVD) provides a robust means of determining the dominant modes in a collection of signals that can then be used in adaptive beamforming to suppress loud interferers that would otherwise cover signals of interest. However, on serial architectures this decomposition can quickly become a bottleneck, hindering the real-time performance of the beamformer. Commercial off-the-shelf graphics processing units (GPUs) are an inexpensive means of adding parallel processing capabilities to a system and can reduce computation time by several orders of magnitude. The following work presents an algorithm that computes the full SVD of a dense matrix in parallel using a one-sided Jacobi method on a standard GPU. Both the runtime performance and relative accuracy of the algorithm are compared to the Intel Math Kernel Library (MKL) LAPACK implementation run on a CPU. Potential limitations to this approach due to restrictions imposed by the hardware are also discussed.

10:40

5aUW6. Interactive gpu-based sound auralization in dynamic scenes. Qi Mo, Micah Taylor, Anish Chandak, Christian Lauterbach, Carl Schissler, and Dinesh Manocha (Comput. Sci., Univ. of North Carolina at Chapel Hill, 201 S Columbia St., Chapel Hill, NC 27599, qmo@cs.unc.edu)

We present an auralization algorithm for interactive virtual environments with dynamic objects, sources, and listener. Our approach uses a modified image source method that computes propagation paths combining direct transmission, specular reflections, and edge diffractions up to a specified order. We use a novel multi-view raycasting algorithm for parallel computation of image sources on GPUs. Rays that intersect near diffracting edges are detected using barycentric coordinates and further propagated. In order to reduce the artifacts in audio rendering of dynamic scenes, we use a high order interpolation scheme that takes into account attenuation, cross-fading, and delay. The resulting system can perform auralization at interactive rates on a high-end PC with NVIDIA GTX 280 GPU with 2–3 orders of reflections and 1 order of diffraction. Overall, our approach can generate plausible sound rendering for game-like scenes with tens of thousands of triangles. We observe more than an order of magnitude improvement in computing propagation paths over prior techniques.

11:00

5aUW7. Some comments about graphic processing unit architectures applied to finite-difference time-domain room acoustics simulation: Present and future trends. Jose J. Lopez (ITEAM, Tech. Univ. of Valencia, Camino de Vera s/n, Valencia, Valencia 46022, Spain, jjlopez@dcom.upv.es), Juan M. Navarro (Telecommunications, Universidad Catolica de San Antonio, Guadalupe, Murcia, Spain), Diego Carnicero, and Jose Escolano (ITEAM, Tech. Univ. of Valencia, Valencia, Valencia, Spain)

The parallelization of the finite-difference time-domain (FDTD) method for room acoustic simulation using graphic processing units (GPUs) has been subject of study even prior to the introduction of general-purpose computing environments such as the CUDA architecture. Nowadays CUDA offers enough flexibility and processing power to obtain performance gains higher than 200 times compared to single-threaded CPU codes. In this paper, different aspects related to the implementation of FDTD in CUDA are analyzed; first, how the evolution of the different CUDA architectures affects implementations is inquired, paying special attention to the Kepler architecture, the latest available. Also performance increasing by using the different memory subsystems the GPU offers is discussed. Moreover, the performance in the use of the available computing power in the GPU is also analyzed together with the limiting factors such as memory consumption and computing time that prevent the simulation of large rooms at very high frequencies. Finally, some comments and ideas about the possible evolution

of the computing power of GPU in the next years are evaluated together with future possibilities that GPU architectures might bring to FDTD applied to room acoustics.

11:20

5aUW8. Efficient implementation of power-law attenuation of elastic waves in time-domain numerical simulations. David C. Calvo (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC, david.calvo@nrl.navy.mil) and Gaetano Canepa (Ctr. for Maritime Res. and Eng., La Spezia, Italy)

Numerical simulation of wave propagation in the time domain is easily parallelizable on high performance computing systems due to the spatially local nature of the governing equations. The disadvantage of working in the time domain arises when lossy media must be modeled which generally gives rise to convolution-type loss terms in the governing time-domain equations. Computation of these convolutions usually requires the storage of several solution fields at thousands of previous time steps. This requirement can be memory prohibitive in three-dimensions. In this talk we present a recursive convolution approach to computing lossy (power-law) elastic wave propagation that is an extension of the one-way, one-dimensional acoustic wave equation work done by Liebler [Liebler *et al*., J. Acoust. Soc. Am. **116** (2004)] in order to handle multiple dimensions and shear waves. Convolutions are computed recursively by first using a nonlinear least-squares technique to fit the kernel of the convolution with a series of decaying exponentials. We demonstrate how graphical processing units (GPUs) can be used to obtain speed-up factors as high as 35 on a test computation of time-domain scattering from a highly resonant but lossy elastic cylinder. [Work sponsored by the Office of Naval Research.]

11:40

5aUW9. The use of graphical processing unit processing in rough surface scattering. Ahmad T. Abawi and Paul Hursky (HLS Res., 3366 North Torrey Pines Court, La Jolla, CA 92037, abawi@hlsresearch.com)

The use of graphical processing units (GPU's) in scientific computation has drawn significant interest in recent years. In this paper we use GPU processing to evaluate the performance of a number of approximate techniques in computing scattering from two-dimensional rough surfaces by comparing their results with those obtained using the boundary element technique, which produces a numerically exact solution of the problem. To compute scattering from a two-dimensional surface, we use a technique that we developed for computing scattering from compact objects, which uses an analytical expression for scattering from a single, flat triangle. In this technique the surface is meshed using triangular patches and the scattering is computed as a coherent sum of scattering from individual triangles. This technique not only provides accurate evaluation of the surface integrals that appear in scattering theory, but it also lends itself easily to the benefits of GPU processing. We apply this technique to the Kirchhoff approximation, the small slope approximation and to a rather less familiar technique based on the work of Dashen *et al*. [J. Math. Phys.**32**, 986–996 (1991)].

12:00

5aUW10. Reverberation modeling on graphics processing units. Paul Hursky and Ahmad T. Abawi (HLS Res. Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

We will discuss modeling reverberation on GPUs. An accompanying talk will discuss using GPUs to model scattering from rough surfaces. Here we discuss using GPUs to model the two sets of propagation paths, one set from the source to scatterer, the other set from the scatterer to the receiver, and to combine these two sets via a scattering operation and assemble the reverberation waveform at the receiver. We will discuss how we have adapted various aspects of our modeling to a GPU platform. GPUs provide several differentiated memory architectures that an application can exploit. For example, the texture memory provides hardware-assisted interpolation—as a result, we can load a 3D environment into texture memory, sparsely sampled, and then reconstruct interpolated slices as needed the modeling task. GPU architectures have been evolving for many years to meet the demands of computer gaming and rendering, applications most ambitiously served by ray tracing (of light). As a result, NVIDIA provides a ray tracing framework called OptiX, sufficiently general-purpose, that it can be linked with externally provided functions to specialize the mathematics implemented during the ray trace process. We will describe our work on adapting OptiX to underwater acoustic ray tracing.

FRIDAY AFTERNOON, 7 JUNE 2013 — 514ABC, 1:00 P.M. TO 2:40 P.M.

Session 5pPP

Psychological and Physiological Acoustics: Recent Trends in Psychoacoustics II

Hugo Fastl, Cochair
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Sonoko Kuwano, Cochair
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Invited Papers

1:00

5pPP1. What can we learn from simulated acoustic environments? Bernhard U. Seeber (Audio Information Process., Technische Universität München, Arcisstrasse 21, Munich 80333, Germany, seeber@tum.de)

Psychoacoustic research has often used headphones to reproduce sound stimuli. Recently, the spatial dimension has regained attention in basic research and the talk will make a case for the importance of binaural hearing when assessing sound quality. Technological advances made it possible to accurately reproduce real and artificial sound stimuli with high spatial fidelity for their assessment. The Simulated Open Field Environment (SOFE) is a laboratory setup to reproduce sounds from multiple loudspeakers in an anechoic chamber. The free-field presentation allows participants to interact with sound stimuli in a natural way using head turns and movements - important when working with participants inexperienced with laboratory procedures. In connection with room simulation software the SOFE can also create acoustic scenes with multiple sources and sound reflections—thereby increasing the realism. I will give an overview of recent findings gained with the SOFE and their relation to evaluating sound quality.