Session 2aID

Interdisciplinary: Plenary Lecture: Basics and Applications of Psychoacoustics

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

Chair's Introduction—7:55

Invited Paper

8:00

2aID1. Basics and applications of psychoacoustics. Hugo Fastl (AG Technische Akustik, MMK, TU München, Arcisstr.21, München 80333, Germany, fastl@mmk.ei.tum.de)

The field of psychoacoustics studies relations between acoustic stimuli, defined in the physical domain, and the hearing sensations elicited by these stimuli. The plenary lecture will address questions of stimulus generation and presentation. Both traditional methods and more recent methods like wave field synthesis will be touched. Concerning hearing sensations, basic magnitudes as for example absolute thresholds or loudness, but also advanced topics like pitch strength will be covered. Applications of psychoacoustics will include rather different fields such as music, noise evaluation or audiology, and address also cognitive effects as well as audio-visual interactions.

TUESDAY MORNING, 4 JUNE 2013 513ABC, 8:55 A.M. TO 12:00 NOON

Session 2aAAa

Architectural Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Adapting, Enhancing, and Fictionalizing Room Acoustics I

K. Anthony Hoover, Cochair
McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362

Alex Case, Cochair
Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854

Chair's Introduction—8:55

Invited Papers

9:00

2aAAa1. Electronically variable room acoustics—Motivations and challenges. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com) and Steve Ellison (Meyer Sound, Sierra Madre, CA)

Electronically variable room acoustics, or "active acoustics," has become an effective solution to a variety of room acoustics challenges. The motivations for considering such systems often go beyond acoustics. Energy conservation, financial considerations, historic preservation, and balancing the needs of a venue's constituencies all can play a role in the determination to employ active acoustics. This paper will discuss some practical examples, including planned renovations to the Santa Monica Civic Auditorium, home of the Academy Awards during the 1960s, a resident orchestra, legendary rock concerts, and a unique hydraulic floor to convert the Civic from a performance space to an exhibit space. The active acoustic system objectives, design strategies, and challenges will be discussed.

9:20

2aAAa2. Sound system in a small college auditorium. Sergio Beristain (IMA, ESIME, IPN, P.O. Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

A new 220 seats college auditorium needed an electro acoustics system in order to adequately perform all its normal activities, which included live voices for lectures and conferences, a sound reproduction system working alone or together with the projection equipment, sometimes background music, and eventually some small groups with live music presentations as requested in the specs of the auditorium usage; it also needed recording capabilities for the most important lectures and presentations. A 12/8 channel with stereo output system with peripherals was chosen for the installation, where three microphones were reserved for the front table, six distributed in fixed stands located in the aisles of the audience area for questions and dialog, and the other three were movable. Background noise was not an issue because the auditorium is located in a tree full area within the university campus, away from busy streets. Budget for the acoustical conditioning and the electronic equipment was very limited.

9:40

2aAAa3. Active acoustics and sound reinforcement at TUI Operettenhaus, Hamburg: A case study. Roger W. Schwenke (Res. and Development, Meyer Sound Lab., 2832 San Pablo Ave., Berkeley, CA 94702, rogers@meyersound.com)

TUI Operettenhaus is a proscenium theater with one balcony, which is host to drama, musical theater, and concerts. The venue hosts different sound reinforcement systems for different shows, and now has a permanent active acoustic system. The physical acoustics are very dry as is appropriate for modern theater with spatial sound reinforcement, and the active acoustic system allows the reverberation time to be extended as appropriate for different performances. The active acoustic system can also pass through signals to its speakers for spatial surround reproduction. The installation of the active acoustic system in an older building posed many challenges. This case study presents the challenges that were overcome during installation, the integration of the active acoustic system with sound reproduction, and the measured performance of the system.

10:00

2aAAa4. The raw and the cooked in architectural acoustics. William L. Martens and Dagmar Reinhardt (Faculty of Architecture, Design and Planning, Univ. of Sydney, 148 City Rd., Wilkinson Bldg. G04, NSW 2006, Australia, william.martens@sydney.edu.au)

Whereas the "raw" experience of live sound events is often quite far removed from the "cooked" auditory imagery that is presented when live acoustical events are amplified by a sound reinforcement system, there are many audio signal processing tools that can be applied in the attempt to simulate the more natural auditory characteristics of live (unplugged?) musical performances. This paper builds a discussion of perceptual results of modern sound reinforcement technology based upon the Lévi-Strauss notion regarding what modern culture does to the "raw" to make it "cooked"). A key concept in evaluating the quality of a sound reinforcement system is that of the standard of reference against which the perceptual results can be compared. As there is no shared opinion nor well-established optimal acoustical character for a space upon which some consensus could be built, the question presents itself again and again. This paper will address related issues of reference, preference, and adequacy in sound reinforcement.

10:20

2aAAa5. Adapting spaciousness of artificial, enveloping reverberation in multichannel rendering based on coded sequences. Ning Xiang, Jiseong Oh, Jonas Braasch (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, Greene Building, 110 8th Street, Troy, NY 12180, xiangn@rpi.edu), and Bosun Xie (School of Sci., South China Univ. of Technol., Guangzhou, China)

For virtual room environments, adapting realistic reverberation and enhancing reverberation are critical for producing a convincing immersive experience. Also, in perceptual studies of room-acoustics using virtual room environments, using the appropriate enveloping reverberance to correlate perceived room size to the virtual space is a challenging task. This research applies to both binaural rendering and a multi-channel loudspeaker reproduction that can be employed in simulating such an environment. Approaches to adapting and enhancing spaciousness within the context of artificially generated reverberation are investigated via psychoacoustics tests. The pseudo-random properties of coded signals based on reciprocal maximum-length sequences allow for a deterministic, controllable decorrelation between all reverberation channels. For this challenging task, shapes of both sound energy decay and spatial profiles have been found to be decisive for creating successful immersive environments. This paper will discuss potential values for fundamental research in room-acoustics and for educational purposes in seeking a broadened understanding of perceived spaciousness and reverberance in varying contexts.

10:40

2aAAa6. E-Venue—Affordable electronic acoustic enhancement for small venues. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

The advent of modern digital signal processing made altering the acoustical conditions in a venue using electro-acoustic tools practical. Such systems have been in constant use for many years in concert halls, opera houses performance spaces, houses of worship, and a variety of other spaces and applications throughout the world. However, the cost associated with specialized nature of these systems has put them out of the reach of many small venues that stand to benefit most from use of this technology. This paper describes a new low cost, integrated, electro-acoustic system designed specifically for use in small venues—including but not limited to; performance venues, recital halls, rehearsal spaces, houses of worship, etc.

11:00

2aAAa7. Interaction between critical listening environment acoustics and listener reverberation preference. Brett Leonard, Richard King (The Grad. Program in Sound Recording, The Schulich School of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, brett.leonard@mcgill.ca), and Grzegorz Sikora (Bang & Olufsen Deutchland GmbH, Pullach, Germany)

Reverberation is a central effect in many modern music productions. In the case of classical music, it may even be the only effect used. There is, however, minimal literature concerning the interaction between reverberation preference and the listening environment used during critical mixing tasks. In order to explore this critical interaction, a group of highly trained subjects are tasked with adding reverberation to dry, premixed stereo program material in two different acoustic environments: a recording studio control room and a highly reflective room. The control room is representative of most studios, with an RT of approximately 200 ms. The reflective environment more closely approximates an untreated residential room, with an RT of over 350 ms, with a marked increase in lateral energy. Somewhat predictably, the mean preferred reverberation level is higher in a less reverberant environment, but the distributions of reverberation level preference are shown to be narrower for the more reflective mixing environment. The time it takes for subjects to reach a decision is similar in both environments, but the reflective environment shows a longer period of adaptation at the beginning of each trial set.

11:20

2aAAa8. Adapting, enhancing, and fictionalizing room acoustics. Ted Ohl (acouStaCorp, 701 E. 132 St., Bronx, NY 10454, tedohl@pdoinc.com) and Niels Adelman-Larsen (FlexAcoustics, Kgs Lyngby, Denmark)

The need for adjustable acoustics extends far beyond the Performing Arts Center or world class concert hall. The small scale of the existing market for adjustable acoustics impedes the development of products economical enough to expand the market. Expand the market by: Raising awareness of the options available for adjusting the acoustics in their room will expand the market. User and designer feedback regarding desirable features also will contribute. Broadening the concept of variable acoustics to include noise control creates opportunities in the industrial sector. Focusing on cost saving efficiencies in design and production of products making the cost of adjustable acoustic solutions accessible to more clients; developing products suitable for retrofit as well as new construction Product Development: identify critical and desirable features in collaboration with users and designers. Develop product performance data to enable acousticians to incorporate variable acoustics into their designs and predict performance for clients. Incorporate other design team members' input to anticipate conflicts such as space use, look, cost, and other systems. Available Products Available product types will be quickly reviewed. *In-situ* test data from existing projects will be presented. More innovation to reach a broader market will be encouraged

11:40

2aAAa9. Electric guitar—A blank canvas for timbre and tone. Alex Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz), Agnieszka Roginska, Justin Matthew, and Jim Anderson (New York Univ., New York, NY)

The electric guitar is a complex mechanical, electrical, and acoustic system, invented less than a century ago. While more traditional instruments such as voices and violins, trumpets and tympani, piano and piccolo might possess innate traits that most listeners easily identify, the electric guitar is a sound synthesizer capable of a vast range of sounds. The guitar, the amp, and the recording techniques used enable the performer and the engineer to define and refine elements of tone, almost without limit. Electric guitar has no single reference tone quality, but instead invites, and even inspires performers and recordists to create new sounds and explore alternative timbres as desired.

TUESDAY MORNING, 4 JUNE 2013 — 513DEF, 8:55 A.M. TO 12:00 NOON

Session 2aAAb

Architectural Acoustics, Noise, and Physical Acoustics: New Materials for Architectural Acoustics

Matthew V. Golden, Chair
Scantek, 6430c Dobbin Rd., Columbia, MD 21045

Chair's Introduction—8:55

Invited Papers

9:00

2aAAb1. National Gypsum Company's acoustically enhanced gypsum board—SoundBreak XP. Stephen A. Cusa (Sales/Marketing, National Gypsum Co., 2001 Rexford Rd., Charlotte, NC 28211, stevec@nationalgypsum.com)

Gold Bond® BRAND SoundBreak XP Gypsum Board is an acoustically enhanced gypsum board used in the construction of high STC wall assemblies. This innovative gypsum board allows for construction of high STC wall assemblies that are thinner, cost effective, and more reliable than traditional methods. In addition to achieving a higher performance wall relative to acoustics, SoundBreak XP helps to achieve a higher performance wall in general. SoundBreak XP comes standard with mold and mildew resistance and is manufactured with an abrasion resistant paper. Both the mold resistance and the abrasion resistance achieve the best possible scores relative to the standard ASTM test methods. SoundBreak XP installs and finishes like standard gypsum board.

9:20

2aAAb2. Increased scattering efficiency through cell optimization in acoustic diffusers. Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com)

This presentation will provide test data, experiments, and analysis in showing the interaction of absorption and diffusion characteristics. It will be shown that absorption, either additive or design inherent, has an adverse effect on the efficiency of the acoustic diffusers. The effects of low frequency interference on various cell structures of quadratic residue diffusers will be highlighted as a prime example of this theory. The inclusion of additive absorption, i.e., fabric, fiberglass, and other materials, further complicates the matter of diffuser efficiency. While design characteristics of some diffusers, such as prime root and quadratic residue diffusers, can be used effectively for their absorption coefficients, overall these characteristics detract from the efficiency of the diffusion coefficients. Product designs utilizing optimized cell designs to inhibit absorption in diffusers will be shown. It will be shown that this cell optimization dramatically increases the efficiency of the diffuser. This optimization includes both higher scattering coefficients and smoother frequency responses. Comparative analysis will be provided with traditional designs and newer optimized designs.

9:40

2aAAb3. From felt to fungus: New materials and applications—Focus on innovation and exploration. Dawn Schuette and Scott Pfeiffer (Threshold Acoust, LLC, 53 W Jackson Blvd., Ste. 815, Chicago, IL IL, dschuette@thresholdacoustics.com)

A two-part presentation of new materials for use in architectural acoustics. This presentation emphasizes new materials, both commercially available and those pressed into use for acoustic benefit. The companion session is presented in “Cultivating the Sustainable in Architectural Acoustics.” Contemporary architectural design often seeks to push the standard of construction, resulting in the need to explore new acoustic solutions to new architectural challenges. Innovative use of commercially available materials or exploration into the development of new materials or modifications of known materials is required to find the best solutions both acoustically and architecturally. Use of acoustical products and non-acoustical products for acoustical benefit are reviewed through case studies.

Contributed Papers

10:00

2aAAb4. Simulation of normal incidence sound absorption coefficients of perforated panels with/without glass wool by transmission line parameters in a two- port network. Takayoshi Nakai and Kota Yoshida (Dept. of Elec. & Electron. Eng., Faculty of Eng., Shizuoka Univ., 3-5-1 Johoku, Naka-ku, Hamamatsu 432-8561, Japan, tdtnaka@ipc.shizuoka.ac.jp)

This paper describes simulation of normal incidence sound absorption coefficients of perforated panels by ABCD-parameters in a two-port network. Maa and Sakagami have investigated micro perforated panels, MPP. But their theories can treat only near 1% perforation rates of perforated panels with back cavities. If sound propagates as a plane wave, sound propagation can be represented as ABCD-parameters in a two-port network. Perforated panels, back cavities, and glass wool absorption materials are represented as matrix of ABCD-parameters, respectively. ABCD-parameters of a perforated panel with a back cavity are calculated as multiplication of their matrices. An input impedance can be calculated from the calculated ABCD-parameters. A normal incident absorption coefficient is calculated from the input impedance. Holes of the perforated panels have losses of viscous friction and thermal conduction at their walls. Simulations are done in the condition of 0.25 to 5 mm diameters of holes, 0.25% to 25% perforation rates, 0.5 to 5 mm thickness of the perforated panels with back cavities in which there are or are not glass wool absorption materials. The results of these simulations are good agreements with the results of our measurements by transfer function method.

10:20–10:40 Break

10:40

2aAAb5. Sound absorption and transmission through flexible micro-perforated structures. Cédric Maury (Centre National de la Recherche Scientifique (CNRS), Equipe Sons - Laboratoire de Mécanique et d’Acoustique (UPR CNRS 7051), Laboratoire de Mécanique et d’Acoustique, 31, chemin Joseph Aiguier, Marseille cedex 20 13402, France, cedric.maury@centrale-marseille.fr), Teresa Bravo (Consejo Superior de Investigaciones Cientificas (CSIC), Centro de Acustica Aplicada y Evaluacion No Destructiva (CAEND), Madrid, Spain), and Cédric Pinhède (Centre National de la Recherche Scientifique (CNRS), Equipe Sons - Laboratoire de Mécanique et d’Acoustique (UPR CNRS 7051), Marseille, France)

This work presents a theoretical and experimental study on sound absorption and transmission through structures made up of single and multiple-layer micro-perforated panels (MPPs). As they contribute to improve acoustical comfort, speech intelligibility and comply with lightweight, transparency and fiberless requirements, increasing applications are found in architectural acoustics or in the aeronautic and surface transport industries. A fully coupled modal approach is proposed to calculate the absorption coefficient and the transmission loss of finite-sized layouts made up of multiple flexible MPPs separated by air gaps. Validation results are obtained for single and double-layer thin MPPs against the transfer matrix approach and against measurements performed in a standing wave tube and in an anechoic chamber. Analytical approximations are derived from coupled-mode analysis for the Helmholtz-type and structural resonance frequencies of a single layer MPP structure together with relationships on the air-frame relative velocity over the MPP surface at these resonances. Principled guidelines are provided for enhancing both the sound absorption and transmission properties of multiple-layer MPP structures through suitable setting of the design parameters.

11:00

2aAAb6. Analysis of sound absorption behavior of polyester fiber material faced with microperforated panels. Davide Borelli, Corrado Schenone, and Ilaria Pittaluga (DIME - Sez. TEC, Università degli Studi di Genova, Via all’Opera Pia 15/A, Genova, GE 16145, Italy, corrado.schenone@unige.it)

Perforated facings used in lined ducts or sound absorbing panels can have various purposes: protecting the porous sound absorbing material from dust or grazing flow, acting as a rigid support for the porous material, or also affecting the behavior of the “backing” material, modifying this way the acoustical performance of the porous layer. This paper describes the effect of perforated facings on sound absorption characteristics of samples made by polyester fiber, experimentally investigated in accordance with ASTM C384 04 standard by means of two Kundt’s tubes with different diameters. The polyester (PET) fiber material had bulk density of 30 kg/m^3 and melting point at 260°C. The analysis was performed for a sample thickness equal to 100 mm. The samples were faced by means of different metal plates perforated with circular holes. The holes diameter was equal to 2 mm for all facings, while the percent open area was varied from 4.9% to 30%. The microperforated panels (MPPs) were positioned in adherence of the PET fiber material or at a distance of 2, 4, and 6 mm. The different behaviors due to the multiple combinations of percent open area and distance from the sample have been then analyzed and discussed.

11:20

2aAAb7. Structures of resonators in a cavity for improving a sound insulation of a thin double-leaf panel. Shinsuke Nakanishi (Faculty of Eng., Hiroshima Int.Univ., 5-1-1, Hiro-koshingai, Kure 737-0112, Japan, s-nakani@it.hirokoku-u.ac.jp)

The specific acoustic problem of a double-leaf panel is a less sound insulation caused by a mass-air-mass resonance. For improving the sound insulation, many studies have suggested Helmholtz resonators in the cavity, which are tuned at the resonant frequency. They have measured and analyzed this problem of double-walls spaced with 100 mm thickness of air gap. They have suggested that the resonators improve the sound insulation to the resonant transmission, and discussed its optimization for a gain by the resonators and structures set in the cavity. But it is unclear that those results can apply to sound insulation by a double grassing with 5 mm thickness of air gap, which is often seen even as a thermal insulated window, and whose air gap is quite thinner than that of the walls. Then, this study measured effects of various resonators in the cavity for improving the sound insulation of thin double-leaf panels, and discusses effects of structures and perforation ratio to the sound insulation. Moreover, for analyzing the effects of resonators, this study discusses measured results with theoretical studies of sound absorption models for resonators.

11:40

2aAAb8. Rectangular slot perforations to improve the sound absorption of walls. Umberto Berardi (DICAR, Politecnico di Bari, via Orabona 4, Bari 70125, Italy, u.berardi@poliba.it)

This paper focuses on theories to predict the sound absorption properties of a surface with rectangular perforations. First, the paper gives a synthetic review of available knowledge about theoretical and experimental works about rectangular slot perforations. A comparison of these models is also reported. Then, the adaptability of these models to 3-d structures of finite size, such as walls, is discussed. Later the paper presents a model based on the transfer matrix method, which takes into account surface, hole and cavity impedances. In particular, the surface impedance is obtained by the sum of a resistance term, a mass term, and characteristic impedance. The numerical model is hence adapted to the case of stone walls. Experimental tests have been performed in a reverberant chamber to measure the sound absorption of several configurations of slotted walls. The bricks of these walls have been mounted creating vertical slots of different dimensions, which, combined with different air gaps, allowed the wall to work as an assembly of Helmholtz resonant absorbers. Comparisons with theoretical models are carried out to explain the different mechanisms of sound absorption. Finally, this paper aims to establish practical roles to develop optimized noise control solutions through rectangular slot perforated walls.

TUESDAY MORNING, 4 JUNE 2013 — 510B, 9:00 A.M. TO 12:00 NOON

Session 2aAB

Animal Bioacoustics and Signal Processing in Acoustics: Conditioning, Segmentation, and Feature Extraction in Bioacoustic Signals

David K. Mellinger, Chair

Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Science Dr., Newport, OR 97365

Invited Papers

9:00

2aAB1. A supervised approach for segmentation of bioacoustics audio recordings. Forrest Briggs, Raviv Raich, and Xiaoli Fern (School of EECS, Oregon State University, 1148 Kelley Engineering Center, Corvallis, OR 97331-5501, raich@eecs.oregonstate.edu)

Segmentation is one of the most important tasks in preprocessing audio recordings for species recognition. For examples, bird songs or calls often consist of distinct short utterances. Correctly segmenting such utterances is an essential step in the analysis of bird songs or calls. Energy based time-domain segmentation is commonly used and can be fairly effective when dealing with high signal-to-noise ratio recordings of a single individual. We consider the scenario in which omnidirectional microphone are deployed for round-the-clock in-situ species monitoring. In such scenario, recordings may suffer from two problems: (i) low signal-to-noise ratio and (ii) simultaneous vocalizations of multiple individuals. We are interested in segmentation of such recordings. We propose a framework for supervised time-frequency segmentation of audio recordings. Using manually labeled spectrograms, a classifier is trained to separate vocalization from the background noise in the two dimensional time-frequency domain.

9:20

2aAB2. Joint classification of whistles and echolocation clicks from odontocetes. Yang Lu, David K. Mellinger, Holger Klinck (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Science Dr., Newport, OR 97365, lu.yang@noaa.gov), and Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., Newport, Oregon)

We propose to use acoustic features of both clicks and whistles to classify odontocete sounds to species. The species studied are Cuvier's beaked whales (*Ziphius cavirostris*), bottlenose dolphin (*Tursiops truncatus*), melon-headed whale (*Peponocephala electra*), and short- and long-beaked common dolphin (*Delphinus delphis* and *D. capensis*). An energy-based detector is used for echolocation click detection, and Roch's Silbido algorithm is used for whistle detection. Detected whistles are characterized by maximum and minimum frequencies, duration, slope, spectral maxima, spectral gaps, number and frequency of inflection points, number of "loop" repetitions, and other acoustic characteristics. Detected clicks are characterized by cepstral characteristics, as well as by a set of noise-resistant statistics. Clicks that occur within a certain time neighborhood of a whistle have the corresponding feature vectors merged to produce the input to the classification system. Random forest and Gaussian mixture model classifiers are tested on the resulting features and performance is characterized. [Funding from ONR.]

9:40

2aAB3. Sparse coding for scaled bioacoustics: From Humpback whale songs evolution to forest soundscape analyses. Herve Glotin (CNRS LSIS, Univ Sud Toulon, Inst. Univ. de France, USTV, avenue Université, BP20132, La Garde 83957, France, glotin@univ-tln.fr), Jérôme Sueur (MNHN, CNRS UMR, Paris, France), Thierry Artières (CNRS LIP6, UPMC, Paris, France), Olivier Adam (CNRS CNPS, Univ. Paris Sud, Paris, France), and Joseph Razik (CNRS LSIS, Univ Sud Toulon, Inst. Univ. de France, Paris, France)

The bioacoustic event indexing has to be scaled in space (oceans and large forests, multiple sensors), and in species number (thousand). We discuss why time-frequency featuring is inefficient compared to the sparse coding (SC) for soundscape analysis. SC is based on the principle that an optimal code should contain enough information to reconstruct the input near regions of high data density, and should not contain enough information to reconstruct inputs in regions of low data density. It has been shown that SC methods can be real-time. We illustrate with an application to humpack whale songs to determine stable components versus evolving ones across season and years. By sparsing at different time scale, the results show that the shortest humpack acoustic codes are the most stable (occurring with similar structure across two consecutive years). Another illustration is given on forest soundscape analysis, where we show that time-frequency atomes allow an easier analysis of forest sound organization, without initial classification of the events. These researches are developed within the interdisciplinary CNRS project "Scale Acoustic Biodiversity," with Univ. of Toulon, Paris Natural History Museum, and Paris 6, consisting into efficient processes for conditioning and representing relevant bioacoustic. Information, with examples at sabiod.univ-tln.fr.

10:00

2aAB4. Bat species identification from zero crossing and full spectrum echolocation calls using Hidden Markov Models, Fisher scores, unsupervised clustering and balanced winnow pairwise classifiers. Ian Agranat (Wildlife Acoust., Inc., 970 Sudbury Rd., Concord, MA 01742-4939, ian@wildlifeacoustics.com)

A new classification technique for the identification of bats to species from their echolocation calls is presented. Three different datasets are compiled and split in half for training and testing classifiers. Combined, the data include 9014 files (bat passes) with 226,432 candidate calls (pulses or extraneous noise) representing 22 different species of bats found in North America and the United Kingdom. Some files are of high quality consisting of hand-selected search phase calls of tagged free flying bats while others are from a variety of field conditions including both active (attended) and passive (unattended) recordings made with a variety of zero crossing and full spectrum recording equipment from multiple vendors. Average correct classification rates for the three datasets on test data are 100.0, 97.9, and 88.8%, respectively, with an average of 92.5, 72.2, and 39.9% of all files identified to species. Most importantly, classifiers in the third dataset for two species of U.S. endangered bats, *Myotis sodalis* (MYSO) and *Myotis grisescens* (MYGR) have a correct classification rate of 100 and 98.6%, respectively, and identify 67.4% and 93.8% of all files to species suggesting that the classifiers are well suited to the accurate detection of these endangered bats.

10:20–10:40 Break

10:40

2aAB5. Conditioning for marine bioacoustic signal detection and classification. David K. Mellinger (Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Science Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu)

Marine acoustic signals are characterized by certain types of noise and interference. Conditioning methods applied to spectrograms can be used to reduce or even remove these sounds, making bioacoustic signals more evident and simplifying the tasks of detection and classification. One family of methods is for making a long-term estimate of noise at each frequency and subtracting this estimate from the spectrogram; this has the beneficial effects of whitening the noise spectrum and removing relatively stationary noise sources such as vessel sound, but has the detrimental effect that relative spectrum levels—important in echolocation click classification—are altered. Another method estimates the spectrum in narrow bands at each time step and subtracts this estimate from the corresponding spectrogram frame; this method is useful for tonal sound detection and classification in that it removes short-duration clicks from snapping shrimp and echolocating animals. Other methods for removing other, more rare types of noise are presented as well. Examples and performance characterization of these methods are presented. [Funding from ONR and N45.]

Contributed Papers

11:00

2aAB6. Robustness of perceptual features used for automatic aural classification to propagation effects. Carolyn M. Binder, Paul C. Hines, Sean P. Pecknold, and Jeff Scrutton (Defence R&D Canada, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, carolyn.binder@drdc-rddc.gc.ca)

Previous effort has shown that a prototype aural classifier developed at Defence R&D Canada can be used to reduce false alarm rates and successfully discriminate cetacean vocalizations from several species. The aural classifier achieves accurate results by using perceptual signal features that model the features employed by the human auditory system. Current work focuses on determining the robustness of the perceptual features to propagation effects for two of the cetacean species studied previously—bowhead and humpback whales. To this end, classification results are compared for the original vocalizations to classification results obtained after the vocalizations were re-transmitted underwater over ranges of 2 to 10 km. Additional insight into the propagation effects is gained from transmission of synthetic bowhead and humpback vocalizations, designed to have features similar to the most important aural features for classification of bowhead and humpback vocalizations. Each perceptual feature is examined individually to determine its robustness to propagation effects compared to the other aural features. To gain further understanding of propagation effects on the features, preliminary propagation modeling results are presented in addition to experimental data.

11:20

2aAB7. Odontocete click train deinterleaving using a single hydrophone and rhythm analysis. Olivier Le Bot (STIC/AP, ENSTA Bretagne, 2, rue Francois Verny, Brest 29200, France, olivier.le_bot@ensta-bretagne.fr), Julien Bonnel (LabSTICC/TOM, ENSTA Bretagne, Brest, France), Jérôme I. Mars (Image and Signal, GIPSA-Lab, Saint Martin d'Hère, France), and Cédric Gervaise (Image and Signal, GIPSA-Lab, Saint Martin d'Hère, France)

Most odontocetes live in pods of several individuals, resulting in an overlapping of click trains recorded by passive acoustic monitoring systems. Localization algorithms and click classifiers are usually used for train separation. However, their performances fall down if individuals are too close to each other or if acoustical parameters vary greatly from click to click, respectively. Assuming odontocete clicks follow rhythmic patterns, we propose to use a rhythm analysis to separate mixed click trains from a single hydrophone. The proposed algorithm is based only on inter-click-intervals (ICI) to cluster clicks into trains. It uses information given by complex-valued autocorrelation to compute a histogram, which will exhibit peaks at ICIs corresponding to interleaved trains. By this technique, subharmonics corresponding to multiples of ICIs are automatically suppressed. The algorithm is then extended by a time-period analysis leading to a time-varying ICI spectrum. A threshold can be applied on this spectrum to detect the different interleaved trains. The final result is a binary time-ICI map on which trains can be fully and easily distinguished and extracted. We validate it on simulated and experimental data, and we show that the algorithm is particularly suitable as a preprocessing tool prior to localization and classification schemes.

11:40

2aAB8. Complexity index and proportional variability to study dolphin whistles. Carmen Bazúa Durán, E. Julieta Sarmiento Ponce (Facultad de Ciencias, UNAM, Circuito Exterior s/n, Ciudad Universitaria, México, D.F. 04510, Mexico, bazua@unam.mx), Brenda P. González Leal (Universidad del Mar, Oaxaca, Mexico), and Camila Rodríguez Bohorquez (Facultad de Ciencias, Universidad de los Andes, Bogotá, Colombia)

Dolphin whistles are emitted especially during social interactions and feeding activities involving group cohesion, individual recognition, and recruitment. This paper presents a new methodology to describe and compare the whistle repertoire of dolphins. It consists on first extracting the whistle contour using MATLAB BELUGA, then classifying whistles into whistle types using MATLAB ArtWARP, next classifying whistle types into four general categories (high complexity, low complexity, linear long, and linear short), and finally computing a complexity index and a proportional variability of the whistle repertoire. The method was tested with whistles from captive and wild bottlenose dolphins, *Tursiops truncatus*, and from wild Guyana dolphins, *Sotalia guianensis*. Results obtained showed that this very simple method is useful to describe the whistle repertoire and to compare it according to the general behavioral state of dolphins, and between species. It is necessary to implement new methodologies like this one to better understand how dolphins are using whistles, since acoustic communication is the most important sense in dolphin species. This is specially important in areas where dolphins are exposed to humans, and where underwater visibility is limited, like Laguna de Términos, a Marine Protected Area in Mexico. [Work supported by PAPIIT-UNAM.]

TUESDAY MORNING, 4 JUNE 2013 — 510D, 8:55 A.M. TO 12:00 NOON

Session 2aAO

Acoustical Oceanography: Seismic Oceanography

Warren Wood, Cochair
Geology and Geophysics, Naval Res. Lab., 1005 Balch Blvd., Stennis Space Ctr., MS 39529-5004

Berta Biescas Gorriz, Cochair
Oceanography, Dalhousie Univ., 1355 Oxford St., Halifax, NS B3H3k4, Canada

Chair's Introduction—8:55

Contributed Papers

9:00

2aAO1. Uncertainty of transmission loss due to small scale fluctuations of sound speed in two environments. Josette P. Fabre (Acoustics, Naval Res. Lab., 1005 Balch Blvd., Stennis Space Ctr., MS 39529, josie.fabre@nrlssc.navy.mil) and Warren Wood (Geology and Geophysics, Naval Res. Lab., Stennis Space Ctr., MS)

Seismic oceanography techniques reveal detection of small scale variations in sound speed not detectable via conventional oceanographic means, i.e., frequent XBT or CTD casts). Due to computational and practical limitations, such small scale spatial and temporal detail that exists in a real ocean environment is not typically included in acoustic ocean models. However, such measurements can provide insight to the small scale variability (uncertainty) that exists in the ocean but is not predicted by mesoscale ocean models. We show acoustic predictions made with the Range Dependent Acoustic Model (RAM) using measured seismic oceanography and CTD data at two locations in significantly different environments. Additionally, the CTD measurements are smoothed to a resolution comparable to that provided by a dynamic ocean model and acoustic predictions are computed. The Uncertainty Band (UBAND) algorithm (UBAND) [Zingarelli,"A mode-based technique for estimating uncertainty in range-averaged transmission loss results from underwater acoustic calculations," J. Acoust. Soc. Am. **124**(4) (2008)] is applied to the smoothed oceanographic data using estimates of sound speed uncertainty calculated from the high resolution measurements. We find reasonable estimates of uncertainty due to the small scale oceanography that is not characterized by mesoscale ocean models.

9:20

2aAO2. Inversion of density in the ocean from seismic reflection data. Berta Biescas Gorriz, Barry Ruddick (Oceanography, Dalhousie Univ., 1355 Oxford St., Halifax, NS B3H3k4, Canada, berta.biescas@dal.ca), and Valenti Sallares (Marine Geology, Marine Science Inst. - CSIC, Barcelona, Barcelona, Spain)

Vertical stability of the fluid particles, mixing, and mesoscale motions in the ocean interior occur mostly along-isopycnals surfaces. Therefore, potential density profiles with high lateral resolution would provide important information about the fluid dynamic and the general circulation in the ocean.

Could we observe density changes from seismic data? Is seismic oceanography able to measure density with enough accuracy? How is the relation between seismic reflectors and isopycnals surfaces? We have inverted oceanic impedance from seismic data and then derived density and potential density surfaces from the oceanic impedance. Results of the inverted potential density have been compared with digitized seismic reflectors to show the relation between isopycnals and reflectors. We have also compare the seismic profiles of the GO Survey with the space-coincident CTDs and space and time-coincident XBTs to understand the nature of the reflectivity and its relation with the physical parameters of the ocean.

9:40

2aAO3. Seismic oceanography imaging of thermal intrusions in strong frontal regions. Jeffrey W. Book, Warren T. Wood (Naval Res. Lab., Oceanogr. Div., Stennis Space Ctr., MS 39529, jeff.book@nrlssc.navy.mil), Ana E. Rice (National Res. Council, Stennis Space Ctr., MS), Sandro Carniel (C.N.R - Inst. of Marine Sci., Venezia, Italy), Richard Hobbs (Univ. of Durham, Durham, United Kingdom), Isabelle Ansorge (Univ. of Cape Town, Cape Town, South Africa), Tim Fischer (GEOMAR Helmholtz Cte. for Ocean Res., Kiel, Germany), and Hartmut Prandke (ISW Wassermesstechnik, Fünfseen, Germany)

The Naval Research Laboratory and collaborating partners carried out two dedicated seismic oceanography field experiments in two very different strong frontal regions. Adriaseismic took seismic oceanography measurements at the confluence of North Adriatic Dense Water advected along the Western Adriatic Current and Modified Levantine Intermediate Water advected around the topographic rim of the Southern Adriatic basin. ARC12 took seismic oceanography measurements in and around the Agulhas Return Current as it curved northward past the Agulhas Plateau and interacted with a large anticyclone that collided with the current. Despite one study focused on coastal boundary currents and the other focused on a major Western Boundary Current extension, the complex horizontal structures seen through seismic imaging are tied to the processes of thermal intrusions and interleaving in both systems. Seismic Oceanography provides a unique capability of tracking the fine-scale horizontal extent of these intrusions. In both systems they occur primarily along isopycnals and are largely density compensating. The formation of these structures is associated with advective processes rather than diffusive processes, despite gradients favorable for double diffusion mixing. Results from these studies also show that submesoscale eddies are playing an important role in the formation of thermal intrusions near these strong fronts.

10:00

2aAO4. Exploring the shelf-slope dynamics in the Adriatic Sea using numerical models and seismic oceanography. Andrea Bergamasco, Francesco Falcieri (Oceanography, CNR-ISMAR, Castello 2737, Venice 30122, Italy, andrea.bergamasco@ismar.cnr.it), Jeff W. Book (Oceanography, Naval Res. Lab., Stennis, MS), Sandro Carniel (Oceanography, CNR-ISMAR, Venice, Italy), Warren W. Wood (Oceanography, Naval Res. Lab., Stennis, MS), Mauro Sclavo (Oceanography, CNR-ISMAR, Venice, Italy), and Richard W. Hobbs (Univ. of Durham, Durham, United Kingdom)

Dense shelf waters are formed and spread in the Adriatic Sea during winter periods, which dynamics are usually investigated by means of sea truth campaigns and modeling efforts. The former are either based on observational approaches (moored instruments, CTD, current meters, etc.) or on more innovative techniques, e.g., employing Seismic Oceanography (SO). Recent studies have shown that SO techniques can produce maps of vertical transects along the survey lines with horizontal and vertical resolution of, respectively, 10 and 100 m, suitable to explore the horizontal structures of BBL dynamics. Elaborating on these considerations, a novel approach combining the SO dataset collected during the ADRIASEISMIC cruise and high-resolution numerical model (ROMS) results was performed in two restricted areas of the Adriatic Sea: off the Gargano promontory and off the Bari shelf break. We present the first steps along the definition of a novel methodology. On one hand, SO can help to image the existing dynamical structures and their spatial/temporal resolution; on the other, the numerical model can quantify these acoustic snapshots in terms of temperature, salinity, and density, integrating the XBTs that are acquired during SO lines, and help identifying the nature of other processes (e.g., turbulence, internal waves, etc.).

10:20–10:40 Break

10:40

2aAO5. Mapping turbidity layers using a combination of high resolution seismic oceanographic and physical oceanographic data. Ekaterina A. Vsemirnova (Geospatial Res, Ltd., Durham Univ., Durham, County Durham, United Kingdom) and Richard W. Hobbs (Dept. of Earth Sci., Durham Univ., Durham, DH1 3LE, United Kingdom, r.w.hobbs@durham.ac.uk)

Synchronized seismic and oceanographic data were acquired during the Geophysical Oceanography (GO) project cruise in the Gulf of Cadiz in April–May 2007. The small volume (117 cu-in.) mini GI-gun seismic source used during the GO calibration experiment provided high resolution seismic data, which unveiled new features of the internal structure of the ocean. The seismic acquisition design gave a usable bandwidth of 50–250 Hz with a vertical resolution of 1.25 m, which is similar to that achieved by co-located CTD casts. We focus on the reflections observed on seismic data covering the moorings area. To test the hypothesis that measurable reflections can be generated by suspended sediment, we perform forward modeling of seismic response based on the temperature, salinity, and light attenuation measurements, available from CTD casts. Forward modeling based solely on temperature and salinity profiles show that thermohaline structure does not always explain reflections in water column, but they are consistent with light attenuation measurements.

11:00

2aAO6. Characterization of thermohaline staircases in the Tyrrhenian sea using stochastic heterogeneity mapping. Grant G. Buffett (Marine Geodynamics, GEOMAR Helmholz Ctr. for Ocean Res. Kiel, Gebdude Ostufer, Wischhofstr. 1-3, Geb. 8D/217, Kiel D-24148, Germany, gbuffett@geomar.de), Richard W. Hobbs (Dept. of Earth Sci., Durham Univ., Durham, United Kingdom), Ekaterina Vsemirnova (Dept. of Earth Sci., Durham Univ. Geospatial Res. Ltd., Durham, United Kingdom), Dirk Klaeschen (Marine Geodynamics, GEOMAR Helmholz Ctr. for Ocean Res. Kiel, Kiel, Schleswig-Holstein, Germany), Charles Hurich (Dept. of Earth Sci., Memorial Univ. of Newfoundland, St. John's, NF, Canada), C. Ranero (Barcelona Ctr. for Subsurface Imaging, Instituto de Ciencias del Mar, Barcelona, Spain), and V. Sallares (Inst. of Marine Sci., CMIMA, CSIC, Barcelona, Spain)

We apply Stochastic Heterogeneity Mapping based on the band-limited von Kármán power law function to stacked migrated seismic data of thermohaline staircases in the Tyrrhenian Sea. This process allows the estimation of stochastic parameters such as the Hurst number (a measure of surface roughness) and scale length. Thermohaline staircases are regular, well-defined step-like variations in vertical profiles of temperature and salinity. In the ocean, they are thought to arise from double diffusion processes driven by the large difference in the molecular diffusivities of heat and salt. They are thought to have an anomalously weak internal wave-induced turbulence, making them suitable for the estimation of a lower detection limit of turbulent dissipation. The Tyrrhenian Sea is a natural laboratory for the study of such staircases because, due to the internal basin's dynamic stability, steps as small as 10's of meters can be seen. Lower Hurst numbers represent a richer range of higher wavenumbers corresponding to a broader range of heterogeneity in reflection events. We interpret a broader range of heterogeneity as indicative of a greater degree of turbulence.

11:20

2aAO7. Mapping non-local turbulent breakdown of oceanic lee waves offshore Costa Rica through seismic oceanography. Will F. Fortin, W. Steven Holbrook (Geology and Geophysics, Univ. of Wyoming, 1000 E. University Ave., 3006, Laramie, WY 82071, wfortin@uwyo.edu), Ray Schmitt (Physical Oceanogr., Woods Hole Oceanogr. Inst., Woods Hole, MA), Jeff Book, and Scott Smith (Naval Res. Lab., Washington, DC)

Data analysis techniques in Seismic Oceanography are rapidly becoming more complex. Beyond first-order observation of oceanic structures, it is possible to extract quantifiable information about internal wave energies and turbulent dissipation rates. We use two co-located seismic surveys taken one day apart to estimate turbulent diffusivities of lee wave breakdown with emphasis on the mid-water turbulence generated. Through a horizontal wavenumber spectral analysis of our seismic images, we estimate turbulent dissipation rates throughout the cross-section. By integrating horizontal seismic slope spectra in the wavenumber domain over the turbulent subrange, we obtain relative turbulent energies across the survey. To resolve absolute turbulent diffusivities, we scale the relative measures to known absolute energies from tracked seismic

reflectors (isopycnals). The analysis section spans 22 km laterally and full ocean depth with a resolution on turbulent diffusivity of 10 m vertically by 400 m laterally. We focus on the region of elevated turbulent diffusivity caused by the breakdown of the lee wave above its generating site. We find the turbulent diffusivities related to the lee wave breakdown to be about five times greater than surrounding waters and 15 times greater than average open ocean diffusivities. We also see increased turbulent diffusivity around the rough bathymetry.

11:40

2aAO8. Current-eddy interaction in the Agulhas Return Current region from the seismic oceanography perspective. Ana E. Rice (National Res. Council, Bldg. 1009, Rm. B137, Stennis Space Ctr., MS 39529, anaerice@gmail.com), Jeffrey W. Book, Warren T. Wood (Naval Res. Lab., Stennis Space Ctr., MS), and Tim Fischer (GEOMAR Helmholtz Ctr. for Ocean Res., Kiel, Germany)

Interleaving in the Agulhas Return Current (ARC) frontal region is commonly manifested in the form of thermohaline intrusions, as sub-tropical and sub-polar water masses of similar density meet. In Jan./Feb. 2012, the Naval Research Laboratory and collaborators carried out a field experiment in which seismic and traditional hydrographic observations were acquired to examine frontal zone mixing processes. The high lateral resolution (10 m) of the seismic observations allowed fine-scale lateral tracking of thermal intrusions, which were corroborated with simultaneous XBT casts. Between seismic deployments both salinity and temperature data were acquired via CTD, underway-CTD and microstructure profiles. This study focuses on analyzing seismic reflection data in a particular E-W transect where the northward flowing ARC interacted with the southward flowing portion of a large anticyclonic eddy. Strong reflectors were most prominent at the edge of a hyperbolic zone formed between the eddy and ARC, where sub-polar waters interacted with waters of sub-tropical origin on either side. Reflectors were shallow within the hyperbolic zone and extended to 1200 m below the ARC. The nature of the observed reflectors will be determined from comparison of seismic reflection and derived $\partial T/\partial z$ fields, and XBT and TS profiles from the available hydrographic data.

TUESDAY MORNING, 4 JUNE 2013 519A, 9:00 A.M. TO 12:00 NOON

Session 2aBA

Biomedical Acoustics, Physical Acoustics, and Acoustical Oceanography: Bubbles Bubbles Everywhere I

Ronald A. Roy, Cochair
Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215

Thomas J. Matula, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., WA 98105-6698

Contributed Papers

9:00

2aBA1. A method for desalination and water remediation by hydrodynamic cavitation. Larry Crum (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, lac@apl.washington.edu), Michael Skinner, and Scott Zeilinger (Globe Protect, Inc., San Francisco, CA)

Water is becoming an increasingly valuable commodity, with population growth demanding more and more amounts of this limited resource. Increased efforts are directed toward recycling and remediation, as well as desalination of the large quantities of seawater available. Dr. Bertwin Langenecker was a pioneer in utilizing hydrodynamic cavitation in a variety of applications that would remove dissolved solids from water and other liquids. His combination of intense cavitation using a rotor-stator combination, as well as simultaneously adding an adsorbent, demonstrated impressive results in desalination and waste water remediation. In this presentation, a description will be given of Dr. Langenecker's technology as well as a sampling of some of his most impressive results. Speculations as to why this approach works as well as it does will be presented.

9:20

2aBA2. A new approach to ultrasonic cleaning. Tim Leighton (Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton, Hampshire SO17 1BJ, United Kingdom, tgl@soton.ac.uk), Peter R. Birkin, and Doug Offin (School of Chem., Univ. of Southampton, Southampton, Hampshire, United Kingdom)

Traditional ultrasonic cleaning baths are limited in that they cannot clean objects that are too large to fit in the bath, and cannot be taken to objects with complex geometries in order to "clean in place." Furthermore, the object to be cleaned sits in a "soup" of contaminated liquid, and while cavitation fields can be set up under test conditions, immersion of the object to be cleaned can significantly degrade the bath's performance by disrupting the sound field. An alternative technique, which does not use ultrasound is the commercial pressure- or -power washer, where high speed jets of water and cleaning agent are pumped onto a surface. Although these can "clean in place," they pump large volumes of water, and produce significant volumes of contaminated run-off and contaminated aerosol, both of which are hazards for secondary contamination of users and water supplies. The momentum of the water and pump requirements mean they are difficult to scale up. This paper presents a low volume flow technique for ultrasonic cleaning in place, benefits being that it operates with low flow rates (1–2 L/min), and there is no need to expend energy on heating the water.

9:40

2aBA3. The effect of surfactant shedding and gas diffusion on pressure wave propagation through an ultrasound contrast agent suspension. Jean-Pierre O'Brien, Nick Ovenden (Dept. of Mathematics, Univ. College London, UCL, London, United Kingdom, jean-pierre.o'brien@ucl.ac.uk), and Eleanor Stride (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Interest in microbubbles as agents for therapeutic and quantitative imaging applications in biomedical ultrasound has increased the need for their accurate modeling. However, effects such as gas diffusion, the properties of the shell, and changes in bubble behavior under repeated exposure to ultrasound pulses are still not well understood. A revised equation for microbubble motion is proposed that includes the effects of gas diffusion as well as a nonlinear surface tension, which depends on a non-constant surfactant surface concentration. This is incorporated into a nonlinear wave propagation

model to account for these additional time-dependent effects in the response of microbubble contrast agent populations. The results from the model indicate significant changes in both bubble behavior and the propagated pulse compared with those predicted by existing propagation models; and show better agreement with experimental data. Our analysis indicates that changes in bubble dynamics are dominated both by surfactant shedding on ultrasonic timescales and gas diffusion over longer timescales between pulses. Therefore, incorporating such time-dependent phenomena in ultrasound imaging algorithms should lead to better quantitative agreement with experiments.

10:00

2aBA4. Inertial cavitation at the nanoscale. James J. Kwan, Susan Graham, and Constantin Coussios (Inst. of Biomed. Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford, Oxfordshire OX3 7DQ, United Kingdom, james.kwan@eng.ox.ac.uk)

Our group has recently developed novel nano-sized drug carriers that spatially target a tumor and release their payload in the presence of ultrasound-induced inertial cavitation. To maximize drug release and distribution within the tumor, co-localization of the drug carrier and cavitation nuclei is necessary. We have recently demonstrated that rough-patterned silica nanoparticles can reduce inertial cavitation thresholds to clinically relevant levels, and will extravasate in tumors alongside the liposomes by virtue of their size. We now report on the underlying mechanisms that these nanoparticles, which are orders of magnitude smaller than the acoustic wavelength, can instigate inertial cavitation. The rough surface of the nanoparticle is modelled as a plane with a crevasse that traps a nanobubble. Using this model, we predict the motion of a gas bubble as it emerges from the cavity in response to the compressional and rarefactional ultrasonic pressures. We show that cavitation occurs when the nanobubble breaks free from the surface, growing unstably before collapsing during the compressional half cycle of the acoustic wave. Calculations show that a nanoscaled cavity greatly reduces the cavitation threshold across all frequencies and geometries studied. In addition, cavitation thresholds nonlinearly decrease with increasing cavity size.

10:20

2aBA5. Cavitation-induced streaming in shock wave lithotripsy. Yuri A. Pishchalnikov (Impulse Devices, Inc., 13366H Grass Valley Ave., Grass Valley, CA 95945, yurapish@gmail.com) and James A. McAteer (Dept. of Anatomy and Cell Biol., Indiana Univ. School of Med., Indianapolis, IN)

Cavitation generated by lithotripter shock waves (SWs) in non-degassed water was studied using a 60 frames-per-second camcorder—recording the migration of microbubbles over successive SWs. Lithotripter SWs were produced using a Dornier DoLi-50 electromagnetic lithotripter at 0.5 and 2 Hz pulse repetition frequency (PRF). Cavitation was affected by PRF and by the power level (PL) of the lithotripter. At slow PRF, such as shots fired many seconds apart, cavitation was relatively sparse and bubble clouds flowed in the direction of SW propagation. When PRF was increased, the bubble clouds generated by one SW were amplified by subsequent SWs. Cloud amplification was accompanied by an apparent change in the pattern of bubble migration. Whereas bubbles continued to enter the field of view from the prefocal side, the main bubble cloud remained near the focal point. This was due to a streaming of bubbles opposite to the direction of SW propagation. Increasing the PL grew the cavitation field and enhanced the flow of bubbles opposite to the direction of SW propagation. Stepping up the PL acted to push the broad cloud progressively pre-focally (toward the SW source), shifting the position of the plane at which the opposing directional bubble flows collided. [NIH DK43881.]

10:40

2aBA6. Bubbles trapped on the surface of kidney stones as a cause of the twinkling artifact in ultrasound imaging. Oleg Sapozhnikov (Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, oleg@acs366.phys.msu.ru), Wei Lu, Michael Bailey, Peter Kaczkowski, and Lawrence Crum (Appl. Phys. Lab., Ctr. for Industrial and Med. Ultrasound, Univ. of Washington, Seattle, WA)

A twinkling artifact (TA) associated with urinary calculi has been described as rapidly changing colors on Doppler ultrasound. The purpose of this study was to investigate the mechanism of TA. Doppler processing was performed on raw per channel radio-frequency data collected when imaging human kidney stones in degassed water. Suppression of twinkling by an ensemble of computer generated replicas of a single received signal demonstrated that the TA arises from variability among the acoustic signals and not from electronic signal processing. This variability was found to be random in nature, and its suppression by elevated static pressure, and its return when the pressure was released, suggests that the presence of surface bubbles on the stone is the mechanism that gives rise to the TA. Submicron size bubbles are often trapped in crevices on solid objects, but the presence of these bubbles *in vivo* is unexpected. To further check this mechanism under conditions identical to *in vivo*, stone-producing porcine kidneys were harvested en bloc with a ligated ureter and then placed into a pressure chamber and imaged at elevated atmospheric pressure. The result was similar to *in vitro*. [Work supported by NIH DK43881, DK092197, RFBR, and NSBRI through NASA NCC 9-58.]

11:00

2aBA7. The use of twinkling artifact of Doppler imaging to monitor cavitation in tissue during high intensity focused ultrasound therapy. Tatiana D. Khokhlova (School of Med., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, tanyak@apl.washington.edu), Tong Li (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Dept. of Acoust. Phys. Faculty, Moscow State Univ., Russian Federation and Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Joo Ha Hwang (School of Medicine, Univ. of Washington, Seattle, WA)

In high intensity focused ultrasound (HIFU) therapy, it is important to monitor the presence and activity of microbubbles in tissue during treatment. The current methods—passive cavitation detection (PCD) and B-mode imaging—have limited sensitivity, especially to small-size, non-violently collapsing microbubbles. Here, a new method for microbubble detection is proposed, based on "twinkling" artifact (TA) of Doppler imaging. TA occurs when color Doppler ultrasound is used to image hard objects in tissue (e.g., kidney stones) and is displayed as brightly colored spots. As demonstrated recently, TA can be explained by irregular scattering of the Doppler ensemble pulses from the fluctuating microbubbles trapped in crevices of the kidney stone. In this work, TA was used to detect cavitation in tissue and in polyacrylamide gel phantoms during pulsed 1 MHz HIFU exposures with different peak negative pressures (1.5–11 MPa). At each pressure level, the probability of cavitation occurrence was characterized using TA and the broadband signals recorded by PCD, aligned confocally with the HIFU transducer. The results indicate that TA is more sensitive to the onset of cavitation than conventional PCD detection, and allows for accurate spatial localization of the bubbles. [Work supported by RFBR and NIH (EB007643, 1K01EB015745, and R01CA154451).]

11:20

2aBA8. Rectified growth of histotripsy bubbles. Wayne Kreider (Univ. of Washington - Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Adam D. Maxwell (Univ. of Washington School of Medicine - Dept. of Urology, Seattle, WA and Univ. of Washington - Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Seattle, WA), Tatiana D. Khokhlova, Julianna C. Simon (Univ. of Washington - Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Seattle, WA), Vera A. Khokhlova, Oleg A. Sapozhnikov (Lomonosov Moscow State Univ.- Phys. Faculty, Moscow, Russian Federation and Univ. of Washington - Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Moscow, Russian Federation), and Michael R. Bailey (Univ. of Washington - Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Seattle, WA)

Histotripsy treatments use high-amplitude shock waves to fractionate tissue. Such treatments have been demonstrated using both cavitation bubbles excited with microsecond-long pulses and boiling bubbles excited for milliseconds. A common feature of both approaches is the need for bubble growth, where at 1 MHz cavitation bubbles reach maximum radii on the order of 100 microns and boiling bubbles grow to about 1 mm. To explore how histotripsy bubbles grow, a model of a single, spherical bubble that accounts for heat and mass transport was used to simulate the bubble dynamics. Results suggest that the asymmetry inherent in nonlinearly distorted waveforms can lead to rectified bubble growth, which is enhanced at elevated temperatures. Moreover, the rate of this growth is sensitive to the waveform shape, in particular the transition from the peak negative pressure to the shock front. Current efforts are focused on elucidating this behavior by obtaining an improved calibration of measured histotripsy waveforms with a fiber-optic hydrophone, using a nonlinear propagation model to assess the

impact on the focal waveform of higher harmonics present at the source's surface, and photographically observing bubble growth rates. [Work supported by NIH EB007643 and DK43881; NSBRI through NASA NCC 9-58.]

11:40

2aBA9. Ultrasonic atomization: A mechanism of tissue fractionation. Julianna C. Simon (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jcsimon@uw.edu), Oleg A. Sapozhnikov, Vera A. Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ. Moscow, Russian Federation and Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Yak-Nam Wang, Lawrence A. Crum, and Michael R. Bailey (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

High intensity focused ultrasound (HIFU) can be used to atomize liquid by creating a fountain on the surface exposed to air. The mechanism of atomization can be most accurately described by the cavitation-wave hypothesis wherein a combination of capillary waves excited on the liquid surface with cavitation beneath the surface produces a fine spray. Here, we show experimentally that a free tissue surface can also be atomized resulting in erosion of tissue from the surface. A 2-MHz spherically focused transducer operating at linearly predicted *in situ* intensities up to 14,000 W/cm^2 was focused at *ex vivo* bovine liver and *in vivo* porcine liver tissue surfaces without the capsule. The end result for both *in vivo* and *ex vivo* tissues was erosion from the surface. In bovine liver at the maximum intensity, the erosion volume reached 25.7 ± 10.9 mm^3 using 300 10-ms pulses repeated at 1 Hz. Jet velocities for all tissues tested here were on the order of 10 m/s. Besides providing a mechanism for how HIFU can mechanically disrupt tissue, atomization may also explain how tissue is fractionated in boiling histotripsy. [Work supported by NIH EB007643, NIH DK43881, and NSBRI through NASA NCC 9-58.]

TUESDAY MORNING, 4 JUNE 2013 — 512AE, 8:55 A.M. TO 12:00 NOON

Session 2aEA

Engineering Acoustics: Directional and Non-Directional Microelectromechanical Microphones

Gary W. Elko, Chair
mh Acoust. LLC, 25A Summit Ave., Summit, NJ 07901

Chair's Introduction—8:55

Invited Papers

9:00

2aEA1. A biologically inspired silicon differential microphone with active Q control and optical sensing. Ronald Miles (Dept. of Mech. Eng., SUNY Binghamton, Vestal, NY 13850, miles@binghamton.edu), Levent Degertekin (G.W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Weili Cui, Quang Su, Dorel Homentcovschi (Mech. Eng., SUNY Binghamton, Binghamton, NY), and Banser Fredrick (G.W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

A MEMS differential microphone is described in which the diaphragm design is inspired by the mechanics of directional hearing in the fly *Ormia ochracea*. The 1 mm by 3 mm diaphragm is designed to rotate about a central pivot in response to sound pressure gradients. The diaphragm is designed to have its dominant resonance mode within the audible frequency range and to have as little viscous damping as possible (to minimize the effects of thermal noise). The motion of the diaphragm is detected using an optical sensing scheme that includes a semiconductor laser (VCSEL), photodetectors, a mirror, and a diffraction grating. To minimize the adverse effects of the light damping on the response, an active feedback system is implemented to achieve active Q control. This uses the output of the optical detection scheme to drive the diaphragm through a capacitive actuator. The microphone and optoelectronics are packaged into an assembly that can be incorporated into a mock behind-the-ear hearing aid. The microphone is shown to achieve a noise floor that is approximately 17 dBA lower than what can be achieved using a pair of existing low noise hearing aid microphones to create a directional hearing aid.

9:20

2aEA2. Biomimetic flow sensors for environmental awareness. Gijs Krijnen, Harmen Droogendijk (MESA+ Res. Inst., Univ. of Twente, P.O. Box 217, Enschede 7500AE, Netherlands, gijs.krijnen@utwente.nl), Jerome Casas (IRBI, Université de Tours, Tours, France), and Ahmad Dagamseh (MESA+ Research Inst., Univ. of Twente, Enschede, Netherlands)

Crickets possess hairy organs attached to their abdomen, the so-called cerci. These cerci contain highly flow-sensitive mechanosensors that enable the crickets to monitor the flow-field around them and react to specific stimuli form the environment, e.g., air-movements generated by hunting spiders. Salient is the sensitivity of these sensors, which work at thermal noise threshold levels, and the large number of hairs which, together with the necessary neural processing, allows the cricket to use the cerci as a kind of "flow-camera." Biologists and engineers have been working together in the recent past to regenerate part of the outstanding sensing capabilities of crickets in manmade, bio-inspired flow-sensor arrays. Using micro-electromechanical systems (MEMS) technology, sensors are created that are sensitive and show a high degree of directivity. By analyzing the governing physics, the sensors have been optimized to the point that currently the electronic interfacing is the limiting factor. Nonlinear parametric effects are used to increase the range of applicability of the sensors. Stochastic resonance is investigated to further enhance sensing capabilities. Arrays of sensors, interfaced using frequency division multiplexing (FDM), have been demonstrated to enable the tracking of the movement of small spheres.

9:40

2aEA3. Small directional microelectromechanical systems microphone arrays. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com)

Directional microphone arrays that are physically small compared to the acoustic wavelength are of great interest for hand-held communication devices. Spatially directive microphones can reduce the impact of background acoustic noise without adding distortion to the signal. This talk will present some design topologies and requirements as well as a new physical design that could enable directional microphone responses while being small in size.

Contributed Papers

10:00

2aEA4. Leveraging microelectromechanical microphones inherent matching to reduce noise using multiple microphone elements. Wade Conklin (Knowles Electron., 1151 Maplewood Dr., Itasca, IL 60143, wade.conklin@knowles.com)

Signal-to-noise ratio (SNR) is a critical parameter in the adoption of small scale (~1 mm) microphones for use in hearing aids. As a result, electret microphones have dominated the market since their invention in the 1960's. Significant effort is being invested to increase the SNR of microelectromechanical (MEMs) microphones near that of electrets. This work covers the approach of using multiple microphone elements to increase SNR. It explores the theory, examines the dependence of the SNR improvement on the matching of the microphone elements, and compares measurements on a single element microphone versus a multiple element microphone. Finally, it examines why the MEMs fabrication process lends itself to this usage and compares the trade-offs in scaling elements versus scaling size.

10:20–10:40 Break

10:40

2aEA5. A novel two dimensional particle velocity sensor. Olti Pjetri, Remco J. Wiegerink, Theo S. Lammerink, and Gijs J. Krijnen (Transducers Sci. and Technol., Univ. of Twente, MESA+ Inst. for Nanotechnology, Drienerlolaan 5, Enschede 7522NB, Netherlands, o.pjetri@utwente.nl)

In this paper, we present a two wire, two-dimensional particle velocity sensor. The miniature sensor of size $1.0 \times 2.5 \times 0.525$ mm, consisting of only two crossed wires, shows excellent directional sensitivity in both directions, thus requiring no directivity calibration, and is relatively easy to fabricate. The sensor consists of two crossed beams of SiRN with a platinum layer on top. These beams are used both as heaters and sensors. Two currents with equal amplitude are injected in both terminals of one of the beams and are extracted from the terminals of the other beam. A particle velocity component in the direction of a beam will cause its temperature, and thus resistance, profile to change asymmetrically. This asymmetry in resistance will give rise to a voltage difference across that beam which is proportional to the particle velocity level. The sensor shows a frequency bandwidth between 20 Hz and 10 kHz.The two figures of eight are exactly perpendicular to each other as desired, which was difficult to obtain in earlier implementations using parallel beams. Furthermore, the structure consisting of two crossed wires increases the mechanical robustness of the beams resulting in fabrication yields of 94% as opposed to 70% in earlier implementations.

11:00

2aEA6. Characterization of directional microphones in an arbitrary sound field. Quang T. Su, Joshua H. Merlis, Daniel Antonelli, and Ronald N. Miles (Mech. Eng., Binghamton Univ., P.O. Box 6000, Binghamton, NY 13902-6000, qsu@binghamton.edu)

An acoustic characterization method for directional microphones is presented that does not require an anechoic chamber to provide a controlled plane wave sound field. Measurements of a directional microphone under test are performed in a nearly arbitrary sound field for several angles of sound incidence, and the corresponding sound pressure and pressure gradients in the vicinity of the test microphone are measured using an automated probe microphone scanning system. From these measurements, the total acoustic frequency response of the directional microphone can be decomposed into its sensitivities to sound pressure and pressure gradient using a least squares estimation technique. These component responses can then be combined to predict the directional response of the microphone to a plane wave sound field. This technique is demonstrated on a commercially available pressure gradient microphone, and also on a combination sound pressure-pressure gradient microphone. Comparisons with the plane wave responses measured in an anechoic environment show that the method gives accurate results down to 100 Hz.

11:20

2aEA7. Calibration of smartphone-based devices for noise exposure monitoring: Method, implementation, and uncertainties of measurement. Romain Dumoulin and Jeremie Voix (École de technologie supérieure, Université du Québec, 1100, rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, romain.dumoulin@ens.etsmtl.ca)

Standardized noise exposure campaigns have as their principle disadvantages the cost of instrumentation and the difficulties associated with practical deployment in the field. Our ongoing research evaluates the suitability of an alternate solution based on smartphone sensing: the occupational noise exposure and its associated measurement uncertainties are estimated from a spatio-temporal analysis of smartphones noise measurements and GPS data. This paper presents a diffuse field calibration method for such smartphone-based devices. The measurements methods and the calculation of expanded uncertainties for a large range of sound levels are detailed. The calibration corrections include a frequency response linearization and an A-weighted sound level correction, which is function of the C-A spectral balance of the sound pressure levels measured. To later ensure a realistic correction, these spectral balance values come from distribution of referenced industrial noise databases. An Android™ "app" has also been developed to measure the noise levels and to compute the calibration and corrections factors. Finally, an laboratory validation is conducted to evaluate, on a population of calibrated smartphone-based devices, the measurement errors associated with such devices as a function of microphone directivity, linearity, and frequency response.

11:40

2aEA8. Examination of acoustic mechanism for compact acoustic reproduction systems. Kosuke Sugihara (Faculty of Eng. Sci., Kansai Univ., Suita-Shi yamate-cho 3-3-35, Osaka-hu 556-8680, Japan, sukekiyo15@gmail.com), Masashi Nakamura (Fujitsu TEN, Hyogo, Japan), Yoshinobu Kajikawa, Yasuo Nomura (Faculty of Eng. Sci., Kansai Univ., Osaka, Japan), and Takashi Miyakura (HOSHIDEN, Osaka, Japan)

In this paper, we propose a method for analyzing compact acoustic reproduction systems (e.g., mobile phones) through acoustic equivalent circuits. Measured responses of compact acoustic reproduction systems cannot be represented accurately by the analysis based on the conventional acoustic theory. Acoustic engineers consequently are obliged to design compact acoustic reproduction systems by trial and error. Moreover, the sound quality of those systems is likely to deteriorate due to the difficulty of such an acoustic design. We therefore clarify the cause of the difference between the measured response and the analysis one calculated by the finite element method (FEM) analysis and consider the possibility of obtaining new acoustic theoretical formula based on the analysis results in order to make it easier for acoustic engineers to design compact acoustic reproduction systems.

Session 2aED

Education in Acoustics: Tools for Teaching Advanced Acoustics

David T. Bradley, Chair
Phys. + Astronomy, Vassar College, 124 Raymond Ave., #745, Poughkeepsie, NY 12604

Invited Papers

9:00

2aED1. Summer school for acoustics graduate students. Steven L. Garrett, Anthony A. Atchley (Grad. Prog. in Acoust., Appl. Res. Lab., Penn State, State College, PA 16804-0030, sxg185@psu.edu), Logan E. Hargrove (Retired, Reston, VA), Thomas J. Matula (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Joseph R. Gladden, and Henry E. Bass (Deceased) (National Ctr. for Physical Acoust., Univ. of Mississippi, University, MS)

In addition to subject mastery and the focused effort required to complete a thesis project, graduate students also need to develop a broad understanding of their field and cultivate a familiarity with the larger community of researchers and practitioners. The “summer school” format has been shown to enhance both subject-matter breadth and build community awareness in physical acoustics. Physical Acoustics Summer School (PASS) has been held in late-May, in even-numbered years, since 1992. The format for each day is usually two three-hour lectures followed by evening discussion groups to answer questions and explore extensions of the day’s lecture topics. One lecture session is typically dedicated to acoustics demonstrations. Attendance for the full week is required of all participants who also dine together three times each day. Venues are chosen to provide isolation that minimizes distraction and maximizes interactions among all participants. Typical enrollment has been 10 distinguished lecturers (including many Silver Medal winners in Physical Acoustics), 10 discussion leaders, and 30 graduate students. This format has been successfully extended to one other ASA Technical Committee: the marine bioacoustics community has held their summer school twice (SeaBASS). PASS has now been functioning long enough that former students have become lecturers.

9:20

2aED2. Experience teaching acoustics at the senior-undergraduate and first-year graduate levels. David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109-2133, drd@umich.edu)

Perhaps without appreciating it, college students are more fully equipped to understand and study acoustics than any other field of science. This assertion stems from the fact that most college students have two exquisite broadband receivers with impressive dynamic range (ears), and a matched multi-functional sound projector (voice). Given that nearly all college students have used their ears and voice for many years before arriving in an acoustics classroom, the advanced-acoustics instructor’s task is primarily to link theoretical results with the acoustic intuition that students already possess. Thus, a worthy pedagogical goal is to activate this submerged knowledge and connect it to mathematical results through practical examples, classroom demonstrations, and relevant homework. At the senior-level, useful demonstrations include the following: acoustic resonances of a cardboard tube, the dipole characteristics of small raw loudspeaker, directional reflection with a metal salad bowl, and sound volume changes as a loud speaker is lifted out of a cabinet. At the graduate level, useful homework assignments include boundary-element and finite-element calculations with commercial software that can be checked with established theory. In addition, worthwhile homework problems that attempt to provide sufficient reward for students who master the mathematical content have been developed for both classes.

9:40

2aED3. Combining theory and experiment to teach acoustic concepts. Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N181 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

A rigorous theoretical development is desirable to help students at both the undergraduate and graduate levels develop a deep understanding of acoustic phenomena. However, numerous students labor through the mathematics associated with the concepts without ever developing an understanding of how that translates over into the physical world. Many acoustic phenomena lend themselves to experimental demonstrations that can greatly aid students’ understanding of the physical concepts and help them connect the theoretical developments with what physically happens. These demonstrations also provide a means for introducing common issues associated with making acoustic measurements that can also be educational for students. As an example, this paper will focus on how we have developed concepts associated with vibrating strings in a class through both theoretical development and use of a relatively simple experimental apparatus. Students gain a much better understanding not only of modes associated with the string, but the relative accuracy of the underlying theory. In addition, basic signal analysis topics and measurement accuracy also surface in the process of making the measurements.

10:00

2aED4. Acoustic interference and diffraction experiments in the advanced laboratory class. Andrew Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

Many acoustic analogs to classical optics experiments can be performed with low-cost ultrasonic transducers. Mounts for the ultrasonic transducers have been designed to be produced with desktop 3D printing technology. The designs are open and available for use and modification. Examples of experiments that can be done with this system include single-slit diffraction, double-slit interference, and Lloyd’s mirror experiments. Although simple in appearance, these experiments provide a rich opportunity for students to explore acoustic phenomena in the advanced laboratory such as the radiation pattern of the transducer and rudimentary acoustic beamforming techniques. The lab activities for use in intermediate or advanced acoustics lab classes are included in a revised laboratory manual currently in development. The lab manual has experiments appropriate for both introductory and advanced acoustics labs covering a range of acoustics subfields.

10:20

2aED5. Using your ears: A novel way to teach acoustics. Lauren Ronsse, Dominique J. Chéenne, and Sarah Kaddatz (Dept. of Audio Arts and Acoust., Columbia College Chicago, 33 E. Congress Pkwy, Ste. 601, Chicago, IL 60605, lronsse@colum.edu)

Auditory simulations of physical phenomena pertaining to acoustics have been developed to enhance student learning and understanding of these conditions. The demonstrations range from simulations of fundamental concepts, such as reverberation, flutter echoes, reflections, and room modal effects, to more applied topics, such as sound transmission through barriers, mechanical system noise spectra, and varying absorption distribution in rooms. The simulations were generated by using auralization tools and processed recordings. The demonstrations may be utilized in the classroom to introduce new acoustical concepts by having students first listen to a simulation, then write and/or discuss what they hear, providing conjectures about the parameters that could create such acoustical conditions. The goal of the demonstrations is to encourage students to use their ears as part of a quantitative and qualitative assessments of acoustical phenomena.

10:40

2aED6. Creating interactive acoustics animations using Mathematica's Computable Document Format. Daniel A. Russell (Grad. Prog. in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

The computational and graphical prowess of Mathematica has long made it a powerful educational tool for creating effective animations of acoustic and vibration phenomena [Russell, J. Acoust. Soc. Am. **106**, 2197 (1999) and Sparrow and Russell, J. Acoust. Soc. Am. **103**, 2987 (1998)]. Once an animation has been created within Mathematica it is relatively easy to convert the animation to an animated GIF file for display on a website [Russell, J. Acoust. Soc. Am. **114**, 2308 (2003)]. However, such animations, while effective at conveying or illustrating complicated acoustic phenomena, are "static" in the sense that they are not interactive and a person viewing the animation cannot change parameters. Recently, Wolfram Research implemented a new Computable Document Format that allows interactive plots and animations to be inserted into webpages and electronic documents. A free CDF player from Wolfram allows viewers to interact with plots and animations by moving sliders to change values of parameters. This talk will demonstrate the process of creating a CDF animation for embedding in a webpage. Other, more complex, demonstrations will also be showcased do illustrate the potential capabilities of CDF as an educational tool.

11:00

2aED7. Teaching advanced undergraduate students principles of outdoor sound propagation using football game measurements. Kent L. Gee, Tracianne B. Neilsen, Alan T. Wall, and Nicholas J. Eyring (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

As part of a sound system evaluation at Brigham Young University's football stadium and to assist in planning for future system design, measurements were made before and during games by an upper-level undergraduate acoustics class. The measurement experience provided significant training opportunities for the students. Teams of students used sound level meters to make recordings at numerous locations both inside and outside the stadium. These measurements were then correlated with data from stationary microphones placed near the field. From the data, the predicted slow, A-weighted equivalent levels in and around the stadium were calculated relative to an assumed 90 dBA on the sideline. Straightforward outdoor sound propagation prediction methods involving geometric spreading, atmospheric absorption, barriers, etc. were successfully used to validate the measured data within 1-2 decibels at many locations, including a location in the foothills to the southeast of the stadium at a distance of approximately 3 km. The students appreciated the hands-on experiences gained by participation in the measurements and analysis.

11:20

2aED8. Spectrogram puzzles: A tool for teaching acoustic phonetics. Tessa Bent and Emily Garl (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

One of the most useful tools for the acoustic analysis of speech is the spectrogram. A spectrogram is a visual representation of speech which includes time, frequency, and amplitude information. To conduct appropriate and accurate acoustic-phonetic analyses, students must learn to identify important features of vowels and consonants on spectrograms. To help students learn to identify these features, the spectrogram puzzle exercise was developed. In this exercise, spectrograms of sentences are printed using a large-format printer and cut into phoneme sections. Students then arrange the segments into the appropriate order based on a provided sentence. Depending on students' level of knowledge and experience, task difficulty can be increased or decreased by: (1) providing phonetic transcription versus orthography, (2) including more or less easily identifiable consonants, (3) including citation-style speech versus conversational or disordered speech, and (4) having teams versus individual students complete the exercise. Through these modifications, this activity can be used with a wide range of students from beginning undergraduate to advanced graduate students. For all students, spectrogram puzzles provide a hands-on, interactive learning experience that can facilitate critical thinking, collaborative learning, and acquisition of knowledge about the representation of speech sounds on spectrograms.

11:40

2aED9. Mechanical model of the human ear. E. C. Everbach (Engineering, Swarthmore College, 500 College Ave., Swarthmore, PA 19081, ceverba1@swarthmore.edu)

Diagrams showing cutaway views of the human ear are all very well for teaching the mechanics of hearing, but a tabletop model the students can manipulate is even better. The author presents a mechanical model based upon the water-tube cochlea previously developed by Robert Keolian, but including the outer- and middle-ear components. The model allows phenomena such as the acoustic reflex, critical bands, and masking of higher-frequency by lower-frequency tones.

Session 2aMU

Musical Acoustics and Signal Processing in Acoustics: Aeroacoustics of Wind Instruments and Human Voice I

Shigeru Yoshikawa, Cochair
Grad. School of Design, Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan

Xavier Pelorson, Cochair
CNRS, 11 rue des mathematiques, Saint Martin d'Heres 38330, France

Invited Papers

9:00

2aMU1. Aeroacoustics of the panpipes. Roman Auvray, Benoît Fabre (LAM, d'Alembert, UPMC Univ Paris 06, CNRS UMR 7190, 11 rue de Lourmel, Paris 75015, France, auvray@lam.jussieu.fr), Felipe Meneses, Patricio de la Cuadra (CITA, Pontificia Universidad Catolica de Chile, Santiago, Chile), and Pierre-Yves Lagrée (FCIH, d'Alembert, UPMC Univ Paris 06, CNRS UMR 7190, Paris, France)

The generic term "flute-like instruments" includes a wide variety of instruments whose sound production is ensured by the coupling of an air jet with an acoustic resonator. Within the family, different kinds of resonator (for instance Helmholtz resonator, open-open, or open-closed tube), may be used with different kind of air supply systems such as the ones found in the recorder, the flue organ pipe, or the shakuhachi. It is common to extent the results obtained on one of the member of the family to the whole family. However, when an accurate description of the sound production mechanisms is required, small discrepancies may arise due to the wide variability in the geometries or in the air supply systems. Among other, a closed-end flute may have a different behavior than an open-open flute since the recirculation of air flow within the pipe may alter the hydrodynamics of the jet, and thus the auto-oscillation process. While most of the studies on flute-like instruments have focused on open pipes (organ pipes and recorder), the panpipes (a representative closed-end flute) has only received little attention. We present experimental data, including flow visualization and pressure signal measurement gathered on a closed pipe. A model of the flow in the pipe allows to interpret the data and compare the behavior of a closed pipe blown with a turbulent jet with that of an open pipe blown with a laminar jet.

9:20

2aMU2. Aerodynamical sounding mechanism in flue instruments: Acceleration unbalance between the jet vortex layers. Shigeru Yoshikawa (Grad. School of Design, Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, shig@design.kyushu-u.ac.jp)

According to particle image velocimetry (PIV) measurement applied to the sound production in organ flue pipes and flutes, the vortex shedding at the pipe edge proposed by Howe (1975) are not observed but the formation of the vortex layer is clearly observed along both sides of the jet flow. This has been confirmed in various sounding conditions with different blowing pressures and resulting pitches. The acceleration unbalance is generated from an incomplete cancelation of the aeroacoustical source term ($\omega\times\boldsymbol{U}$) between both sides of the jet, where $\boldsymbol{U}$ is the jet velocity and ω ($=\mathrm{rot}\boldsymbol{U}$) the vorticity. In addition, the vortex layer is essentially unstable because it is formed along the inflection point of the lateral jet-velocity profile. Therefore, the acceleration unbalance and inflection instability of the vortex layer activates the jet wavy motion to reinforce the inward or outward acoustic velocity $\boldsymbol{u}$ at the pipe mouth. Phase relations between acoustic quantities approve conventional acoustical models based on the volume-flow drive and momentum drive. Since $\omega\times\boldsymbol{U}$ can also activate the jet movement in edge-tone generation, the vortex-layer formation may be regarded as the fluid-dynamical mechanism common to the edge-tone generation and the pipe-tone generation.

9:40

2aMU3. Effective techniques and crucial problems of numerical study on flue instruments. Kin'ya Takahashi, Takuya Iwasaki, Takahiro Akamura (The Phys. Lab., Kyushu Inst. of Technol., Kawazu 680-4, Iizuka, Fukuoka 820-8502, Japan, takahasi@mse.kyutech.ac.jp), Yuki Nagao (Res. Information Sci. and Elec. Eng., Kyushu Univ., Fukuoka, Japan), Ken'ichiro Nakano (The Phys. Lab., Kyushu Inst. of Technol., Iizuka, Japan), Taizo Kobayashi, Toshiya Takami, Akira Nishida, and Mutsumi Aoyagi (Res. Inst. for Information Technol., Kyushu Univ., Fukuoka, Japan)

In the last several decades, there have been many important proposals for study on flue instruments from theoretical and experimental points of view. Analyses based on aerodynamic sound theory are crucial for understanding the sounding mechanism of flue instruments. According to the growth of computer power and the improvement of numerical schemes, numerical simulations based on fluid dynamics now become important tools for the study of aerodynamics sound. In this talk, we will discuss accuracy, efficiency, and reliability of numerical calculations with large-eddy-simulation (LES) of compressible flow and we will show to what extent LES can reproduce the fluid and acoustic behavior of flue instruments observed experimentally. Furthermore, we will consider how to calculate the important theoretical formulae of aerodynamics sound theory, e.g., Lighthill's quadrupole, Powell-Howe vortex sound source, Howe's formula that allows us to estimate the energy transfer between the acoustic field and the hydro-dynamic field. Actually, those quantities given by the theoretical formulae play an important role in the analyses of sounding mechanisms of flue instruments. M. Miyamoto *et al.*, "Numerical study on acoustic oscillations of 2D and 3D flue organ pipe like instruments with compressible LES," Acta Acustica (accepted for publication).

10:00

2aMU4. Experimental and numerical characterization of aerodynamic noise applied to moderate Reynolds number airflow. Yo Fujiso, Annemie Van Hirtum (GIPSA-Lab, Grenoble Univ., 11 rue des Mathematiques, Grenoble Campus, Saint Martin d'Heres 38402, France, yo.fujiso@gipsa-lab.grenoble-inp.fr), Kazunori Nozaki, and Shigeo Wada (Grad. School of Eng. Sci., Osaka Univ., Toyonaka-city, Japan)

In the study of aerodynamic noise, a tackling challenge is to understand the underlying aeroacoustic mechanisms leading to its generation. The current paper aims at contributing to the noise characterization by focusing on moderate Reynolds number ($100 \leq Re \leq 10000$) airflow circulating through a rectangular channel containing a trapezoidal obstacle near the outlet. The outcome of large Eddy simulation and acoustic experiments are compared for different experimental boundary conditions at inlet and outlet, and for different apertures below the obstacle.

10:20

2aMU5. Numerical analysis of the interaction between fluid flow and acoustic field at the mouth-opening of a flue instrument. Takahiro Akamura (The Phys. Lab., Kyushu Inst. of Technol., Kawazu 680-4, Iizuka, Fukuoka 820-8502, Japan, red0ta@yahoo.co.jp), Yuki Nagao (Res. Information Sci. and Elec. Eng., Kyushu Univ., Fukuoka, Japan), Takuya Iwasaki, Ken'ichiro Nakano, Kin'ya Takahashi (The Phys. Lab., Kyushu Inst. of Technol., Iizuka, Japan), Taizo Kobayashi, Toshimi Takami, Akira Nishida, and Mutsumi Aoyagi (Res. Inst. for Information Technol., Kyushu Univ., Fukuoka, Japan)

The fluid-sound interaction is the key to understanding the sounding mechanism of flue instruments. The formula introduced by Howe allows us to estimate the energy transfer between acoustic field and hydro-dynamic field. For calculation of Howe's formula, it is necessary to divide acoustic fields from fluid, but we do not have any established method to do it, yet. Recently, several authors developed approximate methods to evaluate Howe's formula and applied to experiments of cavity noise, flue instruments and so on. In this talk, we introduce a numerical method to calculate Howe's formula, which is similar to those above. Our model is a small flue-organ like instrument with an end-stop. We use compressible large-eddy simulation (LES), which is able to reproduce the fluid flow and acoustic field, simultaneously. First, fluid flow and acoustic oscillation excited in the pipe by a jet-injection from the flue are reproduced by LES. Next, an acoustic field is reproduced by LES without the jet-injection but with driving at the far end, pressure driving, particle velocity driving or oscillating wall driving (like a loudspeaker). Combining those results allows us to calculate Howe's formula and to estimate the fluid-sound interactions.

10:40

2aMU6. Numerical study on the function of tone holes of a recorder like instrument from the viewpoint of the aerodynamic sound theory. Takuya Iwasaki (Kyushu Inst. of Technol., Kawazu 680-4, Iizuka, Fukuoka 820-8502, Japan, gyjcjc56@gmail.com), Taizo Kobayashi (Res. Inst. for Information Technol., Kusyu Univ., Fukuoka, Fukuoka, Japan), Kin'ya Takahashi (Kyushu Inst. of Technol., Iizuka, Japan), Toshiya Takami, Akira Nishida, and Mutsumi Aoyagi (Res. Inst. for Information Technol., Kusyu Univ., Fukuoka, Japan)

We have investigated properties of tone holes of a small recorder like instrument by using compressible large-eddy simulation (LES), which reproduces fluid flow and acoustic field, simultaneously. When an acoustic flow of strong sound pressure passes through the junction between a tone hole and the main body of the bore, vortex shedding occurs, which induces unpleasant noises due to re-radiation of aerodynamics sound from the vortices. We have succeeded in reproducing this process and attempted to explain its mechanism from aerodynamic sound theory. We have also investigated how the position of the pad of a key above a tone hole, i.e., the distance between the pad and the top of the tone hole, affects the pitch of the excited wave in the pipe. Furthermore, we attempt to numerically reproduce the function of tone holes, namely change of notes. Opening and closing tone holes change the topology of bore geometry, which yields a moving boundary problem accompanied by topological change of numerical grids. We have developed a LES solver resolving the moving boundary problem and are planning to apply it to the problem of opening and closing tone holes.

11:00

2aMU7. Sound radiation of trained vocalizers. Braxton B. Boren and Agnieszka Roginska (Music and Audio Res. Lab., New York Univ., 30-91 Crescent St., 5B, Astoria, NY 11102, bbb259@nyu.edu)

Current research at NYU has focused on computational estimation of vocal loudness of George Whitefield, an Anglican preacher in the 18th century who reportedly spoke to crowds of 30,000 or more. After having established an overall level for his voice, we have begun investigating how his voice would have radiated spatially. Existing literature on the radiation of the spoken voice has focused extensively on the context of conversation in workspaces. These studies typically examine one soft, one normal, and one raised voice condition. Trained actors and orators, however, employ more methods of projection than are used in conversational speech and can achieve higher loudness as well. The radiation patterns from these types of communication have not been quantifiably studied yet. This paper investigates the radiation patterns of different methods of projection in trained vocalizers using relative intensity levels at 1 m from the front of the speaker. The results are compared to the existing data for conversational speech, and the implications to open-air oratory are discussed.

11:20

2aMU8. Vibrato rate variability in three professional singing styles: Opera, rock, and Brazilian country. Guilherme Pecoraro (Otolaryngology, Univ. of São Paulo Med. School, Rua Machado Bittencourt 361, Rua Marcelo Muller 1297, Sao Paulo, Sao Paulo 03223060, Brazil, guifono@hotmail.com), Daniella Curcio (Dept. of Morphology, Santa Casa School of Med. Sci. of São Paulo, São Paulo, SP, Brazil), and Mara Behlau (CEV, Ctr. for Voice Studies, São Paulo, SP, Brazil)

Vibrato is one of the most expressive aesthetic characteristics of singing voice. Indicative of good voice quality is typical of lyrical singing but it is also found in others styles of popular music. Acoustically, vibrato is defined as a long-term periodic modulation of the fundamental frequency. It occurs as a result of the laryngeal muscular system and is comprised of three main parameters: rate, extent, and amplitude variation. The main controversy refers to the physiological mechanism of vibrato production, specifically concerning its neurological conscious control, as well as the intra-subject variability of its acoustic parameters. In this study, we compare the characteristics related to vibrato rate (VR), assessing 423 emissions, from recorded samples, produced by 15 professional singers, publicly and artistically acclaimed in occidental culture, to represent three music styles: opera, rock and Brazilian country (sertanejo). We analyzed the samples through GRAM 5.01 and found that the VR was kept constant intra-subject, independently of the singing style. The mean values for the VR for opera and Brazilian country singers were higher than for rock singers. Effects of vocal training, kinship and aging on the vibrato rate, as well as technical skills to control it are objects of our future studies.

11:40

2aMU9. Phase-space visualization and real-time pitch tracking of vocal sounds. Dmitry Terez (SDI, 264 Eleanor, Cherry Hill, NJ 08003, dmitry-terez@gmail.com)

Novel techniques for sound waveform visualization and real-time pitch tracking are presented. The use of techniques is demonstrated on vocal sounds such as singing and speech. The visualization of a sound waveform in two- or three-dimensional reconstructed phase space obtained via signal time-delay embedding provides a compelling alternative to traditional spectral envelope, for example, for looking at a timbral structure of a sound. This phase-space visualization is performed continuously on a sample-by-sample basis and results in a moving trajectory on computer display—a living and breathing picture of voice in real time. In addition, pitch of voiced sounds is detected and displayed in real time using original pitch tracking algorithm. The algorithm has minimal possible latency which allows obtaining reliable fundamental frequency estimates in less than two cycles of a quasi-periodic waveform, before human auditory system can register pitch sensation. The techniques are efficiently implemented in real-time software using ASIO drivers. The software can be used as a tool for teaching singing or vocal intonation patterns as it provides immediate visual feedback on minute changes in pitch, timbre, or loudness.

TUESDAY MORNING, 4 JUNE 2013 511BE, 9:00 A.M. TO 12:00 NOON

Session 2aNSa

Noise: Transportation Noise

Kenneth Kaliski, Chair
RSG Inc., 55 Railroad Row, White River Junction, VT 05001

Contributed Papers

9:00

2aNSa1. Automatic classification of road vehicles considering their pass-by acoustic signature. Xavier Valero Gonzalez and Francesc Alías Pujol (La Salle-Universitat Ramon Llull, Quatre Camins, 30, Barcelona 08022, Spain, xvalero@salleurl.edu)

In order to assess the impact of environmental noise on a community, it is essential to accurately describe all the aspects and characteristics of the encountered noises. In this context, it would be of special interest to dispose of environmental noise monitoring stations capable of not only measuring the noise levels but also identifying the sources producing those levels. To offer such functionality, an algorithm to automatically recognize the noise sources is required. According to previous works, designing algorithms able to optimally distinguishing between road vehicle noise sources (i.e., light vehicles, heavy vehicles, and motorbikes) is a challenging issue. This paper proposes a recognition scheme that takes into account the perceived characteristics of road vehicles pass-by, which may be divided into different phases: approaching, passing and receding. By taking independent decisions for the pass-by phases, the proposed recognition scheme is able to improve the recognition of road traffic vehicles with respect to a traditional recognition scheme, specifically in 7% for light vehicles and in 4% for heavy vehicles.

9:20

2aNSa2. An acoustic based method for jointly estimating speed and wheelbase length of two-axle road vehicles as they pass by. Patrick Marmaroli, Xavier Falourd, and Lissek Hervé (Ecole Polytechnique Fédérale de Lausanne (EPFL), EPFL IEL STI LEMA, ELB 033, Station 11, Lausanne 1015, Switzerland, patrick.marmaroli@epfl.ch)

This paper focuses on acoustic road traffic monitoring and looks, more specifically, into the problem of speed and wheelbase length estimation of two-axle vehicles as they pass by. It is known that both front and rear axle trajectories may be dissociated using cross-correlation based methods in conjunction with a well designed two-element microphone array placed on the roadside. This is mainly due to the broadband nature of the tyre/road noise which makes two peaks appear, one per axle, in the correlation measurement when the vehicle is in the broadside direction. This paper aims at analyzing such a "bimodal'' observation in order to automatically extract the position, speed, and wheelbase length of passing-by vehicles. We propose to conduct this tracking problem using a particle filter that model the position-variant bimodal sound source nature of the vehicles. The theoretical developments presented in this paper are experimentally assessed through real *in-situ* measurements.

9:40

2aNSa3. The effect of the shadowing phenomenon on emergency vehicle siren noise. Peter J. D'Angela, Frankie Angione, Colin Novak, and Helen Ule (Univ. of Windsor, 2313 St. Clair, Windsor, ON N9E4S7, Canada, dangelp@uwindsor.ca)

It has been observed by some that emergency siren noise has gone unnoticed by drivers due to a shadowing phenomenon where the propagating siren noise is blocked from a receiver vehicle. The event is postulated to occur when a large vehicle is positioned between the emergency responder and a receiving vehicle. The sound of the siren is projected along the surface of the large vehicle and does not fall in time to reach the receiving vehicle. This situation is common at controlled intersections where the smaller vehicle is traveling perpendicular to the emergency vehicle but can also occur when the vehicles are in a common line on the road. The intent of this study is to investigate this phenomenon and quantify the resulting hindrance of a driver's ability to detect an approaching emergency vehicle. Included will be the use of the electrical "wail" siren and accompanying air horn commonly employed by Fire and Rescue Services. The outcome will be a determination of what frequency spectra are most affected by shadowing with an eventual goal to improve emergency siren design.

10:00

2aNSa4. Detectability study of warning signals in urban background noises: A first step for designing the sound of electric vehicles. Nicolas Misdariis, Anais Gruson, and Patrick Susini (UMR STMS Ircam-CNRS-UPMC, IRCAM, 1, place Igor Stravinsky, Paris F- 75004, France, misdarii@ircam.fr)

Electric vehicles tend to become a growing category of today's human means of transport. But, because these kind of vehicles are actually quiet, or even silent, the question of a dedicated sound design arises almost inevitably in order to make them more present—then secure—both for their proximity (pedestrians) and their users (driver). This being, current issues for a sound design research framework is then to exploit and explore sound properties that, first, will fix a goal of functionality (emergence, recognition, acceptance) and, second, will define guidelines for the development of new aesthetics to be included in a general design approach. Thus, a first study focusing on detection of warning signals in urban environments was achieved. Based on the state-of-the-art, a corpus of elementary signals was built and characterized in a time/frequency domain for representing basic temporal and spectral properties (continuous, impulsive, harmonic, etc.). A corpus of representative urban environments was also recorded and realistic sequences were mixed with a dynamic approaching-source model. A

reaction time experiment was conducted and leads to interesting observations: especially, specific properties promoting the emergence. Moreover, a seemingly significant learning effect also rises from the data and should be further investigated.

10:20

2aNSa5. Detectability and annoyance of warning sounds for electric vehicles. Etienne Parizet and Ryan Robart (Laboratoire Vibrations Acoustique, INSA-Lyon, 25 bis, av. Jean Capelle, Villeurbanne 69621, France, etienne.parizet@insa-lyon.fr)

Electric or hybrid vehicles are very quiet at low speeds, which represents a very good opportunity to reduce traffic noise annoyance in cities. On the other hand, this may be very hazardous for vulnerable pedestrians (e.g., visually impaired people). The aim of the eVADER project is to propose solutions in order to add warning sounds to such cars, while fulfilling two contradictory goals: sounds should be detectable but should not contribute to traffic noise annoyance. Different perceptual experiments have been conducted: the first one evaluated the influence of various timbre parameters on sound detectability. It was shown that an electric vehicle equipped with one particular sound was as easily detected as a diesel one, while keeping a very low level. Then, the influence of some timbre parameters (pitch and temporal modulation frequency) on the distance and speed as perceived by listeners was measured. These two experiments were conducted with sighted and visually impaired subjects. Finally, a third one evaluated the consequence on traffic noise annoyance of such warning sounds.

10:40

2aNSa6. Measurement, evaluation, and analysis of noise and vibrations produced by an Argentinean medium tank. Alan Chorubczyk, Francisco Ruffa (Sound Eng., Tres de Febrero Univ., Amenabar 1819, Ciudad Autónoma de Buenos Aires 1428, Argentina, alanchoru@gmail.com), Nicolás Urquiza (Sound Eng., Tres de Febrero Univ., Caseros, Argentina), Pablo Margaretic (Sound Eng., Tres de Febrero Univ., Bernal, Buenos Aires, Argentina), Damián Morandi (Sound Eng., Tres de Febrero Univ., Bella Vista, Argentina), Andrés Piegari, and Sebastián Ausili (Sound Eng., Tres de Febrero Univ., Ciudad Autónoma de Buenos Aires, Argentina)

In the present paper it is presented the procedure of measurement, evaluation and results' analysis of an Argentinean medium tank T.A.M. Since there is no regulation on noise emissions and vibrations caused by combat units either inside or outside the vehicle, standards that evaluate similar situations were used. Then, noise and vibrations inside the moving vehicle and noise emissions of acceleration and static situations were assessed. Consequently, the procedure and results were analyzed in order to propose a proper heavy combat units assessment procedure.

11:00

2aNSa7. Contribution analysis of vehicle exterior noise with operational transfer path analysis. Jakob Putner (AG Technische Akustik, MMK, Technische Universität München, Arcisstraße 21, Munich 80333, Germany, putner@tum.de), Martin Lohrmann (Müller-BBM VibroAkustik Systeme GmbH, Planegg, Germany), and Hugo Fastl (AG Technische Akustik, MMK, Technische Universität München, Munich, Germany)

Vehicle development regarding the emitted exterior noise is a challenging task. In addition to stringent legal requirements to reduce noise exposure, also high expectations of the sound quality have to be considered during the development process. In order to manipulate the vehicle exterior noise in a manner more efficient than trial and error, knowledge about the vehicle's sound sources, and their contributions to the overall noise is essential. In order to analyze the contributions of the several sound sources of a vehicle to the exterior noise Operational Transfer Path Analysis is used in the presented experiment. Therefore, transfer characteristics are estimated from measurements of the vehicle in typical operating conditions on an acoustic roller dynamometer. These data are used to synthesize the contributions at the response positions, i.e., the microphones of a simulated pass-by array, which also allow the simulation of the contributions during a pass-by measurement. Outcomes of the Operational Transfer Path Analysis are comprehensible contributions of the dominant sound sources to the vehicle exterior noise. The validation of the analysis results shows very good accordance between the simulated and measured overall vehicle exterior noise.

11:20

2aNSa8. Improving the acoustic performance of low noise road surfaces using resonators. Manuel Männel (Müller-BBM, Robert-Koch-Str. 11, Munich 82152, Germany, manuel.maennel@muellerbbm.de), Jens Forssén, and Bart van der Aa (Appl. Acoust., Chalmers Univ. of Technol., Gothenburg, Sweden)

Road surfaces made of porous asphalt are widely used to reduce the tire-road-noise generated during the rolling process of passenger cars and trucks. As the engine noise was reduced significantly in the last decades the tire-road-noise is the main sound source for driving speeds of 40 km/h (25 mile/h) and higher for passenger cars. This means that low noise road surfaces may not only be used on highways but also on inner-city main roads to generate a significant reduction on traffic noise. However, the acoustic performance of road surfaces made of porous asphalt is limited as a result of the trade-off between acoustic properties and road surface durability. By including resonators, e.g., of Helmholtz type in the porous road surface, it is possible to improve its absorbing performance without loss in durability. The paper describes recent research activities on such resonators in porous road surfaces made in the European project HOSANNA. The acoustic properties in terms of insertion loss have been calculated for different arrays of resonators. Measurements on realized porous road surfaces including resonators were carried out. The results show that resonators can improve the acoustic performance of porous road surfaces substantially.

11:40

2aNSa9. Guidance for new policy developments on railway vibration. Eulalia Peris, James Woodcock, Gennaro Sica, Calum Sharp, Andy Moorhouse, and David Waddington (The Univ. of Salford, Flat 2, 2 Claremont Grove, Manchester M202GL, United Kingdom, E.Peris@salford.ac.uk)

Vibration is one of the main problems associated with railways in residential areas. To ensure quality of life and well being of inhabitants living in the vicinity of route paths, it is important to evaluate, understand, control, and regulate railway noise and vibration. Much attention has been focused on the impact of noise from railway but the consideration of railway-induced vibration has often been neglected. This paper aims to provide policy guidance based on results obtained from the analyses of relationships estimated from ordinal logit models between human response and vibration exposure. This was achieved using data from case studies comprised of face-to-face interviews and internal vibration measurements ($N = 755$) collected within the study "Human Response to Vibration in Residential Environments" by the University of Salford. First, the implications of neglecting vibration in railway noise policies are presented. Second, the influence of different times of day when residents are exposed to railway vibration are presented and compared to current standards. Finally, the main factors that were found to influence railway vibration annoyance are presented and expressed as weightings. This work will be of interest to researchers and environmental health practitioners involved in the assessment of vibration complaints, as well as to policy makers, planners, and consultants involved in the design of buildings and railways.

Session 2aNSb

Noise: Distinguished Lecture

Victor Sparrow, Chair
Grad. Program in Acoust., Penn State, 201 App. Sci. Bldg., University Park, PA 16802

Chair's Introduction—8:55

Invited Paper

9:00

2aNSb1. The work of the Committee on Aviation Environmental Protection and the Development of International Noise Standards. Jane Hupe (Environ. Branch, Air Transport Bureau, ICAO, 999 Rue Univ., Montreal, QC H3C 5H7, Canada, jhupe@icao.int)

Environmental Protection is one of the Strategic Objectives of ICAO. The overall aim is to minimize the adverse environmental effects of global civil aviation activity. One of the key objectives is to establish noise Standards to limit and reduce the number of people affected by aircraft noise. This mandate is carried out by the Committee on Aviation Environmental Protection (CAEP), which, as a technical committee of the ICAO Council, is a recognized international forum of environmental experts from both member and observer States, intergovernmental organizations, including airlines, aircraft and engine manufacturers, airports, environmental non-governmental organizations, and UN bodies. ICAO has set International Standards for aircraft noise certification since the 1970s, and the purpose of this talk is to describe the process of developing these Standards while providing some details on recent developments, including the key outcomes of three years' worth of research leading up to the ninth meeting of the CAEP.

TUESDAY MORNING, 4 JUNE 2013 511CF, 10:00 A.M. TO 11:00 A.M.

Session 2aNSc

Noise and ASA Committee on Standards: International Aviation Noise Standards

Victor Sparrow, Chair
Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802

Invited Papers

10:00

2aNSc1. Setting noise stringency by international consensus. Yves R. Cousineau (Civil Aviation, Transport Canada, Place de Ville, 330 Sparks St., Ottawa, ON K1A 0N8, Canada, yves.cousineau@tc.gc.ca)

The paper will evoke the background of aviation noise and examine the international consensus process used to set aircraft noise stringency requirements and present the role of controlling the noise at the source within the context of the overall community noise issue. The paper will also examine the role of technology in this process and examine the growing interdependencies of noise reduction technology on CO_2 emissions, and on other emissions that impact air quality.

10:20

2aNSc2. Developing noise standards for future supersonic civil aircraft. Robbie Cowart (Gulfstream Aerosp. Corp., POB 2206, M/S R-07, Savannah, GA 31402, robbie.cowart@gulfstream.com)

With renewed interest in civil supersonics, NASA and industry researchers continue to make progress toward enabling quiet civil supersonic aircraft. Gulfstream Aerospace has long been interested in the development of an economically viable supersonic business jet; however, many regulatory challenges still remain for routine supersonic operation. Gulfstream's approach includes unrestricted supersonic flight over land to enable the same operational flexibility of its subsonic fleet. The largest technical barrier to achieving this end is mitigating the sonic boom created by flying at cruise speeds greater than Mach 1.2. At present, the United States and many other countries prohibit supersonic flight over land due to the loudness and public annoyance associated with sonic boom noise. In the United States, the FAA prohibits supersonic flight under FAR 91.817. Although the FAA has shown interest in reconsidering its position, the agency supports the noise and emissions standards setting process through the International Civil Aviation Organization and its

Committee on Aviation Environmental Protection. Development of future standards for sonic boom noise is a key component to enabling continued investment in civil supersonic research. This paper will outline the steps currently underway to assess the viability of defining low amplitude supersonic signature acceptability.

10:40

2aNSc3. Aircraft noise technology review and medium and long term noise reduction goals. Brian J. Tester (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom), Dennis Huff (Aeropropulsion, John H. Glenn Res. Ctr., Cleveland, OH), and Luc Mongeau (Mech. Eng., McGill Univ., 817 Sherbrooke St. West, Montreal, QC H3A 0C3, Canada, luc.mongeau@mcgill.ca)

This presentation will summarize the recommendations of the second ICAO Committee on Aviation Environmental Protection noise technology independent expert panel. The technologies being developed for reducing aircraft community noise in the mid term (10 years) and the long term (20 years) were reviewed. The review consisted of detailed presentations on aircraft noise reduction by various industry and government representatives on component noise reduction programs, highlighting novel noise reduction concepts being pursued and the progress that has been demonstrated or projected so far.

TUESDAY MORNING, 4 JUNE 2013 519B, 8:55 A.M. TO 10:20 A.M.

Session 2aPAa

Physical Acoustics: Nanoacoustics

Srikantha Phani, Chair

Dept. of Mech. Eng., The Univ. of British Columbia, 6250 Appl. Sci. Lane, Vancouver, BC V6T1Z4, Canada

Chair's Introduction—8:55

Invited Paper

9:00

2aPAa1. Bandgap formation mechanisms in periodic materials and structures. Lalitha Raghavan and Srikantha Phani (Dept. Mech. Eng., The Univ. of British Columbia, 6250 Appl. Sci. Lane, Vancouver, BC V6T1Z4, Canada, srikanth@mech.ubc.ca)

Bandgaps are frequency intervals of no wave propagation for mechanical waves. Spatial periodicity provides one mechanism for the emergence of bandgaps in periodic composite materials such as lattice materials, phononic crystals, and acoustic metamaterials. Coupling a propagating wave in a periodic medium with a local resonator provides an alternate mechanism for band-gap emergence. This study examines these two band-gap formation mechanisms using a receptance coupling technique. The receptance coupling technique yields closed-form expressions for the location of bandgaps and their width. Numerical simulations of Blochwaves are presented for the Bragg and sub-Bragg bandgaps and compared with the bounding frequency predictions given by the receptance analysis of the unitcell dynamics. It is observed that the introduction of periodic local resonators narrows Bragg bandgaps above the local resonant bandgap. Introduction of two fold periodicity is shown to widen the Bragg bandgap, thus expanding the design space. The generality of the receptance technique presented here allows straightforward extension to higher dimensional systems with multiple degrees of freedom coupling. Implication of this study for nano-electro-mechanical systems (NEMS) based filters will be discussed.

Contributed Papers

9:20

2aPAa2. Multi-resonance transduction near acoustic Brillouin zone in microscale ferroelectric. Igor Ostrovskii (Phys. and NCPA, Univ. of Mississippi, Oxford, Lewis Hall, Rm. 108, University, MS 38677, iostrov@phy.olemiss.edu) and Lucien Cremaldi (Phys. and Astronomy, Univ. of Mississippi, University, MS)

Excitation of the acoustic plate waves (PAW) in ferroelectric wafers with microscale domains and nanoscale interdomain walls is investigated theoretically and experimentally. The periodically poled structures were fabricated in the Z-cut 0.5-mm-thick $LiNbO_3$ wafers. Rf-current applied along the X-axis generates PAW in the fabricated multidomain acoustic superlattices having stop-bands and acoustic Brillouin zones. Two main effects are observed. First, a frequency of maximum acoustic amplitude does not coincide with the domain resonance frequency when a half-wavelength equal to domain length. Second, instead of known single-frequency domain resonance such as in bulk crystals, the series of two or more transduction resonances do exist. The theory, simulation, and experiments allow concluding that the transduction multi-resonances occur due to ultrasound diffraction by the interdomain nano-walls when the acoustic wavelength is close to multidomain period; it happens near acoustic Brillouin zone boundaries. Since different PAW-modes may be excited, the transduction multi-resonances appear at different frequencies. For example, 300-micron multidomain structure in $LiNbO_3$ demonstrates multiple transduction peaks in the frequency range of 5.6 to 8.3 MHz, just for lowest four symmetric and anti-symmetric PAW modes. The experimental results are in agreement with theory. The findings may be applied for designing new MEMS and therapeutic ultrasonic transducers.

9:40

2aPAa3. Using nonlinear ultrasound to measure microstructural changes due to radiation damage in steel. Kathryn H. Matlack (G.W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 788 Atlantic Dr., Sustainable Education Bldg., Ste. 306, Atlanta, GA 30332, katie.matlack@gatech.edu), Jin-Yeon Kim (School of Civil and Environ. Eng., Georgia Inst. of Technol., Atlanta, GA), James J. Wall (Elec. Power Res. Inst., Charlotte, NC), Jianmin Qu (Dept. of Civil and Environ. Eng., Northwestern Univ., Evanston, IL), and Laurence J. Jacobs (School of Civil and Environ. Eng., Georgia Inst. of Technol., Atlanta, GA)

This work demonstrates how nonlinear ultrasound (NLU) can be used to monitor radiation damage in nuclear reactor pressure vessel (RPV) steels. Radiation damage is a crucial concern in the nuclear industry since many nuclear plants throughout the United States are entering a period of life extension, meaning the RPV will be exposed to higher levels of neutron radiation than it was originally designed to withstand. Currently, there is no nondestructive evaluation (NDE) method to unambiguously characterize radiation damage in RPV steels, the development of which would enable the assessment of the integrity of the vessel, allowing operators to determine if they can continue to safely operate. NLU is an NDE technique that is sensitive to microstructural features in metallic materials. The physical effect monitored by NLU is the generation of higher harmonic frequencies in an initially monochromatic ultrasonic wave, arising from the interaction of the ultrasonic wave with microstructural features. Recent research has demonstrated that NLU is sensitive to the same microstructural changes that are produced in radiation damage, such as precipitate formation and changes in dislocation density. Current experimental and modeling results are presented that relate the nonlinear ultrasonic parameter to the amount of radiation damage in RPV steel materials.

10:00

2aPAa4. Synthesis and frequency dependent ultrasonic characterization of NiO-EG nanofluids. Meher Wan, Satyendra K. Verma (Phys. Dept., Univ. of Allahabad, Allahabad, UP 211002, India, meherwan24@hotmail.com), Dharmendra K. Pandey (Phys. Dept., PPN College, Kanpur Univ., Kanpur, UP, India), and Raja R. Yadav (Phys. Dept., Univ. of Allahabad, Allahabad, UP, India)

In the present paper, we have synthesized the NiO nanoparticles with chemical route. Further, the uniform suspensions of NiO nanoparticles in ethylene glycol of different concentrations have been prepared. Samples were characterized with Acoustical Particle Sizer (APS-100) for the frequency dependent ultrasonic attenuation in the respective samples; subsequently, the particle size determination and their distribution have been calculated with the help of ultrasonic attenuation. The structural parameters were also investigated with the microscopic techniques. There is good agreement between data produced by ultrasonic spectroscopy and the microscopic measurements.

TUESDAY MORNING, 4 JUNE 2013 — 519B, 10:20 A.M. TO 12:20 P.M.

Session 2aPAb

Physical Acoustics: Atmospheric Acoustics

D. Keith Wilson, Chair

U.S. Army Cold Regions Res. Lab., 72 Lyme Rd., Hanover, NH 03755

Contributed Papers

10:20

2aPAb1. Source localization results for airborne acoustic platforms. Vladimir E. Ostashev (Cooperative Inst. for Res. in Environmental Sci./Univ. of Colorado at Boulder and NOAA Earth System Res. Lab., Boulder, CO), Sylvain Cheinet (French-German Res. Inst. of Saint-Louis (ISL), 5 Rue General Cassagnou, Saint-Louis 68300, France, sylvain.cheinet@isl.eu), Sandra L. Collier, Chris Reiff, David A. Lygon (U. S. Army Res. Lab., Adelphi, MD), D. Keith Wilson (U.S. Army Cold Regions Res. and Eng. Lab., Hanover, NH), John M. Noble, and William C. Alberts, II (U. S. Army Res. Lab., Adelphi, MD)

Acoustic sensors are being employed on airborne platforms for source localization. Under certain atmospheric conditions, airborne sensors offer a distinct advantage over ground sensors. Among other factors, the performance of airborne sensors is affected by refraction of sound signals due to vertical gradients in temperature and wind velocity. A comprehensive experiment in source localization with an aerostat-mounted acoustic system was carried out in July 2010 at Yuma Proving Ground (YPG). Acoustic sources on the ground consisted of one-pound TNT denotations and small arms firings. The height of the aerostat was approximately 1 km above the ground. In this paper, horizontal, azimuthal, and elevation errors in source localization and their statistics are studied in detail. Initially, straight-line propagation is assumed; then refraction corrections are introduced to improve source localization and decrease the errors. The corrections are based on a recently developed theory [Ostashev *et al.*, J. Acoust. Soc. Am. (2008)] that accounts for sound refraction due to vertical profiles of temperature and wind velocity. During the 2010 YPG field test, the vertical profiles were measured only up to a height of approximately 100 m. Therefore, the European Center for Medium-range Weather Forecasts (ECMWF) is used to generate the profiles for July of 2010.

10:40

2aPAb2. A numerical approach to the climatology of near-surface sound levels. Sylvain Cheinet (French-German Res. Inst. of Saint-Louis (ISL), 5 Rue General Cassagnou, Saint-Louis 68300, France, sylvain.cheinet@isl.eu)

The near-surface sound levels propagated at distance from a known source show a large variability on the long term. This variability is essentially caused by the weather-dependence of the refractive characteristics: wind and temperature stratifications, turbulence. An approach to document this variability is to simulate the sound propagation under these varying characteristics at the selected site. This study uses a numerical model which physically describes the sound propagation including in presence of turbulence. The model is based on the parabolic equation, it ingests standard atmospheric parameters as input. The predicted sound levels for an example 40 Hz-frequency sound propagating at a 1.5 km-range are shown to combine the impacts of stratification and turbulence. The results are used to form the sound level climatology at several sites over the globe, based on existing climatological data. The obtained statistics are modulated by the dominant wind regimes, the seasonal and diurnal cycles. The sensitivity of these results to turbulence assessment is discussed.

11:00

2aPAb3. Statistical moments of broadband acoustic signals propagating in a refractive, turbulent atmosphere with spatial-temporal fluctuations in temperature and wind velocity. Vladimir Ostashev, D. Keith Wilson, Sergey N. Vecherin (U.S. Army Cold Regions Res. and Engineering Lab., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@noaa.gov), and Sandra L. Collier (U. S. Army Res. Lab., Adelphi, MD)

Propagation of broadband acoustic signals in a refractive, turbulent atmosphere with spatial-temporal fluctuations in temperature and wind velocity is considered. Starting from a parabolic wave equation, and using the Markov approximation and the hypothesis of locally frozen turbulence, closed-form equations for the statistical moments of arbitrary order of the sound-pressure field are derived for both sound propagation above an impedance ground and line-of-sight propagation. These equations generalize those obtained previously [Wilson and Ostashev, J. Acoust. Soc. Am. **109**, 1909–1922 (2001)], where the case of monochromatic sound waves and spatial fluctuations in temperature and wind velocity was considered. The general theory can be used for analysis of many statistical characteristics of broadband acoustic signals propagating in the atmosphere, e.g., temporal coherence, frequency decorrelation, and the pulse spread and wonder. Using this theory, the spatial-temporal coherence function of a broadband acoustic signal is calculated and analyzed. Knowledge of the theoretical coherence function is important for performing source localization with physics-based estimators, particularly maximum-likelihood estimators.

11:20

2aPAb4. A physical model for predicting the sound speed and attenuation coefficient in Titan's atmosphere based on Cassini-Huygens data. Andi Petculescu (Physics, Univ. of Louisiana at Lafayette, P.O. Box 44210, Lafayette, LA 70504, andi@louisiana.edu)

NASA and ESA are discussing plans for a collaborative mission to use montgolfieres to gather long-duration data in Titan's atmosphere. Acoustic sensors can listen for thunder, bolide explosions, wind noise, cryo-volcanoes, and many other phenomena. This emphasizes the need for accurate acoustic predictions for Titan. In 2005, during the descent of the Huygens probe on Titan, an active ultrasonic sensor measured the speed of sound over the last 12 km. Using the ambient pressure, density, temperature, and methane concentration measured by Huygens as inputs, as well as temperature- and pressure-dependent transport parameters extracted from NIST, a theoretical model has been developed to predict the sound speed and attenuation coefficient in Titan's atmosphere. Based upon non-ideal equations of state, the sound speed predictions agree quite well with Huygens measurements in the lower troposphere. The effect of measured zonal winds on tropospheric propagation is presented via ray-tracing, showing quiet zone predictions. The model can be extended to the upper atmospheric layers (since ambient data are available); nevertheless care must be taken to account for altitude dependent processes such as winds, clouds, aerosols, chemistry, gravity waves, etc. in order to increase the accuracy.

11:40

2aPAb5. Prediction of sound levels from high-altitude, broadband sources: Is there a Lloyd's mirror effect? D. Keith Wilson (U.S. Army Cold Regions Res. and Eng. Lab., Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), Vladimir E. Ostashev, and Sergey N. Vecherin (U.S. Army Cold Regions Res. and Eng. Lab., Engineer Res. and Development Ctr., Hanover, NH)

For a high-altitude, narrowband sound source, propagation models predict the presence of pronounced rings of constructive and destructive interference for listeners near the ground. This is an example of the Lloyd's mirror effect. But even when propagation predictions are made for realistic, broadband aircraft spectra, by partitioning the spectrum into octave or one-third octave bands, running the model at the center frequency of each band, and then summing, a residual Lloyd's mirror effect can still be apparent. By varying the approach to frequency selection in the calculation, it is shown that the rings actually have random locations and generally disappear when enough frequencies are sampled, thus implying that they are numerical artifacts. This outcome unfortunately implies that coherent calculations at many individual frequencies are required to perform a broadband calculation. Some techniques for improving convergence, such as importance sampling and performing calculations incoherently, are discussed.

12:00

2aPAb6. Pneumatic infrasound source: Theory and experiment. Justin Gorhum, Thomas Muir, Charles Slack, Martin Barlett, Timothy Hawkins (Appl. Res. Lab., The Univ. of Texas at Austin, P.O. Box 8029, Austin TX 78713, Austin, TX 78713, jgorhum@arlut.utexas.edu), Charles Tinney, and Woutjin Baars (Dept. of Aerosp. Eng., The Univ. of Texas at Austin, Austin, TX)

In prior work [J. Acoust. Soc. Am. **132**(3), 2074 (2012)], we experimentally demonstrated the feasibility of releasing compressed air through a rotating ball valve to create infrasound. The present paper seeks to analyze and model those as well as new outdoor measurements, with a view toward establishing a viable theoretical model of the process. Functions involving propagation and the response to frequency, source pressure, and signal type (tone burst and transient) are examined, as is the potential utility of the method in calibration and test applications. [Work supported by ARL:UT Austin.]

Session 2aPAc

Physical Acoustics: General Physical Acoustics I

Annie Ross, Chair
Dept. Mech. Eng., Ecole Polytechnique Montreal, CP 6079 Succ CEntre Ville, Montreal, QC H3C 3A7, Canada

Contributed Papers

9:00

2aPAc1. Guided waves scattering by discontinuities near pipe bends. Mihai V. Predoi (Mechanics, Univ. Politehnica Bucharest, Splaiul Independentei nr. 313, sect. 6, Bucharest, Bucharest 060042, Romania, predoi@cat.mec.pub.ro) and Cristian C. Petre (Strength of Mater., Univer. Politehnica Bucharest, Bucharest, Romania)

Guided waves became in recent years an useful tool in nondestructive testing of pipes used in many industrial applications. The torsional and longitudinal waves in pipes are the main choice for integrity inspection. The first step is the computation of the dispersion curves, for straight pipes. Since most pipes have bends, the problem of guided modes in the toroidal segment remains of interest. Various methods have been applied to solve this problem. The most promising numerical method to obtain the dispersion curves for a torus is based on finite elements (FE), using a standing waves model. Based on these dispersion curves, transmissions of longitudinal and torsional waves through a bend were also investigated. The present paper presents the scattering process produced by geometrical discontinuities such as circumferential welds before and after a pipe bend. Longitudinal L(0,2) mode is sent along the straight pipe in FE simulations, toward the bend. Reflected and transmitted modal amplitudes are determined for frequencies of interest. The capability of detecting a defect close to one of the two welds is thus assessed. The modes transmitted past the bend are also characterized. Comparisons with results obtained by other researchers are used to validate the method.

9:20

2aPAc2. Shear waves in inviscid compressible fluids. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

While elastic solids support compressional and shear waves, waves in ideal compressible fluids are usually thought of as compressional waves. Here, a class of acoustic-gravity waves is studied in which the dilatation is identically zero, and the pressure and density remain constant in each fluid particle. An exact analytic solution of linearized hydrodynamics equations is obtained that describes the shear waves in inhomogeneous, inviscid, compressible fluids with piece-wise continuous parameters in a uniform gravity field. The solution is valid under surprisingly general assumptions about the environment and reduces to some classical wave types in appropriate limiting cases. Free shear waves in bounded and unbounded domains as well as excitation of the shear waves by a point source are considered. Edge waves propagating along vertical and inclined rigid boundaries are found in rotating and non-rotating fluids. A possible role of the shear acoustic-gravity waves in a coupled ocean-atmosphere system is discussed.

9:40

2aPAc3. Photo-acoustic and ultrasonic investigation of the mixtures of water and several glycols. Wioletta Żwirbla, Bogumil B. Linde (Inst. of Experimental Phys., Univ. of Gdańsk, Wita Stwosza 57, Gdansk 80-952, Poland, fizbl@univ.gda.pl), and Ewa B. Skrodzka (Inst. of Acoust., Adam Mickiewicz Univ., Poznań, Poland)

In the paper, the results of the ultrasonic velocity and absorption measurements were presented, as well as the thickness the “ro” in water mixtures the polyethylene glycols (PEG-s), the ethylene glycol and diethylene glycol. The experiments were provided in the temperature range from 291.15 to 303.15 K for whole molar fraction. Adiabatic compressibilities were calculated from Laplace’s equation based on the experimental results obtained. Variations of these values with concentration and temperature were studied. Structural interactions and the formation of a compact pseudostable structure at very low concentrations of ethylene glycol and polyethylene glycols were observed. The plots of the adiabatic compressibility versus the mole fraction of PEG and EG display two characteristic points at low concentrations: the intersection of the isotherms and their minimum. Such relations between adiabatic compressibility, concentration and temperature are usually attributed to the formation of pseudo-stable molecular structures. To formulate a model of local structures present in the investigated molecular systems it is indispensable to get an insight into hydration of molecules and the formation of hydrogen bonds. Therefore, the attention was focused particularly on these problems.

10:00

2aPAc4. Numerical simulations of evolution of weak disturbances in vibrationally excited gas. Igor Zavershinskii, Vlavimir Makaryan (Physics, Samara State Aerosp. Univ., Moskovskoe Sh., Samara 443086, Russian Federation, ipzav63@mail.ru), and Nonna Molevich (Theoretical Phys., P.N. Lebedev Physical Inst. of RAS, Samara Branch, Samara, Russian Federation)

We consider a model of gas with an exponential law of relaxation of vibrational states of molecules excited by the external energy source (Joule heating in discharges, exothermic chemical reactions, optical pumping, etc). In such a medium, the second (bulk) viscosity coefficient inversion can take place due to the positive feedback between the sound perturbation and the nonequilibrium heat release. Such a medium with the negative viscosity is acoustical active. The existence of stationary nonlinear acoustical structures that are different from the step- or saw -wise shock wave structures are discussed basing on the solutions of general nonlinear acoustical equation [Molevich, Klimov, Makaryan, Int. J. Aeroacoust. No. 3–4 (2005)]. Using the numerical simulation of full one -dimensional (1D) system of relaxing gas dynamics, we show that any weak localized acoustical disturbance transforms into the sequence of self-sustained solitary pulses. Collisions of such pulses lead to their full reconstruction after the initial stage of the nonlinear increase of summarized amplitude. Using the 2D-system of relaxing gas dynamics, we consider the evolution of the noise signal into the non-stationary quasi-regular system of colliding self-sustained solitary pulses.

10:20

2aPAc5. Device for ultrasound imaging of standing trees. Andres Arciniegas (Aix-Marseille Université, CNRS, LMA UPR 7051, 31 chemin Joseph-Aiguier, Marseille 13009, France, arciniegas@lma.cnrs-mrs.fr), Loïc Brancheriau, Philippe Gallet (PERSYST/DIR, CIRAD, Montpellier, France), and Philippe Lasaygues (Aix-Marseille Université, CNRS, LMA UPR 7051, Marseille, France)

The aim of ARB’UST project is to develop an ultrasonic device for parametric imaging of standing trees. The device is designed to perform both transmission and reflection measurements, used for quantitative tomographic imaging. It allows various automatic acquisitions since the angular position of sensors can be precisely adjusted. The electronics and associated configuration enable particularly the measurement of velocity and attenuation of the

ultrasonic waves during their propagation within the medium. Two tomography experiments were conducted on a plane tree sample (before and after drilling a hole) and tomograms were calculated by the "Layer Stripping" algorithm. Our first results show that the artificial defect is detected.

10:40

2aPAc6. Cell structure in waves diffracted by a wedge in three-dimensional space and its implications to the field theory. Mitsuhiro Ueda (Predio Meguro Sci. Lab., 4-20-13 Meguro, Meguro-ku, Tokyo 153-0063, Japan, ueda-mt@nifty.com)

In the previous meeting, we have reported that if the aperture angle of a wedge-like region in 2D space bounded by perfectly reflecting planes is expressed as πN/M where N and M are relatively prime integers, the region can be divided into N cells and diffracted waves in a cell can be reconstructed by the sum of those in N-1 cells remained. This property holds for any positions of observation and source points. In this paper, it is shown that the same cell structure exists in the 3D wedge-like region bounded by rigid planes. In this case, the time dependent explicit solution for diffracted waves is available in closed form involving elementary functions only. Consequently the existence of cell structure can be shown analytically whereas in the 2D case it was shown numerically since the stationary solution for diffracted waves is expressed in asymptotic form involving Bessel functions. It is not easy to notice the cell structure without the new physical principle of diffraction, that is, virtual discontinuity principle of diffraction that has been proposed by us. Some physical implications of the principle to the field theory are mentioned lastly.

TUESDAY MORNING, 4 JUNE 2013 514ABC, 9:00 A.M. TO 11:00 A.M.

Session 2aPPa

Psychological and Physiological Acoustics: Binaural Hearing and Binaural Techniques II

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

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

Invited Papers

9:00

2aPPa1. Experiments on authenticity and naturalness of binaural reproduction via headphones. Janina Fels, Josefa Oberem, and Bruno Masiero (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr. 50, Aachen 52074, Germany, janina.fels@akustik.rwth-aachen.de)

Binaural stimuli presented via headphones need to be plausible in localization and sound coloration for a successful reproduction of an acoustic scene, especially for experiments on auditory selective attention. The goal is to provide artificially generated acoustic scenes in a way that the difference between a real situation and an artificially generated situation has no influence in psychoacoustic experiments. The quality and reliability of binaural reproduction via headphones comparing two different microphone setups (miniature microphone in open dome and ear plug) used for individualized head-related transfer functions and headphone transfer function measurements is analyzed. Listening tests are carried out focusing on authenticity, naturalness, and distinguishability in a direct comparison of real sources and binaural reproduction via headphones. Results for three different stimuli (speech, music, pink noise) are discussed. Furthermore, approaches to perform experiments on auditory selective attention with binaural reproduction versus dichotic reproduction are made.

9:20

2aPPa2. Perceptual equalization of artifacts of sound reproduction via multiple loudspeakers. Bernhard U. Seeber (Audio Information Process., Technische Universität München, Arcisstrasse 21, Munich 80333, Germany, seeber@tum.de) and Ervin R. Hafter (Dept. of Psychology, Univ. of California at Berkeley, Berkeley, CA)

Several techniques for reproducing spatial sounds via multiple loudspeakers have been developed in recent years. A key problem for such techniques are comb filter effects caused by the uncertainty of the receiver position when playing coherent sounds from multiple loudspeakers (spatial aliasing). Here we studied if panning between two closely spaced loudspeakers can create a virtual source that resembles that of a true source. This requires not only that panned direction and speaker position correspond, but also that source width, loudness, timbre, and temporal aspects are reproduced without perceptual error. A listening experiment in an anechoic chamber showed that panned sources differ primarily in loudness and timbre from a real source at the panned location. The artifacts are caused by effects of the head, and we investigated if they can be compensated by filtering the sounds. Compensation filters were derived from simulations of the sound field at the ears. Listening tests showed that compensation filters reduced panning errors to be nearly inaudible and level roving or reflections in the reproduction room made errors inaudible. We conclude that a simple equalization is sufficient to render panned sources from nearby speakers perceptually equivalent to real sources.

9:40

2aPPa3. Physical correlates of loudness transfer functions in binaural synthesis. Florian Völk and Hugo Fastl (AG Technische Akustik, MMK, Technische Universität München, Arcisstraße 21, München 80333, Germany, florian.voelk@mytum.de)

The frequency dependent level correction necessary for a binaural synthesis system to elicit via headphones the reference scene loudness of narrow-band signals is referred to as loudness transfer function. An ideal binaural synthesis system provides frequency independent loudness transfer functions for every listener. The frequency dependence of the average of a binaural synthesis system's individual loudness transfer functions has been shown to depend on the degree of individualization of the binaural synthesis system. In this contribution, perceptually acquired loudness transfer functions are compared from an auditory-adapted perspective to physical parameters of signals involved in the binaural synthesis process. The results provide quantitative relations between individual physical cues of the binaural synthesis output signals and the resulting loudness transfer functions.

10:00

2aPPa4. Toward a listening in spatialized noise test using complex tones. Jorg M. Buchholz, Harvey Dillon, and Sharon Cameron (Australian Hearing, National Acoust. Lab., 126 Greville St., Chatswood, NSW 2067, Australia, jorg.buchholz@nal.gov.au)

The Listening in Spatialized Noise-Sentences (LiSN-S) test has been widely applied to diagnose spatial processing disorder in both normally hearing and hearing impaired listeners who are proficient in English. The overall goal of the present study is to develop a spatial listening test that assesses similar spatial auditory processes as the LiSN-S test but does not rely on speech input and thus is language independent. Therefore, a three-alternative forced choice (3AFC) stream segregation task was implemented using a series of continuously in- or decreasing tone-complexes as targets and random tone-complexes as distractors and foils. Similar to the LiSN-S test the signals were either spatially co-located or separated using non-individualized HRTFs and the difference in thresholds defined the spatial release from masking (SRM). In order to achieve similar large SRM effects (of up to 14 dB) as observed with the LiSN-S test in normal hearing listeners, temporal jitter had to be introduced. The effect of the amount of temporal jitter was investigated on the SRM as a function of tone-complex duration. The results revealed that a jitter of about 30ms in combination with a tone-complex duration of about 30ms is sufficient to elicit the desired SRM.

10:20

2aPPa5. Auditory discrimination on the distance dependence of near-field head-related transfer function magnitudes. Yu Liu and Bosun Xie (Acoust. Lab., Phys. Dept., School of Sci., South China Univ. of China, Wushan Rd. 381#, Tianhe District, 301 Bldg. 18, Guangzhou, GuangDong 510641, China, janworc@gmail.com)

Distance dependence of head-related transfer functions (HRTFs) for nearby sound sources within 1 m is regarded as one of auditory distance perception cues. Accordingly, the near-field HRTFs have been used to synthesize sound sources at different distances in virtual auditory display. The present work analyzes the audibility of variation in near-field HRTF magnitudes with source distance. The calculated near-field HRTFs from KEMAR artificial head with a distance resolution of 0.01 m and a binaural model are used in analysis. The changes in loudness level spectra and interaural level difference are used as audible criteria. The result indicates that, as source distance decreases, the variation in HRTF magnitude with source distance become prominent and thereby audible. A psychoacoustic experiment is also carried out to validate the analysis. This work provides insight into the distance resolution of near-field HRTFs required in binaural virtual source synthesis.

10:40

2aPPa6. Issues in binaural hearing in bilateral cochlear implant users. Alan Kan, Heath Jones, and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, ahkan@waisman.wisc.edu)

Despite the success of bilateral cochlear implants (CIs) in restoring sound localization abilities in profoundly deaf individuals, their localization accuracy is still poorer than that of normal hearing listeners. One factor could be the behind-the-ear location of the microphones. However, when CI users were tested with stimuli filtered through individualized head-related transfer functions (HRTFs), they showed very little difference in sound localization performance with different microphone locations (behind-the-ear versus in-the-ear). Another factor is the different implantation depths of the electrode arrays at the two ears. Since CIs are typically fitted independently in each ear at the clinic, it is likely that binaural information at a particular frequency can be presented mismatched across the ears. By simulating different amounts of interaural frequency mismatch at single electrode pairs, CI users showed poorer fusion and lower binaural sensitivity with increasing interaural mismatch. Good lateralization and fusion was achieved on or near a pitch-matched pair of electrode. Additionally, results from a separate study showed lateralization performance was typically maintained with simultaneous stimulation of multiple, pitch-matched pairs of electrodes. These results demonstrate methods beyond just changing the microphone position are needed to improve sound localization performance in CI users. [Work supported by NIH-NIDCD (R01-DC003083) and NICHD (P30-HD03352).]

Session 2aPPb

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Laurel H. Carney, Chair
Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642

Chair's Introduction—11:15

Invited Paper

11:20

2aPPb1. Physiological and behavioral studies of sound localization. Tom C. Yin (Neuroscience, Univ. of Wisconsin, 290 Med. Sci. Bldg., Madison, WI 53706, tcyin@wisc.edu)

A critical job of the auditory system is to localize sounds, which depends upon spectral cues provided by the filtering of the pinna for vertical localization and interaural time (ITDs) and level disparities (ILDs) for horizontal localization. We found anatomical and physiological specializations in the circuits that encode these cues. Cells in the medial superior olive (MSO) function as high resolution coincidence detectors or cross-correlators and their inputs have enhanced temporal synchronization compared to auditory nerve fibers, and the speed of the calyx of Held synapse helps to convey the inhibitory input from the contralateral ear to the LSO synchronously with the excitatory input from the ipsilateral ear even though it has to travel farther with an additional synapse. We have also been studying the psychoacoustics of sound localization in the cat by training them to look at sounds. Cats demonstrate high accuracy and precision when localizing with their head unrestrained. Their mobile ears have a reflex in response to head movement that keeps the ears pointed toward the sound source despite head movements. Cats also experience the precedence effect and a physiological correlate of the effect can be seen in recordings from the inferior colliculus.

TUESDAY MORNING, 4 JUNE 2013 512CG, 8:55 A.M. TO 10:40 A.M.

Session 2aSA

Structural Acoustics and Vibration: History and Application of Constrained Layer Damping

J. Gregory McDaniel, Cochair
Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215

Kenneth G. Foote, Cochair
Woods Hole Oceanogr. Inst., 98 Water St., Woods Hole, MA 02543

Chair's Introduction—8:55

Invited Papers

9:00

2aSA1. Analysis and optimization of constrained layer damping treatments using a semi-analytical finite element method. James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, jgm@bu.edu) and Elizabeth A. Magliula (Vehicle Dynam. and Signature Control Branch 8233, Naval Undersea Warfare Ctr. Div. Newport, Newport, RI)

The present work investigates the physics of constrained layer damping treatments for plates and beams by a semi-analytical finite element method and presents applications of the method to the optimization of damping treatments. The method uses finite element discretizations in the thickness coordinate and propagating wave solutions in the remaining coordinates and therefore provides more generality and accuracy than existing analytical approximations. The resulting dispersion equation is solved for complex-valued wave numbers at each frequency of interest. By choosing sufficiently fine discretizations in the thickness coordinate, the method gives accurate estimates of wave numbers. The numerical implementation of the method is an efficient yet general tool for optimizing damping treatments with respect to material properties and dimensions. It explicitly allows for the possibility of analyzing structures with several layers where the material of each layer may be isotropic or orthotropic. Examples illustrate the numerical efficiency of the implementation and use this efficiency to provide optimizations of constrained layer damping treatments.

9:20

2aSA2. An overview of constrained-layer damping theory and application. Benjamin Shafer (Building Acoust., Conestoga-Rovers & Assoc., Inc., 1117 Tacoma Ave. South, Tacoma, WA 98402, bshafer@craworld.com)

Beginning in the early 1930s a variety of theoretical and experimental research has been published regarding the development and use of damping. What began as an experiment to reduce noise and vibration in metals and plastics has become a common treatment in an amalgam of applications. Constrained-layer damping (CLD) is a specific method of treatment commonly used in the aerospace and military industries. CLD may be described as a type of shear-related energy dissipation achieved by interconnecting two or more structural materials using a relatively thin viscoelastic layer. Among the advantages of using CLD as a damping treatment are the ability to obtain high loss factors with relatively thin configurations and that the stiffness of the composite system is not markedly increased. The analytic development of constrained-layer damping will be presented along with a brief discussion of the applications of CLD throughout history.

Contributed Papers

9:40

2aSA3. Numerical prediction of the vibroacoustic of sandwich panels with add-on damping. Imen Rzig and Noureddine Atalla (Mechanical, Univ. of Sherbrooke, E3-2115, 2500 Boulvard de l'université, Sherbrooe, Qc, QC J1K2R1, Canada, imen.rzig@usherbrooke.ca)

This paper discusses the numerical modeling of the vibroacoustic response of sandwich-composite panels with added-on damping, under mechanical and acoustical excitations. The studied damping is in the form of a viscolaetsic layer located within the panel. A modal synthesis approach is used for the calculation of the structural response and the Rayleigh's integral is used for the acoustic response (the panel is assumed flat and baffled). Since the panel has a viscoelastic core, a methodology is presented to handle efficiently the modeling of the frequency depended properties of the viscoelastic layer. A direct frequency response is used to validate the proposed approach. Next, a parameters study on the effect of the viscoelastic layer location is presented, In particular, three locations are compared: within the Honeycomb core, within the skins and added to the skin with a constraining layer. The effects of the excitation type on the vibration and acoustic response are also discussed.

10:00

2aSA4. A dynamic response of a laminated windshield with viscoelastic core—Numerical vs experimental results. Kaiss Bouayed (ESI Group, 20, rue du Fonds Pernant, Compiègne 60200, France, kaiss.bouayed@esi-group.com) and Mohamed-Ali Hamdi (Laboratoire ROBERVAL, Université de Technologie de Compiègne, Compiegne, France)

The dynamic response of a laminated windshield with a viscoelastic core is computed using a simplified modal method combined with a quadratic sandwich finite element. The method is based on a modal expansion of the displacement field using a constant young modulus of the core layer. The frequency dependence of the complex modulus of the core is taken into account using the residual dynamic stiffness matrix. The method is applied to predict the frequency response of two types of laminated windshield using a standard and acoustic PVB cores. Numerical results are compared in a first step with those obtained using the direct solver of Nastran software, and in a second step with experimental results obtained by a laser vibrometer. Comparisons show a very good agreement between experimental and numerical results and demonstrate the efficiency of the simplified modal solving method and the developed parabolic sandwich element. The method will be applied to compute the coupled vibro-acoustic frequency response of a full vehicle body integrating a laminated windshield and glass surfaces.

10:20

2aSA5. Nonlinear moduli estimation for rubber-like media with local inhomogeneities elastography. Timofey Krit, Valeriy Andreev, and Victor Kostikov (Dept. of Acoust., Faculty of Phys., Moscow State Univ., Leninskie Gory, Bldg. 1/2, Moscow, Moscow 119991, Russian Federation, timofey@acs366.phys.msu.ru)

Static shear deformations of a plane-parallel layer of rubber-like material created simultaneously with the uniaxial compression are considered. The layer is fixed between the rigid plates. Displacement of one plate relative to the other resulted in shear strain of the layer. This strain could reach 0.6 of the layer thickness. At such strain, effects due to the cubic nonlinearity arise. It is shown that measuring the dependence of the shear stress on the shear strain along one axis at different compression along the perpendicular axis one could determine nonlinear Landau parameters. The measurements were performed in two layers of polymeric material plastisol of 7 mm thickness with a rectangular base 8.9×8.9 cm, mounted between three aluminum plates. The upper plate was loaded with masses ranging from 0 to 25 kg and was fixed in each series of the stress-strain measurements. The values of the Landau coefficient A were measured in layers with different value of linear shear modulus. [Work supported by the Russian Foundation for Basic Research (Grant Nos. 12-02-00114 and 12-02-31418), and grant of the Government of the Russian Federation 11.G34.31.0066.]

Session 2aSC

Speech Communication: Linking Perception and Production (Poster Session)

Meghan Clayards, Chair
McGill Univ., 1085 Ave. Dr. Penfield, Montreal, QC H3A 1A7, 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.

2aSC1. The effect of accommodation on perceived vocal aesthetics. Grant L. McGuire (Linguistics, UC Santa Cruz, 1156 High St., Stevenson Acad. Services, Santa Cruz, CA 95064, gmcguir1@ucsc.edu), Molly Babel, and Jamie Russell (Linguistics, Univ. of British Columbia, Vancouver, BC, Canada)

We conducted an auditory naming task (n = 20) using eight model talker voices previously rated for attractiveness and prototypicality such that the most attractive, least attractive, most typical, and least typical voice for each gender served as a model talker. Female shadowers accommodated more than males, particularly to the Most Attractive Female model. This finding led us to question if in the course of accommodation to an attractive female voice, female shadowers themselves become more vocally attractive. We then conducted an AX task where listeners judged whether shadowers' baseline or shadowed productions were more attractive. Our results suggest that shadowers do modulate their perceived attractiveness in the course of accommodating; in particular, the more females accommodated to the Most Attractive Female model, the more attractive her own voice became. We are currently running a second study exploring whether shadowers' voices change in perceived typicality when accommodating the Most Typical and Least Typical voices, both of which also garnered large amounts of accommodation in the original auditory naming task. In general, our results demonstrate that the process of accommodation involves the mirroring of multidimensional speech characteristics, which in turn signal vocal aesthetics to listeners.

2aSC2. Coordinating conversation through posture. Martin A. Oberg, Eric Vatikiotis-Bateson, and Adriano Barbosa (Linguistics, UBC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, martin.oberg@alumni.ubc.ca)

Conversation is dynamic and interactive. The importance of coordinated movement in conversation has been studied through perceptual measures of synchrony and recently through quantitative analyses of multi-dimensional motion data. The present study describes the postural system as being integrated with the communication process through an analysis of interlocutors' coordination of rigid-body head motion, postural shifts on forceplates, and motion computed from audio-visual recordings. Coordination is measured two ways: (1) holistically, as the scaling of speakers' motion over the duration of a conversation (i.e., the presence of movement encourages more movements) and (2) through analyses of the instantaneous correlation between motion signals from each speaker (i.e., a search for similar patterns of movement across time). These two approaches are evaluated in their ability to categorize conversation types. Preliminary results show that a stability emerges in the amount of correlation across conversations. Variations in the pattern of stability are analyzed as evidence of differences between general interactional coordination and linguistic coordination.

2aSC3. The role of voice similarity in accommodation. Sophie A. Walters, Molly Babel (Linguistics, Univ. of British Columbia, Totem Field Studios 2613 W Mall, Vancouver, BC V6T 1Z4, Canada, sophia.alex.walters@gmail.com), and Grant McGuire (Linguistics, Univ. of California Santa Cruz, Santa Cruz, CA)

Studies of accommodation show that some talkers are perceived as accommodating more than others. One possibility is that the similarity of the shadower's voice to a model talker's can account, in part, for the amount of perceived accommodation. To determine this, we conducted an auditory naming task having eight model talker voices previously rated for attractiveness and prototypicality, such that the Most Attractive and Least Attractive and Most Typical and Least Typical voices for each gender were used as models. Twenty participants completed an auditory naming task with these eight voices. A separate group of 20 listeners rated the similarity of model tokens and shadower's baseline productions using a visual analog scale. The results of this task were compared to the perceived accommodation results from a separate AXB rating task. Overall, female voices that were more different from the models showed more accommodation. This effect was not found for males, who generally showed less accommodation overall. These findings suggest that talkers either accommodate more when their voice is more distinct from the model talker's voice, or perhaps more likely, that such changes are more perceptible to listeners. Further explorations of the data are underway to tease apart these possibilities.

2aSC4. Training Korean second language speakers on English vowels and prosody. Dong-Jin Shin and Paul Iverson (Univ. College London, Room 326, Chandler House, 2 Wakefield St., London WC1N 1PF, United Kingdom, dj.shin.09@ucl.ac.uk)

This study trained 36 Korean L2 speakers on vowel identification and prosody recognition (focus and lexical stress), with the aim of investigating the extent to which training improves general speech perception abilities or specific underlying processes. Vowel training was accomplished with a high-variability identification training technique (multiple talkers and words), and prosody training was accomplished using a category discrimination task in which they needed to choose sentences based on focus or words based on syllable stress. The results demonstrated that both trainers reduced vowel epenthesis and improved syllable stress perception, vowel training improved vowel identification more, and prosody training better improved focus perception in sentences. Both types of training can thus work in a complementary fashion to improve overall speech recognition.

2aSC5. Computer-based English /r/-/l/ perceptual training for Japanese children. Yasuaki Shinohara and Paul Iverson (Speech, Hearing and Phonetic Sci., Univ. College London, Rm. 326, 2 Wakefield St., London WC1N 1PF, United Kingdom, y.shinohara@ucl.ac.uk)

Computer-based perceptual training has proven successful for improving English /r/-/l/ perception by Japanese adults, but this has not been tested with younger age groups, who presumably have greater perceptual plasticity. The present study examined phonetic training for children 6–8 years old. The training program included identification and discrimination tasks with word-initial English /r/-/l/ minimal pairs (e.g., rock–lock), with each participant completing ten sessions. The results demonstrated that children improved their English /r/-/l/ identification, although identification in untrained positions such as medial and consonant clusters did not improve as much as in the trained word-initial position. In addition, older children in this age range improved more than did younger children, suggesting that the

ability to use this kind of program may improve with age, even though perceptual plasticity for speech presumably declines with age.

2aSC6. The role of acoustic/perceptual salience in directional asymmetry in infant stop/fricative contrast perception. Young-Ja Nam and Linda Polka (McGill Univ., 1266 Pine Ave. West, Montreal, QC H3G 1A8, Canada, young.nam@mail.mcgill.ca)

The presence of stops in a language implicates the presence of fricatives but the reverse is unattested. Similarly, infants' producing fricatives implies that they acquired stops. The privileged status of stops influences infants' stop/fricative perception. For example, Altvater-Mackensen and Fikkert (2010) reported that Dutch-learning 14-month-olds noticed a fricative to stop change but not vice versa. These findings were interpreted in terms of phonological specifications while dismissing acoustic/perceptual factors. In this study, we assessed whether pre-linguistic infants show perceptual asymmetry. We tested English and French 4–5-month-olds using the look-while-listen procedure in which they were presented native nonsense syllables -/bas/ and /vas/. A mixed ANOVA showed a significant interaction between trial type and group ($p = 0.027$). Infants in /vas/-habituated group noticed the switch when the habituated fricative changed to a stop but infants in /bas/-habituated group did not notice the switch when the habituated stop changed to a fricative. This perceptual asymmetry in infants before babbling and word recognition stage indicates the potential role of acoustic/perceptual factors. We suggest that the above-mentioned privileged status of stops may reflect the possibility that stops are acoustically/perceptually more salient than fricatives. This salience difference is predicted to induce directional asymmetry in stop/fricative contrast perception.

2aSC7. Effects of acoustic variability on infant speech perception. Stephanie L. Archer (School of Commun. Sci. and Disord., McGill Univ., 6255 Rue Sherbrooke Ouest, Apt #5, Montreal, QC H4B 1M6, Canada, stephanie.archer2@mail.mcgill.ca), Suzanne Curtin (Psychology, Univ. of Calgary, Calgary, AB, Canada), and Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada)

From birth, infants are capable of discriminating many of the speech sounds that occur cross-linguistically [Werker and Tees (1984)]. However, there are cases where the direction of presentation of contrasts reveals asymmetries in the discrimination of some speech contrasts, suggesting that some phonetic categories are more salient than others. These asymmetries may be due to an inherently uneven perceptual space [Polka and Bohn (2011)] or shaped by distributional properties of the input [Anderson *et al*. (2003)]. We explore whether acoustic variability also cause perceptual asymmetries. Six- and 9-month-olds participated in a discrimination task comparing English legal and illegal stop-liquid onsets (e.g., /kla/-/tla/ & /pla/-/tla/). Infants discriminated coronal versus bilabial onsets ($p < 0.05$), but not coronal versus velar ($p > 0.05$). Analysis of adult productions revealed that velar stop-liquid onsets showed more variability in their production, suggesting acoustic variability affects infants' perception. The current study provides a more direct test of the hypothesis that acoustic variability drives perceptual biases. Using the same clusters, we are exploring 9-month-olds' directional asymmetries. Preliminary results show that 9-month-olds successfully discriminate /dla/ after familiarization to /bla/ ($p = 0.05$; $n = 5$). Further investigation will reveal whether presentation direction affects infants' sensitivity to acoustic variability.

2aSC8. Infant recognition of infant vocal signals. Matthew Masapollo, Linda Polka (Commun. Sci. & Disord., McGill Univ., 1266 Pine Ave. West, Montreal, QC H3G 1A8, Canada, matthew.masapollo@mail.mcgill.ca), Lucie Menard (Dept. of Linguistics, Univ. of Quebec at Montreal, Montreal, QC, Canada), and Athena Vouloumanos (Dept. of Psychology, New York Univ., New York, NY)

Most of the speech accessible to infants who are not yet babbling will not have infant vocal properties; yet their perception of infant vocal signals is critical for speech development. We report three experiments designed to assess whether 4- to 5-month-olds recognize and preferentially attend to vowels produced by infant talkers over vowels produced by adult talkers. Infants were tested in a sequential preferential looking procedure using isolated vowels synthesized by VLAM. In experiment 1, infants listened significantly longer to vowels produced by an infant talker than an adult (female). In experiment 2, infants failed to show any listening preference for infant versus adult vowels synthesized with matching (infant-appropriate) f0 values, suggesting that infants either recognized the un-natural pairing of f0 and formant structure in these adult vowels or are attracted to high f0 values. Failing to support the latter interpretation, infants in experiment 3 showed no listening preference when presented infant vowels with different (infant-appropriate) f0 values. Together, these findings suggest that young infants recognize the converging vocal (source and filter) properties that specify an adult and an infant talker. These recognition skills appear to be available prior to babbling, and thus are available to support early vocal learning.

2aSC9. Infants' categorization of vowels with infant vocal properties. Linda Polka, Matthew Masapollo (McGill Univ., 1266 Pine Ave. West, Montreal, QC, Canada, linda.polka@mcgill.ca), and Lucie Menard (Dept. of Linguistics, Univ. of Quebec at Montreal, Montreal, QC, Canada)

Prior research shows that infants can recognize phonetic equivalence among vowels produced by adult men, women, and children. It is unknown whether this ability extends to infant vowel productions, which have unique properties due to the size and geometry of the infant vocal tract. The present study was undertaken to determine whether infants recognize infant vowel productions as phonetically equivalent to vowels produced by adults and children. Infants (4–6 months) were tested in a look-to-listen procedure using isolated vowels, /i/ and /a/, synthesized to simulate productions by men, women, children and a 6-month-old. Infants were first habituated to diverse productions of the same vowel produced by several adult male, female, and child speakers while they fixated on a checkerboard. Following habituation, infants were then presented infant productions of the same vowel (familiar) and the other vowel (novel) in four test trials. A novelty effect (novel > familiar) was observed showing that infants recognized the familiar infant vowel to be similar to the habituation vowel. The findings are discussed in terms of the emergence of perceptual constancy in the development of vowel perception raising issues about how and when such knowledge is acquired in relation to the infant's own productions.

2aSC10. Estimation of vocal tract area functions in children based on measurement of lip termination area and inverse acoustic mapping. Kate Bunton, Brad H. Story (Speech, Language, and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721, bunton@u.arizona.edu), and Ingo R. Titze (National Ctr. for Voice and Speech, Univ. of Utah, Salt Lake City, UT)

Although vocal tract area functions for adult talkers can be acquired with medical imaging techniques such as magnetic resonance imaging (MRI), similar information concerning children's vocal tracts during speech production is difficult to obtain. This is largely because the demanding nature of the data collection tasks is not suitable for children. The purpose of this study was to determine the feasibility of mapping formant frequencies measured from the [i, ae, a, u] vowels produced by three children (age range 4 to 6 years), to estimated vocal tract area functions. Formants were measured with a pitch-synchronous LPC approach, and the inverse mapping was based on calculations of acoustic sensitivity functions [Story, J. Acoust. Soc. Am. **119**, 715–718]. In addition, the mapping was constrained by measuring the lip termination area from digital video frames collected simultaneously with the audio sample. Experimental results were augmented with speech simulations to provide some validation of the technique. [Research supported by NIH R01-DC011275.]

2aSC11. Acoustic characteristics of Danish infant directed speech. Ocke-Schwen Bohn (Ctr. on Autobiographical Memory Res., Dept. of Psychology and English, Aarhus Univ., Sejts Alle 20a, Risskov DK-8240, Denmark, engosb@hum.au.dk)

Danish presents several challenges for language learners, such as a very densely packed upper portion of the acoustic vowel space, and a sibilant contrast that is acoustically less distinct than in, e.g., English. The present study examined whether Danish caregivers enhance Danish contrasts when speaking to their 18 month old children (infant directed speech—IDS) as opposed to an adult (adult directed speech—ADS). Caregivers were recorded talking about toy animals in conversations with their child and with an adult interlocutor. The toy names were designed to elicit Danish contrasts differing in voice onset time and in place of articulation for sibilants, and vowels which are close neighbors in the crowded Danish vowel space. The dependent variables for the comparison of IDS to ADS were as follows: VOT differences for homorganic stop consonants, the frequency at the amplitude peak for the sibilants, the Euclidean F1/F2 differences between vowels, F0 of the stressed (first) syllable in the toy name, as well as the duration of the stressed syllable, the vowels, and the fricatives. Results of the acoustic differences between ADS and IDS were compared to the results of parents' reports on the children's productive and receptive vocabulary knowledge. [Work supported by the Danish National Research Foundation –Danmarks Grundforskningsfond.]

2aSC12. Inter-rater agreement on Mandarin tone categorization: Contributing factors and implications. Puisan Wong, Lingzhi Li, and Xin Yu (Otolaryngol.–Head and Neck Surgery, The Ohio State Univ., 915 Olentangy River Rd., Columbus, OH 43212, pswResearch@gmail.com)

Factors that may/may not influence inter-rater reliability in assessing the accuracy of monosyllabic Mandarin tones produced by children and adults were examined in three experiments. Experiment 1 investigated inter-judge reliability in two groups of Mandarin-speaking adults—one group from China and the other from Taiwan—on their categorization of filtered tones produced by adults and children. The results showed that the magnitude of inter-rater agreement was associated with the production accuracy of the speakers; the judges attained lower agreement in categorizing children's tones than adults' tones. All judges who indicated that Mandarin was their strongest language and that they had learned and used Mandarin since birth performed similarly in their tone categorization despite the fact that they came from and were residing in different countries. Similar results was found in experiment 2, in which one group of the judges in experiment 1 categorized tones produced by a new and larger group of adults and children, and in experiment 3, in which a different group of adults categorized another new set of tones produced by a different group of speakers. Implications of the findings in research design will be discussed. [Work supported by NIH-NIDCD (1 F31 DC008479-01A1) and NSF (OISE-0611641).]

2aSC13. Effects of phonological training on tone perception for English listeners. Chang Liu and Bharath Chandrasekaran (Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, changliu@utexas.edu)

The goal of this study was to examine the extent to which phonological training improves categorical perception of Mandarin Chinese tones in native speakers of American English. Two sets of F0 continuums were generated from rising to level tones and from falling to level tones. Participants underwent identification and discrimination tasks before (pre-training) and after (post-training) phonological training. Pre-training tests showed that the tone identification shifted gradually from contoured tones (rising/falling) to level tones as a function of F0 frequency, while tone discrimination was near a chance rate across and within tone boundary. Phonological trainings were provided to listeners in two consecutive days with each session lasting for one hour. In phonological training, listeners were provided immediate feedback (correct/incorrect) after making response in Mandarin tone patterns. Results showed that tone identification function became significantly steeper with phonological training. Although no prominent peaks were found across tone boundaries in the discrimination function, the accuracy rate and response time of tone discrimination improved after training. Ongoing work is now testing the extent to which a longer training regiment can enhance categorical perception of tones.

2aSC14. Are two-year-olds sensitive to anticipatory coarticulation? Caterina Minaudo and Elizabeth K. Johnson (Psychology, Univ. of Toronto, 3359 Mississauga Rd. N. CCT 4110, Mississauga, ON L5L1C6, Canada, c.minaudo@mail.utoronto.ca)

Eyetracking studies have shown that adults are highly sensitive to subphonemic detail in speech [e.g., Shatzman and McQueen (2006)]. In some circumstances, adults use subphonemic coarticulatory information to anticipate which word will occur next in the speech stream [McDonough *et al*. (2009)]. In the current study, we ask whether two-year-old children use anticipatory coarticulation in a similar manner. Sixteen children were presented with pairs of images. In half of the trials, the names of the images presented on the screen had matching phonological onsets (e.g., doggy and ducky) that also matched in syllable length (e.g., monosyllabic or disyllabic). In the remaining trials, the names of the images had mismatching phonological onsets (e.g., cake and strawberry). In addition, a portion of each trial type was identity spliced (e.g., informative anticipatory coarticulation) and a portion was cross-spliced (e.g., misleading anticipatory coarticulation). We predicted that if two-year-olds are sensitive to anticipatory coarticulation, then they should be slowest to recognize named targets when the heard label was cross-spliced and the two objects on the screen had mismatching phonological onsets. However, all children looked to the named targets equally fast regardless of trial condition. Thus, no evidence of sensitivity to anticipatory coarticulation was observed.

2aSC15. Perception of speaker age in children's voices. Peter F. Assmann (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, MS GR 41, Box 830688, Richardson, TX 75075, assmann@utdallas.edu), Santiago Barreda, and Terrance M. Nearey (Dept. of Linguistics, Univ. of Alberta, Edmonton, AB, Canada)

To study the perception of speaker age in children's voices, adult listeners were presented with vowels in /hVd/ syllables either in isolation or in a carrier sentence, and used a graphical slider to register their estimate of the speaker's age. The data showed a moderate correlation of perceived age and chronological age. Age estimation accuracy was fairly constant across age up to about age 11, but there was a systematic tendency for listeners to underestimate the ages of older girls. This tendency was actually enhanced when listeners were informed of the speaker's sex. Age estimation accuracy was higher for syllables embedded in a carrier sentence. Linear regression analyses were conducted using acoustic measurements of the stimuli to predict perceived age. These analyses indicated significant contributions of fundamental frequency, duration, vowel category, formant frequencies, as well as certain measures related to the voicing source. The persistent underestimation of age for older girls, and the effect knowledge of speaker sex has on this underestimation suggest that acoustic information is combined with expectations regarding speakers of a given sex in arriving at an estimate of speaker age.

2aSC16. Serial order recall for rapid auditory presentations of vowel sequences: The effect of age. Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, 1621 Greene St., Columbia, SC 29208, fogerty@sc.edu) and Larry E. Humes (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

Temporal processing declines with age may reduce memory of rapidly presented auditory sequences. The current experiment investigated vowel sequence recall for two- and four-item vowel sequences presented at six different stimulus onset asynchronies (SOA) that spanned identification performance at 50% correct. Young, middle-age, and older adults participated in all tasks. For two-item sequences, a functional difference was observed between the age groups. Older and younger listeners had a qualitatively different pattern of recall, while performance for the middle age group approximated performance of either the young or older group, dependent upon the presentation rate (i.e., SOA). For the four-item sequences, results demonstrated the standard serial position curve. Increasing the rate of presentation by decreasing the SOA had the most profound effect on the middle items of the sequence for which subjects had the poorest retention. Overall, when temporal order performance was equated at the presentation rate corresponding to each individual's 50% threshold, recall accuracy for each position across the age groups was highly similar. These results suggest that declining temporal order performance of rapid sequences for older listeners is not the result of poorer recall performance, but is more related to sensory processing declines of rapidly presented temporal sequences.

2aSC17. Compensatory articulation in amyotrophic lateral sclerosis: Tongue and jaw interactions. Sanjana Shellikeri, Yana Yunusova (Speech Language Pathology, Univ. of Toronto, 253 South Park Rd., PH2, Thornhill, ON L3T0B4, Canada, sanjana.shellikeri@mail.utoronto.ca), Danielle Thomas (Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), Jordan Green (Special Education and Commun. Disord., Univ. of Nebraska-Lincoln, Lincoln, NE), and Lorne Zinman (Sunnybrook Health Sci. Ctr., Toronto, ON, Canada)

Previous acoustic studies on speech deterioration in amyotrophic lateral sclerosis (ALS) demonstrated that those at more advanced stages of disease show reduced F2 (second formant) slopes presumably due to disease-related changes in the tongue. Other studies have shown that patients with ALS use their jaw to compensate for decreased tongue function in speech. However, no study to date

has examined the compensatory role that the jaw has on maintaining the acoustic signatures of vocalic segments. This study will report F2 slope differences in vowels and diphthongs produced with and without jaw stabilization via a bite block. Based on previous studies, I hypothesized that the bite block will affect F2 slope measures in individuals with significant tongue impairment on the oral-motor examination and low speech intelligibility scores. Thirty participants repeat a carrier phrase “Say _ again” with words “wax, sip, yo-yo, sight” three times with and without the bite block. Kinematic measures of the distance, time, and speed, and acoustic measures of F2 slope are reported. The data will be discussed in the context of the role of the tongue in vowel production.

2aSC18. Relationship between articulation and mispronunciation detection in children with speech delay: Perception of unfamiliar speech vs. their own speech. Mark Paullin, Kyoko Nagao, and H Timothy Bunnell (Ctr. for Pediatric Auditory and Speech Sci., Nemours Biomed. Res., CPASS Ste., 1701 Rockland Rd., Wilmington, DE 19803, paullin@asel.udel.edu)

We examined the relationship between speech production and mispronunciation detection ability in children with speech delay (SD). Thirty-three SD children aged between 6;0 and 10;0 participated in a mispronunciation detection test using three types of stimuli: words pronounced correctly by other unfamiliar children (OTHERS); words mispronounced by OTHERS; and the participant’s own speech (SELF) pronounced either correctly or incorrectly. The participant’s articulation was assessed by the standardized GFTA-2 scores. Results indicated that SD children made significantly more errors when judging SELF speech than when judging OTHERS speech. Multiple regression analyses revealed that accuracy of detecting OTHERS mispronounced words was a significant predictor of GFTA-2 scores in these SD children. Interestingly, in the regression model, accuracy for detecting SELF mispronunciations made a significant independent contribution in addition to accuracy at detecting OTHERS mispronunciations. Overall these two measures accounted for a significant proportion of the variance in GFTA-2 scores (R-squared = 0.45). These findings suggest that children with SD may have more coarse phonological representations of their own speech than the speech of other children.

2aSC19. An EMA-based articulatory feedback approach to facilitate L2 speech production learning. Atsuo Suemitsu (Japan Adv. Inst. of Sci. and Technol., 1-1 Asahidai, Nomi 9231292, Japan, sue@jaist.ac.jp), Takayuki Ito, and Mark Tiede (Haskins Lab., New Haven, CT)

When acquiring a second language (L2), learners have difficulty in achieving native-like production even if they receive instruction on how to position the speech articulators for correct production. A principal reason is that learners lack information on how to modify their articulation to produce correct L2 sounds. A visual feedback method using electromagnetic articulography (EMA) has been previously implemented for this application with some success [Levitt *et al*. (2010)]. However, because this approach provided tongue tip position only, it is unsuitable for vowels and many consonants. In this work, we have developed a more general EMA-based articulatory feedback system that provides real-time visual feedback of multiple head movement-corrected sensor positions, together with target articulatory positions specific to each learner. We have used this system to improve the production of the unfamiliar vowel /ae/ for Japanese learners of American English. We predicted an appropriate speaker-specific /ae/ position for each Japanese learner using a model trained on previously collected kinematic data from 49 native speakers of American English, based on vowel positions for the overlapping /iy/, /aa/, and /uw/ vowels found in both languages. Results comparing formants pre- and post-feedback training will be presented to show the efficacy of the approach.

2aSC20. Articulatory phonetics of coronal stops in monolingual and simultaneous bilingual speakers of Canadian French and English. Francois-Xavier Brajot, Fateme Mollaei (School of Commun. Sci. and Disord., McGill Univ., 1266 des Pins Ouest, Montreal, QC H3G 1A8, Canada, fx.brajot@mail.mcgill.ca), Megan Callahan (Ctr. for Res. on Brain, Lang. and Music, McGill Univ., Montreal, QC, Canada), Denise Klein (Cognit. Neurosci. Unit, Montreal Neurological Inst., Montreal, QC, Canada), Shari R. Baum, and Vincent L. Gracco (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada)

Previous studies of bilingual speech production have relied on individuals whose age of acquisition of their second language varies. In the proposed research, we take advantage of the unique multilingual environment of Montreal and examine speech production in individuals who have acquired two languages from birth and compare the results to monolingual speakers. Electromagnetic recordings of single-word productions were carried out on three groups of female Canadian speakers (French monolingual, English monolingual, French-English simultaneous bilingual). Spectral moment and formant transition analyses of coronal burst segments showed cross-linguistic differences across vowel contexts. Tongue place of articulation and shape likewise indicated cross-linguistic differences in static articulatory positions. Kinematic analyses further identified language-specific movement patterns that helped clarify certain results from the acoustic analyses, namely that spatiotemporal characteristics of coronal articulation help enhance vocalic dimensions important to the respective language. Similar patterns were observed among the bilingual subjects, with the notable exception that acoustic and kinematic spaces narrowed considerably, resulting in reduced overlap between languages. It appears that simultaneous bilingual speakers not only follow language-appropriate articulatory and acoustic patterns, but further minimize areas of cross-linguistic convergence otherwise found among monolingual speakers.

2aSC21. Vowel production in sighted adults and blind adults: A study of speech adaptation strategies in high-intensity background noise. Pamela Trudeau-Fisette, Christine Turgeon, and Dominique Côté (Département de Linguistique, Université du Québec à Montréal, 405, rue Sainte-Catherine Est, Montréal, QC H2L 2C4, Canada, trudeau-fisette.pamela@courrier.uqam.ca)

Recent studies have shown that congenitally blind speakers have greater auditory discrimination acuity than sighted speakers [Ménard, Dupont, Baum, and Aubin, J. Acoust. Soc. Am. **126**, 1404–1414 (2009)]. At the production level, however, blind speakers produce smaller displacements of the lips (visible articulator) than their sighted peers. In order to further investigate the impact of visual experience on the articulatory gestures used to produce intelligible speech, adaptation strategies in background noise was studied in blind and sighted speakers. Ten sighted and 10 congenitally blind adult French participants were recorded during the production of the vowels /i/, /y/, /u/, /a/ in a CVC context. Two conditions were elicited: with high-intensity noise heard through headphones and without noise. Synchronous acoustic and articulatory data were recorded using the Carstens AG500 Electromagnetic Articulograph system. Formant measures and movements of the lips and tongue were analyzed. Results reveal that blind speakers produced smaller ranges of lip movement than sighted speakers in the noisy condition, suggesting that the blind subjects made less use of visible articulators to improve intelligibility. Results are discussed in light of multimodal production-perception relationships in speech.

2aSC22. Token-to-token variability and anticipatory coarticulation as indicators of maturity of speech motor control in 4-year-old children. Guillaume Barbier, Pascal Perrier (Speech and Cognition Dept., GIPSA-lab, 11, rue des Mathématiques, Saint Martin d’Hères 38402, France, guillaume.barbier@gipsa-lab.grenoble-inp.fr), Lucie Ménard (Laboratoire de Phonetique, UQAM, Montréal, QC, Canada), Mark Tiede (Haskins Lab., New Haven, CT), and Joseph S. Perkell (Massachusetts Inst. of Technol., Cambridge, MA)

Children’s gestures do not appear to be executed with the same dexterity as adults’. Studies of arm movements have shown that young children’s gestures are less accurate, more variable and slower than those of adults. This difference in behavior can be explained by a lack of experience with the sensory consequences of motor acts and still-developing forward models for the control of those acts. The hypothesis of immature and incomplete sensori-motor representations for speech in 4-year-old native speakers of Canadian French is addressed here through the analysis of ultrasound recordings of tongue contour kinematics and the speech signal from a corpus of isolated vowels and vowel-consonant-vowel sequences. Special attention is devoted to the analysis of vowel variability from two perspectives. Variability across repetitions in a single context provides information about the accuracy of the control. Contextual variability provides insights into the planning process as reflected in anticipatory coarticulation. Analysis of the observed lingual gestures will lead to improved understanding of the development of speech motor control and refinement of sensori-motor representations of speech. [Work supported by FQRNT project N° 147877 and ANR project ANR-08-BLAN-0272.]

2aSC23. Developmental aspects of American English diphthong trajectories in the formant space. Sungbok Lee (Elec. Eng., Univ. of Southern California, 3601 Watt Way, GFS-301, Los Angeles, CA 90089, sungbokl@usc.edu), Alexandros Potamianos (Electron. and Comput. Eng., Tech. Univ. of Crete, Chania, Greece), and Shrikanth Narayanan (Elec. Eng., Univ. of Southern Califronia, Los Angeles, CA)

Formant trajectories of five American English diphthongs embedded in the target words BAIT (/EY/), BITE (/AY/), POUT (/AW/), BOAT (/OU/), BOYS (/OY/) are investigated in the first two formant space as a function of age and gender. Age range considered is from 5 to 18 years. In this report, the focus is given on the differences in position between the start/end points of diphthongs and nine monophthongs. Averaged formant data across subjects in each age group are examined for this purpose. Two findings are worth mentioning. First, across all age groups, the start and end positions of diphthongs hardly match with the monophthongs that are typically used to transcribe the diphthongs across all age groups [cf. Holbrook and Fairbanks (1962)]. For instance, the start position of /EY/ is closer to /I/ than to /e/, and the end points of /EY, AY, OY/ are significantly different with respect to each other. Second, in addition to the larger size of vowel space, an overshoot trend toward the nominal end points of diphthongs is the most prominent developmental trend. That is, formant values of diphthongs produced by younger age children are closer to the nominal monophthongs used to transcribe the diphthongs.

2aSC24. What you see is what you hear: How visual prosody affects artificial language learning in adults and children. Jaspal K. Brar (Psychology, Univ. of Toronto, 13 Dovehaven Cres, Brampton, ON L6P 2N8, Canada, pauly.brar@mail.utoronto.ca), Michael D. Tyler (Psychology, Marcs Inst., Univ. of Western Sydney, Sydney, NSW, Australia), and Elizabeth K. Johnson (Psychology, Univ. of Toronto, Mississauga, ON, Canada)

Speech perception is a multimodal phenomenon, with what we see impacting what we hear. In this study, we examine how visual information impacts English listeners' segmentation of words from an artificial language containing no cues to word boundaries other than the transitional probabilities (TPs) between syllables. Participants ($N = 60$) were assigned to one of three conditions: Still (still image), trochaic (image loomed toward the listener at syllable onsets), or Iambic (image loomed toward the listener at syllable offsets). Participants also heard either an easy or difficult variant of the language. Importantly, both languages lacked auditory prosody. Overall performance in a 2AFC test was better in the easy (67%) than difficult language (57%). In addition, across languages, listeners performed best in the trochaic condition (67%) and worst in the Iambic condition (56%). Performance in the still condition fell in between (61%). We know English listeners perceive strong syllables as word onsets. Thus, participants likely found the Trochaic Condition easiest because the moving image led them to perceive temporally co-occurring syllables as strong. We are currently testing 6-year-olds ($N = 25$) with these materials. Thus far, children's performance collapsed across conditions is similar to adults (60%). However, visual information may impact children's performance less.

2aSC25. What palatalized consonants can tell us about theories of loanword adaptation. Allan J. Schwade (Linguistics, UCSC, 401 Pacific Ave. Apt 227, Santa Cruz, CA 95060, allanschwade@gmail.com)

Phonology- and perception-based theories of loanword adaptation clash over two different assumptions: what language background the adapter has and what cognitive component handles adaptation. Phonology-based theories argue that borrowers know both the source and borrowing language and that the phonology determines output forms; perception-based accounts argue that the borrower does not know the source language and that the phonetic decoder guides adaptation. Since there is no reason to believe that either population of borrowers cannot adapt words, a production experiment was carried out to test both populations/approaches. Four monolingual English and three bilingual English-Russian speakers were played currently unborrowed Russian words containing palatalized consonants and asked to repeat them aloud in an American English accent. Since palatalized velar and coronal stops are often articulated with some degree of affrication and monolinguals are unaware of this, it was predicted that they would sometimes adapt said consonants as affricates ($t^ju > t\int u$). However, since bilinguals are familiar with the co-articulatory affrication, they were not predicted to adapt palatalized stops as affricates ($t^ju > tu$). The results corroborated the hypothesis in that bilinguals never affricated while monolinguals affricated a tenth of palatalized stops—demonstrating that both theories make the correct predictions for their respective populations.

2aSC26. A developmental study of vowels spoken in syllables and in sentence context. Daniel J. Hubbard (School of Behavioral and Brain Sci., GR4.1, Univ. of Texas at Dallas, P.O. Box 830688, Richardson, TX 75083, dhubbard@utdallas.edu), Michael Kiefte (School of Human Commun. Disord., Dalhousie Univ., Halifax, NS, Canada), Shaikat Hossain, and Peter F. Assmann (School of Behavioral and Brain Sci., GR4.1, Univ. of Texas at Dallas, Richardson, TX)

This study examined the effects of context on the production of vowels spoken by children of different ages in isolation and in a carrier sentence. Vowels were extracted from a database of hVd syllables produced by 207 native English talkers from the North Texas region, ranging in age from 5 to 18 years with approximately equal numbers of males and females. Preliminary analysis of a subset of the database (around 25% of talkers) showed a systematic reduction in vowel duration with increasing age for syllables in isolation. Vowels in sentence context were on average 30% shorter than in isolated syllables, and durations were less closely linked to age group. Formant frequencies (F1–F3) showed similar patterns for vowels in isolated syllables and sentences, and decreased as a function of age as expected. However, measures of formant movement across the vowel (from 20 to 80% of the vowel duration) revealed increased F1 and F2 movement for syllables in isolation compared to those produced in carrier sentences. A comprehensive analysis of the database will be presented and implications for vowel recognition will be discussed.

2aSC27. An ultrasound study of the acquisition of North American English /ɹ/. Lyra V. Magloughlin (Dept. of Linguistics, Univ. of Ottawa, 70 Laurier Ave. East, Rm. 401, Ottawa, ON K1N 6N5, Canada, lyra@uottawa.ca)

I report an acoustic and articulatory study of North American English /ɹ/ production in typically developing English-speaking children during early and later-stage acquisition. North American English /ɹ/ is of interest in adult populations because it exhibits acoustic stability (e.g., low F3) despite considerable articulatory variability both within and between speakers [Delattre and Freeman (1968)]. North American English /ɹ/ is also often one of the last sounds to be acquired by children [Smit (1993), Schriberg (1993)], especially in prevocalic position (Smit *et al*. (1990), McGowan *et al*. (2004)]. Tiede *et al*. (2011) have argued that children might attempt different vocal tract configurations during acquisition, particularly in contexts where the articulatory demands are greater. While there is a growing body of literature on articulatory variability in adult production of /ɹ/ [e.g., Mielke *et al*. (2010), Campbell *et al*. (2011)], there remains virtually no articulatory data on typically developing children's production during acquisition. This study uses ultrasound imaging to investigate the articulations of four typically developing English-speaking children, aged between 3 and 6 years, during production of familiar lexical items. Children's early-stage articulations are examined and compared with their later-stage productions, and with adult variability patterns.

2aSC28. Six- and ten-month-old infants' perception of non-contrastive variation. Dena Krieger and Elizabeth K. Johnson (Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, dena.krieger@mail.utoronto.ca)

Recent evidence suggests that infants do not perceive all existing speech sounds from birth. For example, the velar and alveolar nasal place contrasts are so subtle that infants require experience to perceive it [e.g., Narayan *et al*. (2012)]. Here, we examine English-learning infants' perception of another subtle contrast: pre-voicing on stop consonants. Six- and 10-month-olds' ability to discriminate between voiced and voiceless stops (phonemically contrastive in English) as well as voiced and pre-voiced stops (allophonic in English, but contrastive in other languages such as Dutch) was tested using a variant of the stimulus alternation paradigm (SAPP). Six-month-olds ($N = 34$) distinguished between voiced and voiceless stops ($p < 0.05$), but not between voiced and pre-voiced stops. Ten-month-olds ($N = 32$) failed to discriminate either contrast. We conclude that (1) English pre-voicing may be a subtle contrast requiring experience to perceive, and (2) this version of the SAPP might not be an ideal methodology to examine discrimination abilities in 10-month-olds. Overall, our findings thus far fit well with the notion that some contrasts require experience to perceive, as well as with past studies reporting mixed results regarding English-learning infants' ability to perceive pre-voicing contrasts [e.g., Aslin *et al*. (1981), Lasky *et al*. (1975)].

2aSC29. Acoustical cues versus top-down bias in infants' parsing. Mireille Babineau and Rushen Shi (Psychology, Université du Québec à Montréal, Département de Psychologie, Université du Québec à Montréal, C.P. 8888 succursale Centre-ville, Montreal, QC H3C 3P8, Canada, babineau.mireille@courrier.uqam.ca)

French liaison involves the surfacing of an underlying consonant as the onset of the following vowel-initial word (e.g., les amis - /le/ /zami /), creating misalignment. However, acoustic cues that support vowel-initial parsing may exist. We tested French-learning 30-month-olds using a preferential looking procedure. Familiarization stimuli in experiment 1 were sentences each containing a determiner preceding a vowel-initial non-word (e.g., ces onches). Infants' parsing of the non-word was assessed. The vowel-initial condition presented the vowel-initial non-word versus another non-target (onches - èque). The syllabic condition tested the consonant-initial parse (zonches - zèque). Infants in the vowel-initial, but not the syllabic condition, showed discrimination ($p = 0.008$), i.e., they correctly parsed the vowel-initial target, possibly using acoustic cues. However, knowledge of underlying liaison consonants can also explain these results. In experiment 2, we removed acoustic cues to vowel-initial parsing by using a consonant-initial non-word following a determiner as the familiarization stimuli (e.g., un zonches). Infants were tested with the same two conditions as in experiment 1. Infants yielded the same results as in experiment 1, showing discrimination only in the vowel-initial condition ($p = 0.047$). Taken together, 30-month-olds perceived /z/ as an underlying floating element; they used this liaison knowledge, rather than possible acoustical cues, for parsing.

2aSC30. Effect of talker sex on infants' detection of spondee words in a two-talker or a speech-shaped noise masker. Lori Leibold, Crystal Taylor, Andrea Hillock-Dunn (Allied Health Sci., UNC Chapel Hill, 3122 Bondurant Hall, CB#7190, Chapel Hill, NC 27599, leibold@med.unc.edu), and Emily Buss (Otolaryngol.—Head and Neck Surgery, UNC Chapel Hill, Chapel Hill, NC)

Speech recognition performance in the presence of competing speech is typically better for adults when the target and masker talkers are different sexes than when the target and masker talkers are the same sex. One explanation for this result is that the acoustic differences between male and female speech productions promote segregation of the two streams of speech, thus leading to a reduction in informational masking. In this study, an observer-based psychophysical procedure was used to compare infants' (7–13 months) masked speech detection thresholds for spondee words produced by a male or a female talker in either a two-female-talker or a speech-shaped noise masker. Infants were assigned to a single testing condition. Maskers were presented continuously throughout testing at an overall level of 50 dB SPL, fixed throughout testing. Following training to an 80%-correct criterion, thresholds for the target word were measured adaptively using a 2-down, 1-up procedure. Infants' thresholds in the two-female-talker masker were higher for the female compared to the male target word. In contrast, infants' thresholds were similar for the female and male target words in the speech-shaped noise masker. These results suggest that introducing a different sex between the target and masker aids in the segregation of sounds for infants, as has previously been shown for adults. [Work supported by the NIH.]

2aSC31. The effects of voicing and position in infants' perception of coda consonants. Kathleen Engel (Speech Development Lab, Psychology, Univ. of Calgary, 2500 University Dr. NW, Calgary, AB T2N 1N4, Canada, knengel@gmail.com), Stephanie L. Archer (School of Commun. Sci. and Disord., McGill Univ., Montreal, Alberta, Canada), and Suzanne Curtin (Psychology, Univ. of Calgary, Calgary, AB, Canada)

Infants' ability to discriminate contrasting acoustic information has been demonstrated with many of the speech contrasts found in the world's languages. However, this ability seems to be positionally constrained. Contrasts in onsets are discriminated by young infants, but coda contrasts are not discriminated until around 16- to 20-months [Zamuner (2006)]. Here we examine whether the contrast and the position influence discrimination in younger infants. We tested 64 12-month-olds' discrimination of voiceless (/p/, /k/), or voiced stops (/b/, /g/) in either word-final (VC; Exp. 1) or word-medial (VCCV; Exp. 2) position. Experiment 1 habituated infants to ap or ak (voiceless) or to ab or ag (voiced). At test, infants heard the same token presented during habituation (same) and a novel token (switch). Planned comparisons revealed that only infants in the voiced condition looked longer to the switch than the same trial ($p < 0.05$). In experiment 2, infants heard apta or akta (voiceless) or abta or agta (voiced), but no effects of trial type were found, suggesting that the perceptual advantage in word-final position does not exist in word-medially. Thus, in word-final coda position voiced stops are more acoustically salient than voiceless stops, providing infants with added information to aid in discrimination.

2aSC32. Preliminary comparison of second-formant discrimination thresholds in cochlear implant users and young normal-hearing listeners. Catherine L. Rogers, Gail S. Donaldson, Amanda J. Cooley, and Benjamin A. Russell (Dept. of Commun. Sci. and Disord., Univ. of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

Formant discrimination thresholds (FDTs) may provide insights regarding factors limiting vowel perception by cochlear implant (CI) users, but have not been systematically studied in this population. In the present study, estimates of second-formant (F2) FDTs obtained in three CI users were compared to FDTs obtained from three young normal-hearing (YNH) listeners. Procedures and stimuli were modeled after Kewley-Port and Watson [J. Acoust. Soc. Am. **95**, 485–496 (1994)] but employed fewer trials and an expanded F2 frequency range. Stimuli were formant-synthesized versions of three target vowels. FDTs were estimated using an adaptive 3AFC task with feedback and based on six consecutive 80-trial stimulus blocks. FDTs for the three YNH listeners were comparable to previously reported FDTs (2.4% of reference frequency versus 1.5% in Kewley-Port and Watson). FDTs for two of the CI users were about 70% larger than the average for the YNH listeners. FDTs for the third CI user approached YNH average values in one frequency region but were enlarged in another region. Data for this CI user could not be explained by place-pitch thresholds (obtained in a previous study) and suggest that CI users' spectral acuity for complex stimuli may not be directly predictable from measures of spectral acuity for simple stimuli.

2aSC33. Toddlers' comprehension of noise-vocoded speech and sine-wave analogs to speech. Rochelle S. Newman (Dept. Hearing & Speech Sci., Univ. of Maryland, 0100 Lefrak Hall, College Park, MD 20742, rnewman1@umd.edu), Monita Chatterjee (Auditory Prostheses & Percept. Lab., Boys Town National Res. Hospital, Omaha, NE), Giovanna Morini, and Molly Nasuta (Dept. Hearing & Speech Sci., Univ. of Maryland, College Park, MD)

A great deal of research has investigated listeners' ability to compensate for degraded speech signals such as noise-vocoded speech (a signal with reduced spectral structure but intact amplitude envelope information) and sine-wave analogs to speech (a signal that maintains the global dynamic spectral structure of the signal at the expense of amplitude envelope information). Nittrouer and colleagues found developmental changes in the ability to comprehend such signals, reporting that while adults perform more accurately with sine-wave analogs than with noise-vocoded speech, school-aged children show the opposite pattern [e.g., Nittrouer Lowenstein and Packer (2009)]. In a series of studies, we tested toddler's comprehension of these degraded signals. Twenty-seven-month-old children saw two images on each trial (e.g., cat, dog), and heard a voice instructing them which image to look at ("Find the cat!"). Sentences were presented either in full speech or were degraded. Toddlers ($n = 24$ per condition) looked at the appropriate object equally long with vocoded speech of 24 channels (60.2%) or 8 channels (62.4%) as with full speech (62.6%), but performed barely above chance with 4 channels (53.6%) and at chance for 2 channels (49.8%). Preliminary results suggest that performance with sine-wave analogs is poorer than 8-channel vocoded speech (56.1%), but testing is ongoing.

2aSC34. Exploring auditory aging can exclusively explain Japanese adults' age-related decrease in training effects of American English /r/-/l/. Rieko Kubo and Masato Akagi (School of Information Sci., JAIST, 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan, rkubo@jaist.ac.jp)

Age-related decrease in training effect was shown by training of American English /r/-/l/ contrasts on Japanese speakers. This study examined whether the decrease can be explained exclusively by auditory aging, or other, compensatory cognitive processing should be taken into account. Japanese speakers aged 60's participated the experiment. Hearing threshold and spoken word perception test of participants' first language were used to estimate their auditory aging. The word perception test was composed of low-familiar words, high-familiar words, and mono syllables. The audiograms showed low threshold at high frequencies. The result of the perception test showed that low intelligibility for phonemes with high frequency or short duration, and confusion between contracted sounds and basic sounds. These were particular for low-familiar words and mono syllables. These results suggest that participants had auditory aging emerging as high frequency loss and time-frequency-resolution degradation. Nonetheless, the acoustic features to distinguish /r/ and /l/ have long duration, low frequencies and wide frequency distance which are supposed to be unaffected by these auditory aging. The effect of word familiarity suggested compensatory cognitive processing involved. These suggest that age-related decrease cannot be explained exclusively by auditory aging, compensatory cognitive processing should be taken into account.

2aSC35. Listener judgments of age in a single-talker 48-year longitudinal sample. Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu), Eric J. Hunter (National Ctr. for Voice and Speech, Univ. of Utah, Salt Lake City, UT), Catherine A. Mellum, and Lydia R. Rogers (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT, UT)

Numerous studies have demonstrated that listeners can make relatively accurate judgments of a talker's age from hearing the talker's voice. Materials in these previous studies have included sustained vowels (phonated and sometimes whispered), sentences, and passages of discourse. The number of talkers has ranged from 4 to 150, but in nearly all cases talkers were recorded only once. In the present study, the materials were recorded from a single talker who gave regular public speeches over a period of 48 years. Young adult listeners performed age judgments on samples extracted from 20 speeches chosen at 2–3 year intervals spanning the 48-year period, three samples per speech. Samples lasted 5 to 10 s and were chosen to minimize content that would identify the talker or link samples from the same speech to each other. In separate experiments, listeners listened to these 60 samples and after each one judged the talker's age either by choosing from three categories (50–66, 67–83, or 94–100 years) or by making a direct age estimate. Accuracy of these estimates will be compared to previous studies and examined as a function of both the talker's chronological age and acoustic measures performed for this talker in a separate experiment.

2aSC36. Examining the relationship between the interpretation of age and gender across languages. Andrew R. Plummer (Linguistics, Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, plummer@ling.ohio-state.edu), Benjamin Munson (Speech-Lang.-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Lucie Ménard (Linguistics, Univ. of Québec at Montréal, Montréal, QC, Canada), and Mary E. Beckman (Linguistics, Ohio State Univ., Columbus, OH)

Speech signals vary substantially in a number of their key properties, with the variability deriving from, among other things, talkers' age and gender. Speech processing requires resolution of this variation, necessitating interpretation of age and gender information in the signal. In some signals, the age and gender are not clear from acoustic information alone. In these cases, there may be substantial individual variation in judgments of age and gender. This study examined the interplay between the interpretation of age and gender across language communities. Corner vowel stimuli ([i], [u], [a]) generated by an age-varying articulatory synthesizer set at seven different ages (6 months, 2, 4, 5, 10, 16, and 21 years) were presented to native speakers of Cantonese, English, and Japanese. Listeners assigned an age (in years) and a gender (along a visual analog scale ranging from "definitely male" to "definitely female," or the equivalent in Japanese or Cantonese) to each stimulus. Analysis revealed a bifurcation in the interpretation of age and gender for the age 10 stimuli, which subjects rated as either a younger male or older female, suggesting a nonuniformity in the resolution of variability during speech processing. Preliminary analysis further suggests that this nonuniformity may be culture-specific.

2aSC37. Effects of vocal training on voluntary responses to pitch-shifted voice auditory feedback. Sona Patel, Cristina Nishimura, and Charles Larson (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, sona.patel@northwestern.edu)

Control of voice fundamental frequency (F0) relies on the interaction between various forms of sensory feedback and neural motor control mechanisms. Several studies have shown that unexpected changes in pitch in the auditory feedback lead to reflexive compensatory vocal responses. We examined voluntary vocal responses to unpredictable perturbations in pitch auditory feedback. Ten subjects were trained over a five-day period to change their voice F0 in the opposite direction to the pitch-shifted feedback ($\pm$100 cents, 1000 ms) and 10 in the same direction as the feedback. Results showed that responses that followed the stimulus direction had significantly shorter latencies (200 ms) than opposing responses (324 ms). The reduced latency of the following responses suggests a switch from a feedback to a feedforward control strategy. The feedback strategy requires monitoring feedback and correcting for errors between the feedback signal and the intended vocal goal. The feedforward strategy relies less on auditory feedback and more on an internal model of the desired vocal pitch goal. Furthermore, feedback systems generally are slower than feedforward strategies, which would explain the shorter latencies of the responses that followed the stimulus direction. Results of this study will be discussed in terms of the differing strategies that may be used in various vocal contexts.

2aSC38. Segment imitation in unfamiliar language varieties. Julia Forsberg (Philosophy, Linguist. and Theory of Sci., Univ. of Gothenburg, Box 200, Gothenburg 405 30, Sweden, julia.forsberg@gu.se)

Phonetic imitation has been explored in various research: in studies on impersonation [Zetterholm (2001); spontaneous phonetic imitation [Babel (2012)]; and as part of studies investigating overall imitation [Papcun (1988), Babel (2011)]. There are fewer studies focusing on specific phonemes, and the context required for successful imitation. Babel [**186** (2012)] (Am. English) and Zetterholm [**281** (1997)] (Swedish) found spontaneous imitation was more common in the open vowels than in close. A case study by Zetterholm [275 (1997)] of one impersonator shows formant measurements as closer to the target voice than that of the impersonator. This paper presents a comparative acoustic analysis of imitation of unfamiliar phonemes by untrained imitators, based on a small study containing recordings from SWEDIA and SUF. Vowel tokens are edited to three lengths with varying degrees of context. Listeners are asked to imitate the sound, and attempt to use it in a word. Acoustic analysis of F0, 1, 2, and 3 will be compared to the original recordings as well as the speakers' own speech. The conclusion and discussion includes indications of how easy it is to produce phonemes when they are not native to a speaker's own variant, as well as its relation to the forensic context.

2aSC39. Imitability of contextual vowel nasalization and interactions with lexical neighborhood density. Georgia Zellou (Linguistics, Univ. of Pennsylvania, 800 N. 48th St., #26, Philadelphia, PA 19139, gzellou@sas.upenn.edu), Rebecca Scarborough (Linguistics, Univ. of Colorado, Boulder, CO), and Kuniko Nielsen (Linguistics, Oakland Univ., Rochester, MI)

This study investigates the imitability of contextual vowel nasalization in English. Unlike other phonetic features reported to be imitable [e.g., vowel formants (Babel, 2012), VOT (Nielsen, 2011)], vowel nasality is non-contrastive in English. Nasality is, however, systematically variable: words from dense lexical neighborhoods (high-ND words) are produced with greater nasality than words from sparse neighborhoods [Scarborough (2004), (2012)]. Two experiments were conducted to test (1) whether (experimentally manipulated) nasality can be imitated in a shadowing task, and (2) whether direction of manipulation (more or less nasality, enhancing or countering natural neighborhood-conditioned patterns) affects shadowing behavior. Subjects shadowed 16 high-ND words (which are naturally more nasal) containing a vowel-nasal sequence and modified by spectral mixing to exhibit either

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greater-than-natural (experiment 1) or less-than-natural (experiment 2) nasality. Both the increase and the decrease in nasality were imitated (though not overall degree of nasality, as our imitation model was more nasal in both conditions than any of our subjects). This change persisted into a post-shadowing task for just the less-nasal condition. These results indicate that speakers are sensitive to non-contrastive phonetic detail in nasality, affecting their subsequent production. Further, the naturalness of nasality (reflecting neighborhood-conditioned variation) may affect the pattern of imitation.

2aSC40. Individual variability in phonetic convergence of vowels and rhythm. Gayatri Rao (Dept. of Psych., Univ. of Texas, 108 E. Dean Keeton, 1 University Station A8000, Austin, TX 78712, raog@utexas.edu), Rajka Smiljanic (Linguistics, Univ. of Texas, Austin, TX), and Randy Diehl (Dept. of Psych., Univ. of Texas, Austin, TX)

Phonetic convergence (PC) has been demonstrated for segmental (vowels, voice onset time) and suprasegmental (stress, intonation) properties [Nielsen (2008), Delvaux and Soquet (2007), Phillips and Clopper (2010), Rao, Smiljanic, and Diehl (2011)]. Since PC is subject to large individual differences [Ni Chiosáin (2007)], the current study examined individual variability in PC in both segmental and suprasegmental domains for native speakers of American English. Six female and six male pairs read CVC syllables and a short paragraph before and after an interactive map task. For each dyad, convergence in vowels was measured using formants and the cosine similarity metric for individual vowels and for the entire vowel space. Convergence in rhythm was measured using the centroid of the envelope modulation spectrum [EMS + centroid, Rao and Smiljanic (2011)]. Overall, speaker pairs converged to different extents in both measures. Vowel type, dialect background, and gender were found to influence the degree of convergence. In general, men were more likely to converge in rhythm whereas women were more likely to converge in vowels. This supports the findings that gender-based differences in convergence are due to perceptual sensitivity to indexical features [Nami *et al.* (2002), Babel (2009)] and particular sound features in spoken utterances.

2aSC41. What is the default behavior in accommodation: Convergence or maintenance? Bethany MacLeod (Carleton Univ., 211 Bruyere St., Ottawa, ON K1N 5E4, Canada, beth_macleod@carleton.ca)

This study, a secondary analysis of data from a phonetic accommodation study, considers the default behavior of speakers in accommodating to another speaker in an interaction. Should convergence or maintenance be considered the default behavior? There is inherent acoustic variation in our speech. Every time we produce a sound, such as a voiceless stop in English, it varies along some phonetic dimension, such as voice onset time (VOT). We might expect that, in the absence of any external influence, these voiceless stops will be realized with VOT longer than their overall mean 50% of the time and with VOT shorter than their overall mean 50% of the time. During interaction with another person, however, studies in social-psychology have suggested that lack of adjustment (maintenance) may be akin to divergence [Tong *et al.* (1999)]. In addition, convergence is a fairly robust finding in studies of accommodation and imitation [e.g., Nielsen (2011), Babel (2012)], suggesting that perhaps the default behavior in interaction is convergence. The purpose of this talk is to introduce these points of view, discuss the factors that may affect our interpretation, and facilitate discussion on this issue, which has implications for the growing body of research investigating accommodation.

2aSC42. Investigating developmental changes in phonological representation using the imitation paradigm. Kuniko Nielsen (Linguistics, Oakland Univ., 320 O'Dowd Hall, Rochester, MI 48309-4401, nielsen@oakland.edu)

This study investigates the developmental changes in phonological representation, by examining the word- and phoneme-level specificity of phonetic imitation by children. Prevailing linguistic theories assume three levels of phonological representations: word, phoneme, and feature. Previous research suggests that phonological representations develop throughout childhood, and that phonological awareness develops from larger to smaller units [e.g., Edwards *et al.* (2004), Treiman and Zukowski (1996)]. It has been shown that adult speakers implicitly imitate the phonetic properties of recently heard speech [e.g., Goldinger (1998)], and recently, Nielsen (2011) showed the sub-phonemic generalizability and word- and phoneme-level specificity of imitation, indicating that three levels of phonological representations simultaneously contribute to the observed patterns of phonetic imitation. In order to test whether young children manifest similar patterns of imitation and specificity, an experiment with a modified imitation paradigm with a picture-naming task was conducted, in which participants' VOT was compared before and after they were exposed to target speech with artificially increased VOT. Our preliminary results reveal that two groups of children (5 year-olds and 8 year-olds) show greater imitation than adults, while word-and phoneme-level specificity was greater for 8 year-olds than 5 year-olds. These results provide support for the continued development of phonological representations.

2aSC43. Phonetic accommodation in Spanish-English and Korean-English bilinguals. Stephen Tobin (Univ. of Connecticut, 406 Babbidge Rd., Storrs, CT 06269-1020, stephen.tobin@uconn.edu)

Preliminary results from eight participants in a cross-linguistic investigation of phonetic accommodation in speech production and perception are presented. The finding that synchronous actions are more stable than asynchronous ones has been reported in studies of general [Kelso (1981)] and speech-specific [Browman and Goldstein (1992), Byrd *et al.* (2009)] motor control. With reference to glottal-oral timing, near-zero VOTs (voice onset times) are representative of near-synchronous timing, whereas long-lag VOTs are representative of asynchronous timing [Sawashima and Hirose (1980), Dixit (1984), Lofqvist and Yoshioka (1989), Fuchs (2005)]. These observations served as a basis for the prediction that native speakers of Korean, with its long-lag aspirated stops (~120 ms), would more readily accommodate to typical English voiceless stop VOT (~70 ms) than native speakers of Spanish, with its short-lag voiceless stops (~20 ms). Spanish-English and Korean-English bilinguals were recorded reading voiceless stop-initial English words, before and during a task in which participants shadowed recorded productions of a native speaker of American English. Preliminary analysis of the production data provides some support for these hypotheses. The results contribute to our understanding of the conditions that promote phonetic accommodation.

2aSC44. The vowel spaces of Southern Californian English and Mexican Spanish as produced by monolinguals and bilinguals. Criccely Grijalva, Page E. Piccinini (Linguistics, Univ. of California San Diego, 3425 Lebon Dr. #918, San Diego, CA 92122, crgrijal@ucsd.edu), and Amalia Arvaniti (English Lang. & Linguist., Univ. of Kent, Kent, United Kingdom)

The vowel spaces of Southern Californian English and Mexican Spanish were investigated using three groups of speakers: 11 English monolinguals (8 females), 11 Spanish monolinguals (9 females), and 10 Spanish-English bilinguals (7 females). Speakers produced six repetitions of the ten American English vowels [i, ɪ, ɛ, æ, ɑ̃, ɔ, ʊ, u, ʌ, and ɝ] and six repetitions of the five Spanish vowels [i, e, a, u, and o]. Monolinguals produced vowels in one language; bilinguals produced vowels in both languages. Preliminary analysis shows Southern Californian English back vowels were less fronted compared to the results of Hagiwara (1997) from Southern Californian English, but more fronted than those of Hillenbrand *et al.* (1995) on General American English. Mexican Spanish back vowels [u] and [o] were substantially fronted compared to Castilian Spanish vowels [Bradlow (1995)], while [i] was lower and less fronted. In general, Mexican Spanish vowels were produced higher and more backed than Southern Californian English vowels in monolingual productions. Bilinguals produced their two vowel spaces closer together but with less dispersion than monolinguals, showing how bilinguals keep both language categories distinct.

2aSC45. Acoustic analysis of perceived accentedness in Mandarin speakers' second language production of Japanese. Peipei Wei and Kaori Idemaru (East Asian Lang. and Lit., Univ. of Oregon, 2250 Patterson St., 222, Eugene, OR 97405, peipei@uoregon.edu)

Many second language (L2) learners, particularly adult learners, retain foreign accent on their L2 production. What acoustic sources give rise to the perception of foreign accent? This study examines beginning and intermediate Chinese learners' production of Japanese in terms of segmental and suprasegmental features and investigates the relationship between acoustic characteristics of the L2 production and accentedness ratings provided by native Japanese listeners. Results of acoustic examination indicated that learners' production varied considerably from that of native speakers in terms of

durational features of stops, spectral features of some vowels, pitch (F0) peak alignment, and F0 contour. Multiple regression analysis identified the second formant of /u/, F0 peak alignment and contour as the strong predictors of perceived accent, accounting for nearly 90% of variance. These findings confirmed Flege's Speech Learning Model hypothesis—L2 sounds that are similar to L1 sounds, while subphonemically distinct, seem to pose greater difficulty for acquisition than dissimilar sounds. Moreover, longer classroom experience was found to show limited effects in reducing perceived accent, with slightly greater effects on segmental than suprasegmental variables.

2aSC46. Interaction of long-term acoustic experience and local context information on the perceptual accommodation of talker variability. Caicai Zhang (Haskins Lab., Yale Univ., The Chinese Univ. of Hong Kong, Hong Kong N/A, Hong Kong, yzcelia@gmail.com), Gang Peng, and William Shi-Yuan Wang (Dept. of Linguist. and Modern Lang., The Chinese Univ. of Hong Kong, Hong Kong, Hong Kong)

How do listeners recover speech content from acoustic signals, given the immense variability between talkers? In this study, two experiments were conducted on Cantonese level tones, comparing the perception of multi-talker speech stimuli in isolation and within a speech context. Without prior knowledge of a talker's pitch range, listeners resort to the population-average pitch range as a default reference for perception. This effect is attested by the significant correlation between the distance from population-average pitch range and identification accuracy in the isolation condition ($r = -0.24$, $p < 0.01$). The closer a talker's pitch range is to the population-average, the higher the identification accuracy is. The population-average reference is gender-specific, showing separate accommodation scales for female and male talkers. Such default reference is presumably built from one's long-term acoustic experience, reflecting the dense distribution of talkers in a community whose pitch is close to the population-average. Above the effect of long-term experience, the presence of a speech context allows listeners to tune to talker-specific pitch range, boosting the identification accuracy from 43% (in isolation) to 86%. Our findings demonstrate that listeners have built-in knowledge of population-average pitch and can shift from the default reference to talker-specific reference with the facilitation of context information.

2aSC47. Acoustic and articulatory information as joint factors coexisting in the context sequence model of speech production. Daniel Duran, Jagoda Bruni, and Grzegorz Dogil (Institut für Maschinelle Sprachverarbeitung (Inst. for Natural Lang. Process.), Stuttgart Univ., Pfaffenwaldring 5B, Stuttgart 70569, Germany, danielduran@ims.uni-stuttgart.de)

This simulation study presents the integration of an articulatory factor into the Context Sequence Model (CSM) [Wade *et al*. (2010)] of speech production using Polish sonorant data measured with the electromagnetic articulograph technology (EMA) [Mücke *et al*. (2010)]. Based on exemplar-theoretic assumptions [Pierrehumbert (2001)], the CSM models the speech production-perception loop operating on a sequential, detail-rich memory of previously processed speech utterance exemplars. Selection of an item for production is based on context matching, comparing the context of the currently produced utterance with the contexts of stored candidate items in memory. As demonstrated by Wade *et al*. (2010), the underlying exemplar weighing for speech production is based on about 0.5 s of preceding acoustic context and following linguistic match of the exemplars. We extended the CSM by incorporating articulatory information in parallel to the acoustic representation of the speech exemplars. Our study demonstrates that memorized raw articulatory information—movement habits of the speaker—can also be utilized during speech production. Successful incorporation of this factor shows that not only acoustic but also articulatory information can be made directly available in a speech production model.

2aSC48. Longitudinal changes of formant values and vowel space in two Mandarin-speaking children before 9 years of age. Li-mei Chen, Fan-Yin Cheng, and Wei Chen Hsu (Foreign Lang. & Lit., National Cheng Kung Univ., 1 University Rd., Tainan City 701, Taiwan, goodgoodgoodjob@yahoo.com)

Vowel productions of two Mandarin-speaking children were audio recorded in their homes with picture naming tasks once every 3 months, from birth to 9 years old. The present study is the ninth year of a longitudinal observation. Major findings in this stage are as follows: (1) The trend of decrease in formant values was continuously observed in the boy subject. As for the girl subject, it is not until 9 years old, the obvious decrease in formant values was found, especially in F1; (2) F1 values are more stable than F2 values in both subjects. They appeared to acquire jaw movement sooner than tongue movement; (3) Throughout these 9 years, the variability of F1 is around 200–300 Hz, and the variability of F2 is 500–700 Hz in both subjects. No trend of decrease was found; (4) The trend of shrinkage in F1-F2 vowel area continues from 7 to 9 years old for the boy subject, but not for the girl subject; (5) There is a clear decline in fundamental frequencies at 8-9 years of age in the boy subject. Longitudinal data of vowel formant values from the same group of subjects provide important references for assessment and treatment of articulation disorders in children.

2aSC49. Is the mora rhythm of Japanese more strongly observed in infant-directed speech than in adult-directed speech? Keiichi Tajima (Dept. of Psych., Hosei Univ., 36-511, 77 Massachusetts Ave., Cambridge, MA02139, tajima@hosei.ac.jp), Kuniyoshi Tanaka, Andrew Martin, and Reiko Mazuka (Lab. for Lang. Development, RIKEN Brain Sci. Inst., Wako-shi, Saitama, Japan)

Japanese has traditionally been called "mora-timed," but studies have shown that this intuition is based not on durational tendencies but rather on phonological, structural factors in the language. Meanwhile, infant-directed speech (IDS) is said to "exaggerate" certain properties of adult-directed speech (ADS), including rhythm. If so, then it is possible that the mora rhythm of Japanese is more strongly observed in IDS than ADS. To investigate this possibility, the present study utilized the RIKEN Japanese Mother-Infant Conversation Corpus, which contains approximately 11 h of IDS by 22 mothers talking with their 18-to-24-month-old infants, and 3 h of ADS by the same mothers. Results from durational analyses showed that aspects of mora rhythm, such as the distinction between phonemically short and long vowels and singleton and geminate consonants, and the tendency toward isochrony of moras, were not greater in IDS than ADS. Mora duration in IDS was instead more variable, partly stemming from greater phrase-final lengthening and non-phonemic, emphatic lengthening. Results from structural analysis, however, showed that non-CV moras such as moraic nasals that characterize Japanese rhythm occurred more frequently in IDS than ADS. These results suggest that even in IDS, Japanese rhythm is manifested structurally, not durationally. [Work supported by JSPS.]

2aSC50. Voice-onset time in infant-directed speech over the first year and a half. Evamarie Cropsey (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Jessica Gamache (Dept. of Linguist., Michigan State Univ., East Lansing, MI), Tonya Bergeson (Dept. of Otolaryngol.-Head and Neck Surgery, Indiana Univ. School of Med., Indianapolis, IN), and Laura Dilley (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI 48824, ldilley@msu.edu)

Previous research in small-N studies has indicated conflicting findings regarding whether mothers modify voice-onset time (VOT) of word-initial stop consonants in speech to infants compared to speech to adults, as well as the nature of any such modification. In a large-scale study, VOT was measured for voiced and voiceless stop consonants in speech of 48 mothers of infants in one of four cross-sectional age groups (0;3, 0;9, 1;1, 1;8) when they read a phonetically controlled storybook to their infant (ID speech) or an adult (AD speech). VOT measurements showed enhanced clarity (i.e., longer VOTs) in ID speech compared with AD speech for voiceless stop consonants only. An effect of infant gender was also found, showing that enhanced clarity was only produced by mothers of female infants ($N = 19$). Infant age was not found to be a significant factor in VOT production. The results have implications for understanding the nature of linguistic development in young children, specifically by elucidating factors apparently related to phonetic modification for clarity, including speech style and gender. [Work supported by NIH-NIDCD grant R01DC008581.]

2aSC51. Pitch affects voice onset time: A cross-linguistic study. Chandan Narayan (Linguistics, Univ. of Toronto, Sidney Smith Hall, 4th Fl., 100 St. George St., Toronto, ON M5S 3G3, Canada, chandan.narayan@utoronto.ca) and Mary Bowden (Speech-Lang. Pathol., Univ. of Toronto, Toronto, ON, Canada)

Recent research into the acoustics of infant-directed speech (IDS) in English and Korean suggests that voice-onset time (normalized for varying speech rate) in word-initial aspirated stops are shorter than in adult-directed

speech. The present study reports results from experiments conducted to explore the source of this VOT variation in IDS. Female English speakers ($n = 10$) and female Korean speakers ($n = 10$) recorded sentences with words varying in laryngeal condition (English: voiced/voiceless; Korean: plain, tense, aspirated) at three different pitches (low, normal, and falsetto). Results suggest that as speakers' pitch increases, the duration of VOT in aspirated stops decreases. Voiced stops (in English) and plain and tense stops) in Korean showed no difference in VOT across varying pitch conditions. Regression models suggest that VOT becomes less predictive of laryngeal state as speaker pitch increases. Results are discussed in terms of the physiological explanation of the pitch-VOT effect as well as the implications for the development of sound systems in infants.

2aSC52. /oy/ as a marker of local identity in Berlin. Stefanie Jannedy (ZAS Berlin, Schützenstr. 18, Berlin 10117, Germany, jannedy@ling.ohio-state.edu) and Melanie Wierich (Friedrich-Schiller-Universitat, Jena, Germany)

A fairly recent observation of multi-cultural urban German speech as spoken in Berlin is that the diphthongs /oy/ and /ey/ are realized more closed and fronted compared to more standard varieties of German. For this pilot study, spontaneous speech data were collected through standardized interviews from five young female speakers from two different neighborhoods in Berlin: Wedding is more Arab-dominant while Kreuzberg is more Turkish dominant. Their speech was orthographically transcribed and added to a database that allows for searching for all occurrences of the two diphthongs under investigation in their naturally occurring context in unscripted speech. So far, 250 occurrences of these vowels have been analyzed. Formant measurements were taken at five equally distanced points throughout the diphthong. A linear mixed effects model with the midpoint of the F2-formant value as the dependent variable were run, showing that speakers from the arab neighborhood (Wedding) significantly differ in their productions compared to speakers from the Turkish neighborhood ($p < 0.05$) in Kreuzberg. Moreover, there was a significant effect of language spoken around them ($p < 0.01$) on the production even though German is their dominant language. We argue that speakers use the production of these diphthongs as markers of their local urban identity.

2aSC53. Articulatory compensation to second formant perturbations. Chris Neufeld (Oral Dynam. Lab., Speech-Lang. Pathol., Univ. of Toronto, Toronto, ON M5G 1V7, Canada, christopher.neufeld@mail.utoronto.ca), Pascal van Lieshout (Speech-Lang. Pathol., Univ. of Toronto, Toronto, ON, Canada), and David Purcell (School of Commun. Sci. and Disord., Univ. of Western Ontario, London, ON, Canada)

There is a fast-growing literature examining speakers' response to real-time alterations of auditory feedback. The majority of these studies examine the response of the subject in acoustic terms. Since many subjects fail to (acoustically) compensate for the perturbation, the current experiment examines whether there are systematic articulatory responses to formant perturbation in the absence of compensation at the level of acoustics. Articulatory data are collected using a 3D electro-magnetic-articulograph. F2 is gradually shifted up or down and preliminary results from three English-speaking subjects showed that two subjects show no response in their acoustics or articulation. However, the remaining speaker who did not show compensation at the level of acoustics displayed a systematic response in some articulatory variables. The acoustic effects of his response were masked because the other articulators behaved in a more variable way, making the second formant vary randomly from trial to trial. Based on these results we expect to see a spectrum of response patterns from a larger population of speakers, from total non-compensation in both acoustics and articulation, partial compensation in articulation, and global articulatory compensation, which induces the appropriate compensation at the level of acoustic output.

2aSC54. Does compensation in auditory feedback require attention? Agnes Alsius, Takashi Mitsuya (Psychology, Queen's Univ., 62 Arch st., Humphrey Hall, Kingston, ON K7L 3N6, Canada, aalsius@gmail.com), and Kevin G. Munhall (Psych. & Otolaryngol., Queen's Univ., Kingston, ON, Canada)

When speakers receive auditory feedback with a real-time perturbation of formant structure, they hear themselves produce a vowel slightly different form the one intended. In response, they spontaneously change the formant structure to make the feedback more consistent with the intended sound. This compensatory behavior was reported to be automatic [Munhall *et al.* (2009)] because speakers are not able to suppress it even when they are informed about the perturbation and are instructed not to change their articulation. However, whether and to which extent attentional resources are utilized for this behavior have not been directly investigated. In the current study, speakers performed a speech production task where they pronounced a monosyllable whose formant structure was perturbed, while concurrently performing another task (i.e., dual-task paradigm). The preliminary results showed that, when attention was diverted to an unrelated auditory detection task, the magnitude of compensation remained the same as in the single task condition. Follow-up experiments will manipulate the nature and difficulty of the concurrent task to examine whether compensation in speech production is affected, and if so, what levels of the error feedback system are more susceptible to attentional manipulations.

2aSC55. Speech sensorimotor learning through a virtual vocal tract. Jeffrey J. Berry (Speech Pathol. & Audiol., Marquette Univ., P.O. Box 1881, Milwaukee, WI 53201-1881, jeffrey.berry@marquette.edu), Cassandra North, Benjamin Meyers, and Michael T. Johnson (Elec. & Comput. Eng., Marquette Univ., Milwaukee, WI)

Studies of speech sensorimotor learning often manipulate auditory feedback by modifying isolated acoustic parameters such as formant frequency or fundamental frequency using near real-time resynthesis of a participant's speech. An alternative approach is to engage a participant in a total remapping of the sensorimotor working space using a virtual vocal tract. To support this approach for studying speech sensorimotor learning, we have developed a system to control an articulatory synthesizer using electromagnetic articulography data. Articulator movement data from the NDI Wave System are streamed to a Maeda articulatory synthesizer. The resulting synthesized speech provides auditory feedback to the participant. This approach allows the experimenter to generate novel articulatory-acoustic mappings. Moreover, the acoustic output of the synthesizer can be perturbed using acoustic resynthesis methods. Since no robust speech-acoustic signal is required from the participant, this system will allow for the study of sensorimotor learning in any individuals, even those with severe speech disorders. In the current work, we present preliminary results that demonstrate that typically functioning participants can use a virtual vocal tract to produce diphthongs within a novel articulatory-acoustic workspace. Once sufficient baseline performance is established, perturbations to auditory feedback (formant shifting) can elicit compensatory and adaptive articulatory responses.

2aSC56. Exploring production-perception relationships in normal hearing and cochlear implant adults: A lip-tube perturbation study. Christine Turgeon and Amélie Prémont (Linguistique, UQAM, 320 Ste-Catherine Est, Montréal, QC H2X 1L7, Canada, turgeon.christine.2@courrier.uqam.ca)

It has been claimed that auditory feedback mechanisms enable monitoring and calibration of feedforward commands in speech production. Therefore, lack of auditory feedback may interfere with adequate compensation strategies in perturbed situations. This study investigates the effect of hearing status and a lip tube perturbation on vowel production. Eleven normal-hearing controls, and seventeen cochlear implant (CI) users (7 prelingually, 10 postlingually) were recorded during the production of the vowel /u/. Acoustic analyses were conducted with and without a 15-mm-diam tube inserted between the lips. Recording sessions were also made before and after the perturbation, with and without auditory feedback. Deaf participants' auditory feedback was provided by the CI and interrupted by switching off their implant devices. Separate analyses were conducted on the first (F1), the second formant (F2), and the fundamental frequency (F0). Results revealed a main effect of group and an interaction between condition and hearing status. Together, results suggest that auditory feedback plays an important role in speech compensation.

2aSC57. Acoustic vowel space and speech rate in Mandarin-speaking children with cerebral palsy. Wei Chen Hsu, Li-mei Chen, and Fan-Yin Cheng (Foreign Lang. & Lit., National Cheng Kung Univ., 1 University Rd., Tainan City 701, Taiwan, leemmay@gmail.com)

This study examines the variability in speech production in four Mandarin-speaking children: two with cerebral palsy (CP) and two typically developing (TD) from 4 to 5 years of age. Recordings collected from the picture-naming task and spontaneous interaction with adults was analyzed.

Acoustic vowel space and speech rate in their production were investigated. Study findings indicated the following: (1) Due to defect in speech motor control, children with CP have a smaller overall vowel space than TD children; (2) In CP group, there are more variability of formant values of individual vowels and the vowel space of individual vowels thus overlap more; (3) There is a trend of decrease of vowel formant values in both TD and CP; (4) Children with CP tend to spend more time in speech production because of their impaired speech-motor control, in terms of syllable per minute and intelligible syllable per minute; (5) Slower speech rate seems to increase speech intelligibility in CP. However, this needs to be verified in further studies. Extended longitudinal observation can provide more complete profile of individual differences in the development of vowels and speech rate to verify these preliminary findings. The variability features in the production of children with CP provide important references in speech therapy.

TUESDAY MORNING, 4 JUNE 2013 — 510A, 9:00 A.M. TO 11:40 A.M.

Session 2aSP

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

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

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

Invited Papers

9:00

2aSP1. Spherical array processing with binaural sound reproduction for improved speech intelligibility. Noam R. Shabtai and Boaz Rafaely (Dept. of Elec. and Comput. Eng., Ben Gurion Univ. of the Negev, 17 Sheizaf St., Omer 84965, Israel, shabtai.noam@gmail.com)

In telecommunication applications, interfering sounds and reverberation can have a detrimental effect on speech intelligibility. For this reason, microphone arrays have been recently employed in telecommunication systems for natural environments. Currently applied array processing methods typically aim to produce array output, which is optimal on signal-based measures, e.g., signal-to-noise ratio (SNR). These measures may be particularly appropriate when the receiver is a machine. However, in order to enhance speech intelligibility when the receiver is another human, it may be desired to trigger spatial hearing capabilities of the human auditory system, such as the cocktail party effect. In particular, spatial-release from masking has been investigated. This work presents a spherical array signal processing framework in which array output is generated binaurally using the head-related transfer function. In this framework both target direction is enhanced and spatial information of all sources are perceived by the listener. The performance of the proposed binaural beamformer is compared to the performance of a non-binaural maximum directivity beamformer based on a spatial reproduction listening tests. The average percentage correct decision is calculated over five subjects and is shown to be higher when the binaural beamformer is used for every tested SNR.

9:20

2aSP2. Improvement of accuracy of 3D sound space synthesized by real-time "SENZI," a sound space information acquisition system using spherical array with numerous microphones. Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Satoshi Hongo (Dept. of Design and Comput. Applications, Sendai National College of Technol., Sendai, Japan), Takuma Okamoto (National Inst. of Information and Commun. Technol., Soraku-gun, Japan), Yukio Iwaya (Dept. of Elec. Eng. and Information Technol., Tohoku Gakuin Univ., Tagajo-shi, Japan), and Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan)

We proposed a sensing method of 3D sound-space information based on symmetrically and densely arranged microphones mounted on a solid sphere. We call this method SENZI [Sakamoto *et al*., ISUC2008 (2008)]. In SENZI, the sensed signals from each of the microphone is simply weighted and summed to synthesize a listener's HRTF, reflecting the listener's facing direction. Weighting coefficients are calculated for individual listeners based on their HRTFs. These coefficients are changed according to the listeners' head movement, which is known to provide important dynamic perceptual cue for sound localization. Therefore, accurate sound space information can be presented to unlimited number of listeners not only beyond the distance but also beyond the time. Recently, we realized this method as a real-time system using a 252-ch spherical microphone array and FPGAs. By using this system, accurate sound space information up to around 10 kHz can be synthesized to any listeners. However, the SNR of microphones affected to the accuracy of synthesized sound-space information, especially under low frequency region. To avoid the effect, we used condition numbers as an index to synthesize accurate sound-space information in the low frequency region.

9:40

2aSP3. Spatial audio coding with spaced microphone arrays for music recording and reproduction. Archontis Politis, Mikko-Ville Laitinen, and Ville Pulkki (Dept. of Signal Process. and Acoust., Aalto Univ., Otakaari 5A, Espoo 02150, Finland, archontis.politis@aalto.fi)

Spaced microphone arrays are commonly used in multichannel recording of music, due to their inherent quality of natural incoherence between the surround channels at reproduction, at the expense of accurate localization. Recent methods in parametric spatial audio coding, such as Directional Audio Coding, exploit coincident microphone patterns to extract directional information and reproduce it in a perceptually optimal way. In this study, we present how Directional Audio Coding can be adapted for spaced arrays, offering improved localization cues at reproduction, without compromising the qualities of the spaced microphone recording techniques. Examples are presented for some well-established array configurations.

10:00

2aSP4. Investigating perceptual attributes associated with reproduction of early reflections via virtually elevated loudspeaker. Sungyoung Kim (ECTET, Rochester Inst. of Technol., ENT-2151, 78 Lomb Memorial Dr., Rochester, NY 14623, sungyoungk@gmail.com)

Reproducing early reflections related to "height" information through elevated loudspeakers delivers enhanced presence of auditory images and integrates with a three-dimensional visual content homogeneously. Nonetheless, it is practically difficult for consumers to place loudspeakers required for the height-channel reproduction in a listening room. To overcome this limitation, many academic or commercial institutions propose various methods that render vertical sound images and reproduce them with smaller number of loudspeakers that are typically located in the horizontal plane. The rendered image then could deliver vertically extended impression of a sound field, which is likely related to listeners' perception of enhanced presence. To better understand this relationship, this paper investigated idiosyncratic difference between one surround sound field and another with early reflections that is virtually elevated. The elicitation result revealed that listeners used four salient attributes—ASW, LEV, Powerfulness, and Clarity—to describe the difference. The subsequent result showed that perceived magnitudes of those percepts were accounted for by a physical parameter, correlation coefficient between the elevated signal and the loudspeaker signal that is to feed to the closest loudspeaker in the horizontal plane.

Contributed Papers

10:20

2aSP5. Approximate convolution using partitioned truncated singular value decomposition filtering for binaural rendering. Joshua Atkins, Adam Strauss, and Chen Zhang (Beats Electron., LLC, 1431 Ocean Ave., Apt. 1018, Santa Monica, CA 90401, joshatkins@ieee.org)

In conventional binaural rendering, a pair of head-related impulse responses (HRIR), measured from source direction to left and right ears, is convolved with a source signal to create the impression of a virtual 3D sound source when played on headphones. It is well known that using HRIRs measured in a real room, which includes a natural reverberant decay, increases the externalization and realism of the simulation. However, the HRIR filter length in even a small room can be many thousands of taps, leading to computational complexity issues in real world implementations. We propose a new method, partitioned truncated singular value decomposition (PTSVD) filtering, for approximating the convolution by partitioning the HRIR filters in time, performing a singular value decomposition on the matrix of filter partitions, and choosing the N singular-vectors corresponding to the N largest singular values to reconstruct the HRIR filters. We will show how this can be implemented in an efficient filter-bank type structure with N tapped delay lines for real-time application. We also show how improvements to the method, such as modeling the direct path HRIR separately can lead to improved rendering at minimal computational load.

10:40

2aSP6. The influence of regularization on anechoic performance and robustness of sound zone methods. Philip Coleman, Philip Jackson, Marek Olik (Ctr. for Vision, Speech and Signal Process., Univ. of Surrey, Guildford GU2 7XH, United Kingdom, p.d.coleman@surrey.ac.uk), Martin Olsen, Martin Møller, and Jan Abildgaard Pedersen (Bang & Olufsen a/s, Struer, Denmark)

Recent attention to the problem of controlling multiple loudspeakers to create sound zones has been directed toward practical issues arising from system robustness concerns. In this study, the effects of regularization are analyzed for three representative sound zoning methods. Regularization governs the control effort required to drive the loudspeaker array, via a constraint in each optimization cost function. Simulations show that regularization has a significant effect on the sound zone performance, both under ideal anechoic conditions and when systematic errors are introduced between calculation of the source weights and their application to the system. Results are obtained for speed of sound variations and loudspeaker positioning errors with respect to the source weights calculated. Judicious selection of the regularization parameter is shown to be a primary concern for sound zone system designers—the acoustic contrast can be increased by up to 50 dB with proper regularization in the presence of errors. A frequency-dependent minimum regularization parameter is determined based on the conditioning of the matrix inverse. The regularization parameter can be further increased to improve performance depending on the control effort constraints, expected magnitude of errors, and desired sound field properties of the system.

11:00

2aSP7. Sound field planarity characterized by superdirective beamforming. Philip J. Jackson (CVSSP, Dept. of Electron. Eng., Univ. of Surrey, Guildford, Surrey GU2 7XH, United Kingdom, p.jackson@surrey.ac.uk), Finn Jacobsen (Acoust. Technol., Tech. Univ. of Denmark, Lyngby, Denmark), Philip D. Coleman (CVSSP, Univ. of Surrey, Guildford, United Kingdom), and Jan Abildgaard Pedersen (Acoustics, Bang & Olufsen a/s, Struer, Denmark)

The ability to replicate a plane wave represents an essential element of spatial sound field reproduction. In sound field synthesis, the desired field is often formulated as a plane wave and the error minimized; for other sound field control methods, the energy density or energy ratio is maximized. In all cases and further to the reproduction error, it is informative to characterize how planar the resultant sound field is. This paper presents a method for quantifying a region's acoustic planarity by superdirective beamforming with an array of microphones, which analyzes the azimuthal distribution of impinging waves and hence derives the planarity. Estimates are obtained for a variety of simulated sound field types, tested with respect to array orientation, wavenumber, and number of microphones. A range of microphone configurations is examined. Results are compared with delay-and-sum beamforming, which is equivalent to spatial Fourier decomposition. The superdirective beamformer provides better characterization of sound fields and is effective with a moderate number of omni-directional microphones over a broad frequency range. Practical investigation of planarity estimation in real sound fields is needed to demonstrate its validity as a physical sound field evaluation measure.

11:20

2aSP8. Analysis of pre-echo artifact in generating a focused source. Min-Ho Song (Grad. School of Culture Technol., KAIST, YuseongGu Gusung Dong 373-1, Daejeon 373-1, South Korea, godspd@kaist.ac.kr), Jung-Woo Choi, and Yang-Hann Kim (Ctr. for Noise and Vib. Control, Dept. of Mech. Eng., KAIST, Daejeon, South Korea)

A focused source is a virtual source that can provide an auditory illusion of sound radiating between the loudspeaker array and listener. When generating a focused source, a listener cannot avoid undesired pre-arriving direct waves from the control sources, which is known as pre-echo artifact. Investigation of the artifact can be seen in several researches; however, no mathematical definition of the pre-echo artifact is given so far and only the observation in limited case is known with computer simulation. The

objective of this paper is to observe the cause and effect of pre-echo artifact analytically. The paper defines the pre-echo artifact mathematically, and the artifact at the arbitrary listening position is formulated with integral form based on Kirchhoff-Helmholtz integral equation. From the definition of the pre-echo, it is shown that the convergent wave of a focused source can be regarded as a special case of a pre-echo artifact. Furthermore, the derivation shows that the pre-echo artifact occurs in the case of using continuous array and is evolved due to the time-reversed nature of the solution.

TUESDAY MORNING, 4 JUNE 2013 — 511AD, 9:00 A.M. TO 12:00 NOON

Session 2aUW

Underwater Acoustics and Acoustical Oceanography: Wave Propagation in a Random Medium

John A. Colosi, Chair
Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943

Contributed Papers

9:00

2aUW1. Time-varying three-dimensional mapping of internal waves during the Shallow Water 2006 experiment. Mohsen Badiey, Lin Wan, and Aijun Song (College of Earth, Ocean, and Environ., Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

Formation and propagation of internal waves have recently become of interest to ocean acousticians, since the propagation of internal waves in shallow water waveguide plays an important role in intensity fluctuations of acoustic signals [J. Acoust. Soc. Am. **112**(2), 747–760 (2007)]. Modeling the acoustic field in these regions requires detailed knowledge of sound speed and its spatial and temporal variability resulting from propagating internal waves. Although satellite imagery can provide snapshots of the surface impressions of the internal waves, due to low sampling in time (limited images in each orbit) other techniques to obtain the time varying, three-dimensional (3D) internal wave field are desirable. An example to obtain a time-varying, 3D internal wave field is presented in this paper. The internal wave fine structure is reconstructed using simultaneous measurement of temperature data using thermistor arrays and the surface impressions of a moving internal wave packet using a ship's radar during the Shallow Water 2006 experiment (SW06). The parameters of the internal wave train, such as wave speed, propagation direction, and amplitude of the first wave front, are determined. The resulting temperature field induced by the internal waves is used as environmental input to a 3D acoustic model to study the effects of internal wave on acoustic propagation. [Work supported by ONR322OA.]

9:20

2aUW2. The effect of surface and linear internal waves on higher order acoustic moments in shallow water. Kaustubha Raghukumar and John A. Colosi (Oceanography, Naval Postgrad. School, 315 B Spanagel Hall, 833 Dyer Rd., Monterey, CA 93943, kraghuku@nps.edu)

Acoustic fields in shallow water have a statistical nature due to complex, time-evolving sound speed fields, and scattering from rough boundaries. Previously, coupled-mode transport theory [Raghukumar and Colosi (2012)] was applied to high frequency acoustic fluctuations in an environment typical of the Shallow Water 2006 (SW06) experiment on the New Jersey continental shelf. As a consequence of the strong adiabatic component in SW06 propagation, a hybrid approach was used to calculate mode coherences where mode energies from the Dozier-Tappert approach were combined with adiabatic phase terms. Mode energies, coherences and acoustic intensities were examined, and it was found that internal and surface waves preferentially couple low and high modes respectively. Here, we extend that study to include higher moments such as scintillation index and shift focus to modes that are coupled by both internal and surface waves. Oceanographic and sea surface measurements are used to constrain the internal wave and sea surface models. The relative importance of linear internal waves and surface scattering effects are studied using transport theory and Monte Carlo simulations.

9:40

2aUW3. The effects of internal tides on phase and amplitude statistics in the Philippine Sea. John A. Colosi, Tarun Chandrayadula, Weston Coby, Jacob Fischer (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu), Brian D. Dushaw (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Moored oceanographic sensors and satellite altimetry has revealed energetic diurnal and semi-diurnal internal tides in the Western Philippine Sea. Because the internal tides have a complex spatio-temporal pattern and large vertical displacements, these waves have the potential for causing strong acoustic variability. This talk will present a tidal analysis of signal fluctuations from the PhilSea09 experiment in which broadband signals with a center frequency of 275 Hz and a bandwidth of 50 Hz were transmitted at the sound channel axis to a large aperture vertical array 180-km distant. Signal phase and amplitude statistics along distinct branches of the observed wavefronts will be analyzed and compared to ray-based model predictions using internal tide information obtained from moored oceanographic instruments at the source and receiver. Key issues are the acoustic effects of the internal tide nonlinearity, temporal stability, high mode structure, and complex horizontal interference patterns.

10:00

2aUW4. Comparison of transport theory predictions with measurements of the decrease in shallow water reverberation level as the sea state increases. Eric I. Thorsos, Jie Yang, W. T. Elam, Frank S. Henyey (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu), Fenghua Li, and Jianjun Liu (State Key Lab. of Acoust., Inst. of Acoust., Beijing, China)

Transport theory has been developed for modeling shallow water propagation and reverberation at mid frequencies (1–10 kHz) where forward scattering from a rough sea surface is taken into account in a computationally efficient manner. The method is based on a decomposition of the field in terms of unperturbed modes, and forward scattering at the sea surface leads to mode coupling that is treated with perturbation theory. Reverberation measurements made during ASIAEX in 2001 provide a useful test of transport theory predictions. Modeling indicates that the measured reverberation was dominated by bottom reverberation, and the reverberation level at 1 and 2 kHz was observed to decrease as the sea surface conditions increased from a low sea state to a higher sea state. This suggests that surface forward scattering was responsible for the change in reverberation level. By modeling the difference in reverberation as the sea state changes, the sensitivity to

environmental conditions other than the sea surface roughness is much reduced. Transport theory predictions for the reverberation difference are found to be in good agreement with measurements. [Work supported by the U.S. Office of Naval Research, Ocean Acoustics.]

10:20

2aUW5. Scales of time and space variability of sound fields reflecting obliquely from underwater slopes. Timothy Duda, Ying-Tsong Lin (Woods Hole Oceanogr. Inst., AOPE Dept MS 11, Woods Hole, MA 02543, tduda@whoi.edu), and Bruce D. Cornuelle (Scripps Inst. of Oceanogr., La Jolla, CA)

Ocean flows impart a time dependence to interference patterns of sound reflecting at low horizontal angle from sloping bathymetry. This is a higher level of complexity than interference patterns within vertical slices in spherically symmetric environments. The time-space statistics may be co-dependent and non-normal, making conventional methods inapplicable. Here, patterns that occur when 100 to 1000 Hz sound reflects at slopes are simulated with three-dimensional methods. Multiple statistics of sound at virtual arrays are computed over the domain, including incoherent power, beam power, spatial correlation length, and array gain assuming idealized noise conditions. These depend on source and receiver locations with respect to bathymetric features and can be computed for instantaneously sampled ocean conditions to study their evolution within the time frame of the dominant flow, or computed via averaging over a few periods of the dominant flow features (tides, for example). Here, the spatial complexity of patterns found in 100-Hz (and upward) simulations at a slope area off San Diego, CA, in a time varying flow are linked to the imposed seafloor roughness as well as geometry, with mean intensity, scintillation index, and correlation scale used to quantify the effect. Implications for statistics employed in detection and tracking algorithms are discussed.

10:40

2aUW6. The effect of random fluctuations in bottom bathymetry on acoustic coherence in shallow water. Harry A. DeFerrari and Jennifer Wylie (Appl. Marine Phys., Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, hdeferrari@rsmas.miami.edu)

The loss of temporal coherence after long range propagation in shallow water is often studied as a consequence of sound speed variability from internal waves. Here, we add the complication of small amplitude and very long wavelength random fluctuations of bottom bathymetry. It is shown that the same range dependent sound speed fluctuations result in markedly different coherence times depending on acoustic wavelength and mode number—a first order effect. A range dependent PE code (MMPE) is used to predict temporal coherence for individual surface reflected-bottom-reflected (SRBR) mode arrivals. Here, a mode coherence calculation is developed and compared for varying RMS bathymetry. Temporal coherence is inferred from mode coherence. We find first order and /or low frequency modes are insensitive to the bottom but when the (sine of the mode angle approaches 1/10 of an acoustic wavelength) the modes structure in amplitude and phase is randomized and the signal decorrelate rapidly in time from just the slightest temporal variations in sound speed. It does not take much; just 1 m in 200 m of range will randomize all but the first mode at mid frequencies (0.5 to 1 kHz). Predictions are in close agreement with SW06 mode coherence measurements.

11:00

2aUW7. Comparison of statistics of controlled source tones and single ship noise in the deep ocean. Brianne M. Baxa, Gerald L. D'Spain, Peter F. Worcester, Matthew A. Dzieciuch (Scripps Institution of Oceanography, UC San Diego, 2498 Manzana Way, San Diego, CA 92139, bmoskovitz@gmail.com), Kevin D. Heaney (OASIS Inc., Fairfax Station, VA), Jim A. Mercer (Univ. of Washington, Seattle, WA), and Art B. Baggeroer (Massachusetts Inst. of Technol., Cambridge, MA)

The deep ocean experiment, PhilSea09, was conducted April-May, 2009, in the central part of the northern Philippine Sea (22d N, 126d E). During one period in the experiment, the R/V Melville was station-keeping 35 km from the Distributed Vertical Line Array (DVLA) while seven tones, from 79 Hz to 535 Hz, were transmitted from a controlled source suspended below the ship. Recordings on the 1000-m section of the DVLA centered on the surface conjugate depth at 5026 m were dominated by the noise of this ship except at the controlled source tone frequencies. Using non-parametric statistical tests, the statistics of the spectral envelope at the tone frequencies are compared to the statistics of those for the nearby ship-noise-dominated frequency bins. When a high tone-signal to ship-noise ratio exists, the statistics of the tones differ from those of the ship noise at the 5% level of significance by the Kolmogorov-Smirnov two-sample test. Tone statistics are seen to be Gaussian distributed at frequency bands of low tone-signal to ship-noise ratios, whereas at high signal to noise ratios, the controlled source tones are non-Gaussian. When the signal to noise ratio is high, the statistics of the tone and of the noise are from different distributions. Both the tones and shipping noise travel approximately the same path to the DVLA, so these differences in the received field statistics represent differences in the statistical properties of these two acoustic sources themselves, not of the environment.

11:20

2aUW8. Probability distribution for energy of saturated broadband ocean acoustic transmission: Results from Gulf of Maine 2006 experiment. Duong D. Tran and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, d.tran@neu.edu)

The probability distribution of ocean-acoustic broadband signal energy after saturated multipath propagation is derived using coherence theory. The frequency components obtained from Fourier decomposition of a broadband signal are each assumed to be fully saturated with energy spectral density that obey the exponential distribution with 5.6 dB standard deviation and unity scintillation index. When the signal bandwidth and measurement time are larger than the correlation bandwidth and correlation time respectively of its energy spectral density components, the broadband signal energy obtained by integrating the energy spectral density across the signal bandwidth then follows the Gamma distribution with standard deviation smaller than 5.6 dB and scintillation index less than unity. The theory is verified with broadband transmissions in the Gulf of Maine shallow water waveguide in the 300–1200 Hz frequency range. The standard deviations of received broadband signal energies range from 2.7 to 4.6 dB for effective bandwidths up to 42 Hz, while the standard deviations of individual energy spectral density components are roughly 5.6 dB. The energy spectral density correlation bandwidths of the received broadband signals are found to be larger for signals with higher center frequencies and are roughly 10% of each center frequency.

11:40

2aUW9. Analysis of horizontal wave number statistical characteristics to the problem of sound propagation in two-dimensional-fluctuating shallow sea with losses. Oleg Gulin and Igor Yaroshchuk (Ocean Acoust., V.I. Il'ichev Pacific Oceanological Inst. FEB RAS, 43, Baltiyskaya St., Vladivostok, Primorskiy Kray 690041, Russian Federation, gulinoe@rambler.ru)

On the basis of the approach previously proposed by authors, a problem of the middle-frequency sound propagation in 2D-fluctuating shallow sea with losses is considered. Statistical characteristics of horizontal wave numbers corresponding to modal ones are studied. Fluctuations of horizontal wave numbers determine statistical features of wave field in random sea medium if wave field is sought by modal expansion. Within the framework of adiabatic approximation, we present calculations of sound field statistical moments, which demonstrate an effect of transmission loss attenuation along the horizontal distance. There are no references in acoustic literature for this fact. Some estimation has been carried out to explain new effect associated with two reasons that are the medium losses and sound speed fluctuations. They together influence wave numbers of modes in such a way attenuating losses.