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 $\pm$ 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 ($100 \leq \mathrm{Re} \leq 10000$) 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^{j}u > tʃu). However, since bilinguals are familiar with the co-articulatory affrication, they were not predicted to adapt palatalized stops as affricates (t^{j}u > 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

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 ɜr] 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.

Session 2pAAa

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

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

Invited Papers

1:00

2pAAa1. Amplified music and measurement results regarding inflatable membrane absorber technology. Niels W. Adelman-Larsen (Flex Acoust., Diplomvej 377, Kgs. Lyngby 2800, Denmark, nwl@flexac.com)

Previous studies and experience have shown that what distinguishes the best from the less well liked venues for pop and rock music is a shorter reverberation time in the 63–250 Hz octave bands. Since a longer reverberation time in these bands is needed in order to obtain warmth at classical music concerts, variable acoustics must address these frequencies in order to provide the best results in multi-purpose halls. This paper will expand on research on recommendable acoustics for amplified music. Certified measurements from reverberation chambers and installed systems on a patented, inflatable, on/off absorption technology are presented. Since the technology can be used in the entire ceiling area, the T30 of a hall can be lowered by almost 50% in the important octave bands for pop and rock music. Absorption coefficients are almost constant across the frequency bands from 63 to 1000 Hz. The technology, which is the only passive solution to enable variability also at the important lower frequencies, is meant to be used in any hall where both classical as well as amplified music is being played, such as in music schools and performing arts centers.

1:20

2pAAa2. The role of high-frequency cues for spatial hearing in rooms. Hari Bharadwaj, Salwa Masud, and Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol. (CompNet), Boston Univ., 19 Euston St., 1B, Brookline, MA 02446, hari@nmr.mgh.harvard.edu)

The ability to attend to a sound source of interest while ignoring competing sounds is vital to navigating everyday acoustic scenes. Commonly, this ability depends on the ability to focus on a sound source using acoustic spatial cues, particularly interaural time differences (ITDs) and interaural level differences (ILDs). Based on past studies of localization in anechoic settings, low-frequency ITDs have been thought to dominate perception of source location. However, reverberant environments differentially degrade ITDs and ILDs, which may affect their relative influence on localization. Moreover, a recent study suggests that ILDs play a bigger role on spatial perception in reverberant settings than in anechoic settings. Here, in a series of localization and spatial attention experiments using high-pass, low-pass and broadband sounds, we tested the hypothesis that high-frequency ILD and envelope ITD cues are important for spatial judgments in reverberant rooms. We also measured the brainstem frequency following responses (FFRs) of individual subjects in response to click trains and spoken syllables. Results suggest that compared to in anechoic space, in reverberant settings, high-frequency cues are more reliable and influential on perception and that the strength of FFR phase locking to stimulus envelope predicts how well individual listeners can direct spatial attention.

1:40

2pAAa3. Internet rooms from internet audio. Chris Chafe and John Granzow (CCRMA/Music, Stanford Univ., CCRMA/Music, Stanford, CA 94305, cc@ccrma.stanford.edu)

Music rehearsal and concert performance at a distance over long-haul optical fiber is a reality because of expanding network capacity to support low-latency, uncompressed audio streaming. Multichannel sound exchanged across the globe in real time creates "rooms" for synchronous performance. Nearby connections work well and musicians feel like they are playing together in the same room. Larger, continental-size, distances remain a challenge because of transmission delay and seemingly subtle but perceptually important cues which are in conflict with qualities expected of natural rooms. Establishing plausible, room-like reverberation between the endpoints helps mitigate these difficulties and expand the distance across which remotely located musicians perform together comfortably. The paper presents a working implementation for distributed reverberation and qualitative evaluations of reverberated versus non-reverberated conditions over the same long-haul connection.

2:00

2pAAa4. Designing a stackable diffusor/absorber tailored to a violin and cello practice room. Myoung woo Nam, Kyogu Lee (Trans-Disciplinary Studies, Seoul National Univ., D406, Iuidong 864-1, Yeongtonggu, Suwon, Gyeonggi 443-270, South Korea, mnam@snu.ac.kr), and Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts, Lowell, MA)

As distinct from a much larger concert hall, the typical practice room directs added acoustic emphasis to small room challenges such as room resonances and unwanted reflections. Although rich reverberation is not easily achieved in a small space, the proposed diffusor/absorber seeks to make practice more acoustically comfortable and rewarding. The treatment is designed to attenuate the spectral portion

of the violin and cello sound that engages the room resonances. When musicians play violin and cello, omni-directional low frequencies are primarily produced in lower elevation of the room, while more directional higher frequencies of interest to the performer are directed more to upper area. This diffusor/absorber is designed to provide more absorption for bottom area and more diffusion for upper area. In this configuration, the diffusor/absorber gives comfortable acoustical conditions for musicians to practice. Based on sound propagation characteristics [J. Meyer, "The sound of the orchestra," J. Audio Eng. Soc. **41**(4) (1993)] and formant information of violin and cello [M. Nam and K. Lee, "Analyzing string instrument formant," in *Proceeding of Acoustical Society of Korea Conference* (2011)], this design proposes a moveable acoustic panel-box suitable for a typical musician's home practice room and a small sized recording studio.

Contributed Papers

2:20

2pAAa5. Improving the indoor sound quality by using cymatic shapes. Alaa S. Algargoosh (College of Design, Univ. of Dammam, 3140 Alabbas bin Ali, Dammam 32433-4614, Saudi Arabia, alaa_algargoosh@yahoo.com), Hany Hossam Eldien (College of Architecture and Planning, Univ. of Dammam, Dammam, Saudi Arabia), and Hala El Wakeel (College of Design, Univ. of Dammam, Dammam, Saudi Arabia)

Acoustic diffusers are important components in enhancing the quality of room acoustics. This paper investigates a new type of 2D diffusers obtained by the Cymatics phenomena. Cymatics is the study of sound and vibration made visible, typically on the surface of a plate, diaphragm, or membrane. Four shapes of the diffusers were designed by the Cymatic shapes and modeled by using a quadratic residue sequence. The polar response of the diffusers was measured using DIRAC software. Polar response results were generally consistent with expectations. This type of diffusers can generate a uniform polar response over the frequency range we are interested in (400–4000 Hz). It is found that this type of acoustic diffusers can be used to maintain the acoustic energy in a room and at the same time can treat unwanted echoes and reflections by scattering sound waves in many directions.

2:40–3:00 Break

3:00

2pAAa6. Sound concentration caused by curved surfaces. Martijn Vercammen (Peutz, Lindenlaan 41, Molenhoek 6584 AC, Netherlands, m.vercammen@mook.peutz.nl)

In room acoustics, the focusing effect of reflections from concave surfaces is a well-known problem. The occurrence of concave surfaces has tended to increase in modern architecture, due to new techniques in design, materials, and manufacturing. Focusing can cause high sound pressure levels, sound coloration, or an echo. Although the problem is well known, the amount of amplification that occurs in the focusing point and the sound field around the focusing point are not. The pressure in the focusing point can only be calculated using wave-based methods. An engineering method that is based on the Kirchhoff Integral is presented to approximate the reflected sound field in and around the focusing point for a few basic geometries. It will be shown that both the amplification and the area of the focusing is strongly related to wavelength. The focusing caused by surfaces that are curved in two directions (sphere, ellipsoid) is much stronger than that caused by surfaces that are curved in only one direction (cylinders). This method enables designers to evaluate and thereby improve or redesign the geometry. The method is illustrated with a few examples.

3:20

2pAAa7. Can we use the standard deviation of the reverberation time to describe diffusion in a reverberation chamber? Margriet Lautenbach and Martijn Vercammen (Peutz, Paletsingel 2, Zoetermeer 2718 NT, Netherlands, m.lautenbach@zoetermeer.peutz.nl)

It is generally assumed that the limited diffusion properties of reverberation rooms, especially with a strongly sound absorbing sample, is the main reason for the bad reproducibility values for the sound absorption between laboratories. Reverberation rooms should be made much more diffuse to reduce the inter laboratory differences. Although there are practical ways to achieve this, it is most important that there will be a requirement in the ISO 354 standard on the diffusing quality of the sound field. One possibility is to use the standard deviation of the reverberation time for different source-microphone combinations in the reverberation room. Measurements are performed to investigate the influence of different settings of a reverberation room on the standard deviation of the reverberation time, compared to the theoretical standard deviation. This is done with the interrupted impulse method and the integrated impulse method. The results will be presented in this paper. The usefulness of this qualification method for the ISO standard will be discussed.

3:40

2pAAa8. A new third generation time variant electro-acoustic enhancement system. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

This paper describes the hardware and software implementation in a new generation of time variant electronic acoustic enhancement systems designed for use in medium and large sized venues. Examples of currently installed systems and applications will be discussed, as well as capabilities for sound file storage and playback, multi-channel film surround sound, 2D effects panning, and 3D live tracking.

4:00

2pAAa9. Optimization of electroacoustic resonators for semi-active room equalization in the low-frequency range. Etienne Rivet (Laboratoire d'Électromagnétisme et d'Acoustique, École Polytechnique Fédérale de Lausanne, EPFL - STI - IEL - LEMA, Station 11, Lausanne, Vaud 1015, Switzerland, etienne.rivet@epfl.ch), Romain Boulandet, and Hervé Lissek (Laboratoire d'Électromagnétisme et d'Acoustique, École Polytechnique Fédérale de Lausanne, Lausanne, Switzerland)

At low frequencies in listening rooms, standing waves cause large frequency-response variations within the whole space. These unwanted phenomena have a significant impact on the sound quality of an audio system rendering. Unfortunately, state-of-the-art soundproofing solutions cannot efficiently handle such low-frequency sound energy. To alleviate this problem, electroacoustic resonators can be used to damp room modes. This concept is based on the connection of direct-radiator loudspeakers to synthetic electrical loads allowing the passive dissipation of a certain part of the incoming acoustic energy of the sound field. Through judicious control of acoustic impedance, and depending of the placement of the electroacoustic resonators in the room, a significant damping of the dominant natural resonances can be achieved in order to meet the specifications of sound reproduction. This paper presents the design of prototypes of electroacoustic resonators and investigates their optimization and spatial arrangement in the perspective of semi-active room equalization.

4:20

2pAAa10. Physical and subjective factors of spatial envelopment impression of surround sound reproduction. Toru Kamekawa and Atsushi Marui (Musical Creativiry and the Environ., Tokyo Univ. of the Arts, 1-25-1, Senju, Adachi-ku, Tokyo 120-0034, Japan, kamekawa@ms.geidai.ac.jp)

How we feel the envelopment of reproduced sound? For the evaluation of spatial impression, "envelopment" is one of the key factors among several attributes in audio reproduction. In the authors' past research, three attributes were elicited from participants regarding surround sound reproduction using triadic elicitation procedure. The three attributes are "brightness," "temporal separability," and "spatial homogeneity of envelopment." In this paper, authors focused on "spatial homogeneity of envelopment" and considered related physical parameters such as ESC (ear signal correlation; defined as the correlation between left and right ears' signal). From comparison between the result of subjective test and the data measured with a dummy-head facing toward different

angles, it was found that the correlation between the ESC of different head angles and the amount of differences of ESC from focused sound to diffused sound contribute the sense of "spatial homogeneity of envelopment."

4:40

2pAAa11. Influence of low-order room reflections on sound zone system performance. Marek Olik (CVSSP, Dept. of Electron. Eng., Ctr. for Vision Speech and Signal Process. (CVSSP), Univ. of Surrey, Guildford GU2 7XH, United Kingdom, m.olik@surrey.ac.uk), Philip Jackson, Philip Coleman (CVSSP, Dept. of Electron. Eng., Univ. of Surrey, Guildford, United Kingdom), Martin Olsen, Martin Møller, and Søren Bech (Bang & Olufsen, Struer, Denmark)

Studies on sound field control methods able to create independent listening zones in a single acoustic space have recently been undertaken due to the potential of such methods for various practical applications, such as individual audio streams in home entertainment. Existing solutions to the problem have shown to be effective in creating high and low sound energy regions under anechoic conditions. Although some case studies in a reflective environment can also be found, the capabilities of sound zoning methods in rooms have not been fully explored. In this paper, the influence of low-order (early) reflections on the performance of key sound zone techniques is examined. Analytic considerations for small-scale systems reveal strong dependence of performance on parameters such as source positioning with respect to zone locations and room surfaces, as well as the parameters of the receiver configuration. These dependencies are further investigated through numerical simulation to determine system configurations which maximize the performance in terms of acoustic contrast and array control effort. The design rules for source and receiver positioning are suggested, for improved performance under a given set of constraints such as a number of available sources, zone locations, and the direction of the dominant reflection.

TUESDAY AFTERNOON, 4 JUNE 2013 — 513DEF, 12:55 P.M. TO 5:20 P.M.

Session 2pAAb

Architectural Acoustics and Noise: Dah-You Maa—His Contributions and Life in Acoustics

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

Jing Tian, Cochair
Inst. of Acoust., Chinese Acad. of Sci., 21 Beisihuanxilu, Beijing 100190, China

Chair's Introduction—12:55

Invited Papers

1:00

2pAAb1. Dah-You Maa, friend and scholar. Leo L. Beranek (Westwood, MA) and Ning Xiang (Grad. Prog. in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY, Xiangn@rpi.edu)

Maa, Dah-You's life as a scholar and a close friend of many both in China and North America is the theme. A number of his scholarly achievements from his UCLA and Harvard time, to the post-Cultural Revolution period, up to recent years, is reviewed. His two years as a research fellow at UCLA and Harvard which ended with his receiving his Ph.D. at Harvard University in 1940 is detailed. Then began his life in Kunming during World War II where he was a professor in the E.E. Department of the National S.-W. Associated University. He returned to Beijing and became the first Dean of Engineering in the National Peking University (now Beijing University). During the Cultural Revolution, 1966–1976, he was under house arrest. His sponsorship of the first All-China Acoustics Conference after the Cultural Revolution in 1979 heralded his position as China's leading acoustician. Excerpts from his many letters through the years complete the presentation.

1:20

2pAAb2. Prof. Dah-You Maa's Contribution to Acoustics. Jing Tian (Inst. of Acoust., Chinese Acad. of Sci., 21 Beisihuanxilu, Beijing 100190, China, tian@mail.ioa.ac.cn)

In his life-long career pursuing, Professor Dah-You Maa contributed greatly in many areas of acoustics, not only in acoustical research and development, but also in the promotion of acoustical education, application, and legislation. Professor Maa presented a simplified method for calculation of normal modes for room acoustics, invented micro-perforated panel absorbers and micro-perforation jet mufflers, gave a formula of jet noise power via air pressure, and acoustically designed the first and biggest Congress Hall, built the first set of acoustical laboratories, and established the acoustical standard system in China. He also supervised tens of postgraduate students working in environmental acoustics, building acoustics, speech signal processing, nonlinear acoustics, and active noise control. In this paper, the main contribution of Professor Maa is introduced. Several of his typical research works, such as micro-perforated panel absorbers and micro-perforation jet mufflers, are explained in detail, from which we can well appreciate the physical insights and theoretical skills of Professor Maa.

1:40

2pAAb3. Dah-You Maa: Most senior academic brother. David T. Blackstock (Appl. Res. Labs. & Mech. Eng. Dept., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, dtb@austin.utexas.edu)

F. V. Hunt's ONR-supported acoustics lab at Harvard turned out 30 PhD graduates after WW II. As one of those students in the late 1950s, I gradually became aware that Hunt had had a prewar group of graduate students as well. Leo Beranek was Hunt's first Ph.D., Dah-You Maa his second (both in 1940). Maa's two 1939 JASA articles on room acoustics, one coauthored with Beranek and Hunt, were benchmark papers of the day. I became fascinated with Maa's story, partly because he seemed so completely unreachable. He had returned to China during the war; afterward the Cold War intervened. I finally met my much older academic brother in 1980, at the 10th ICA in Sydney, Australia. Thus began a warm and rewarding relationship. In 1987, he began representing China on the International Commission on Acoustics, and for 7 years, we saw each other annually at Commission meetings. I learned of his work on nonlinear standing waves, a problem in which I too shared a keen interest. The high point was at the 14th ICA in Beijing. I was finally able to see his laboratory and meet his doctoral student Ke Liu. A memorable dinner followed that evening.

2:00

2pAAb4. Dah-You Maa and the many facets of modal density. Richard H. Lyon (MIT, 60 Prentiss Lane, Belmont, MA 02478-2021, rhlyon@lyoncorp.com)

Maa's formula for the modal density of a room is known to everyone in acoustics. The wonderful part of it is that it so logical and direct that you don't have to memorize or look it up - you can simply re-derive it on the spot. When it was published by Maa in 1939, there was a competing formulation by Dick Bolt that had the same volume dependent term, but was quite different in the surface area and edge length terms. But by the time of the landmark paper by Morse and Bolt in 1944 ["Sound waves in rooms", Rev. Mod. Phys. **16**(1) (1944)], Maa's formula had won and was the relationship cited by Morse and Bolt. The uses of modal density in the response statistics of acoustical spaces and structures for the purposes of estimating impedances, energy flow, and phase statistics have grown as recognition of this fundamental property of resonant systems has grown.

2:20

2pAAb5. Maa Dah-You and the design of reverberant rooms for determination of sound power. George C. Maling (60 High Head Rd., Harpswell, ME 04079, maling@alum.mit.edu)

Reverberation rooms are useful tools for the determination of sound power emitted by machines and other sound sources. One critical factor in the design of such rooms is the mode spacing at low frequencies. A seminal contribution to our understanding of mode spacing was made by Maa in 1939, and his paper was published in the Journal of the Acoustical Society of America. Richard Bolt also made a contribution at about the same time. In this paper, we begin with Maa's work and trace the development of mode spacing statistics through the work of Richard H. Bolt and Ludwig Sepmeyer. A computer study by the author of mode spacing statistics in 200 cubic meter rooms is described. This work led to recommendations for room dimensions to be included in an international standard on determination of sound power in reverberation rooms.

2:40

2pAAb6. Remembering one of Maa Dah-You's last essential contributions to acoustics. Jiqing Wang (Inst. of Acoust., Tongji Univ., 1239 Siping Rd., Shanghai 200092, China, wongtsu@126.com)

The book *The fundamentals of Modern Acoustic Theory* written by Maa Dah-You was published (in Chinese) in March 2004 at his age of 88. This 421 pages volume with 15 chapters covers almost all main acoustic fields from basics to many extended areas, such as linear to non-linear acoustics, common acoustical measures to active control, and the aerodynamic sound and thermo-acoustics. References listed in each chapter include many Maa's pioneering work of the related acoustic fields. This book systematically reflects his deeper understanding of acoustics theory, and further account new developments of acoustic technology. Moreover, the history of acoustics, particularly of the ancient China in the beginning sections of the first chapter, and the overview of the future developments in the afterword are also the exclusive items in this book. A well prepared exercise at the end of each chapter is most helpful for better understanding and thinking of the related topics. Therefore, it is widely used as a text or main reference book in many universities and research community in China.

3:00–3:20 Break

3:20

2pAAb7. Three months in Beijing—Micro-perforated absorbers, eigenvalues, and other topics. Christian Nocke and Catja Hilge (Akustikbuero Oldenburg, Katharinenstr. 10, Oldenburg 26121, Germany, nocke@akustikbuero-oldenburg.de)

In this contribution, a personal and scientific report on a three month exchange stay at the Institute of Acoustics in Beijing will be reported. Micro-perforation and low frequencies in small rooms have been the background of an investigation under supervision of Professor Daa-You Maa in 1999. Maa's ideas from the 1930s about the distribution on eigenmodes in rooms in combination with micro-perforated absorbers as reference absorbers had been the basis of this work. Abacus and computer calculations as the basis of old/new ideas formed the background of a stay as a student. Even after more than 12 years, micro-perforation is part of the daily work as consultant often dealing with new product ideas and developments. The presentation will give a very brief overview on this work that started in Beijing. Furthermore, in remembrance of a once in a life time experience personal experiences with Professor Daa-You Maa, his wife, and his colleagues will be reported.

3:40

2pAAb8. Surface waves over rigid-porous and rough surfaces. Keith Attenborough, Imran Bashir, and Shahram Taherzadeh (DDEM, The Open Univ., Walton Hall, Milton Keynes MK7 6AA, United Kingdom, k.attenborough@open.ac.uk)

Surface waves are created by near grazing sound propagation from a point source over either a rigid-porous layer or a (slightly) rough surface. In theory, they have similar origins. Frequency- and time-domain measurements have been made on surfaces composed from parallel periodically spaced rectangular strips (width 0.0126 m, height 0.0253 m) on an acoustically hard surface. The edge-to-edge spacing between the strips has been varied between 0.003 and 0.06 m. Frequency domain predictions show that when the spacing is substantially smaller than the strip height these surfaces may be regarded as locally reacting rigid-framed hard-backed porous layers with an effective depth slightly larger than the strip height. When the spacing is comparable to the strip height or greater the surfaces behave as periodically rough surfaces. Both frequency- and time-domain results show that surface waves of comparable magnitudes are created over the range of strip spacings studied but the main frequency content of these acoustically-induced surface waves is lowered as the mean spacing is increased. These data suggest that the surface waves have a similar physical origin and that they can be created also over a micro-perforated surface.

4:00

2pAAb9. Investigation of sound scattering by a surface using the acoustical wave propagator method. Jie Pan, Hongmei Sun, and James Leader (Mech. and Chem. Eng., The Univ. of Western Australia, 35 Stirling Highway, Crawly, WA 6009, Australia, jie.pan@uwa.edu.au)

The time domain acoustical wave propagator (AWP) method is used as a tool to study the sound scattering properties of a surface. As the method allows the modeling of the excitation, absorption, and scattering of sound in a vicinity of the surface within a very short period of time, the effect of other surfaces on these process can be ignored, and the acoustical property of a specific surface can be investigated in detail without solving the sound wave equation in the whole space. Sound scattering by a finite impedance strip mounted on a rigid baffle in two different conditions is solved by the method. The result is used to illustrate the efficiency of the method.

4:20

2pAAb10. A composite sound absorber with micro-perforated panel and shunted loudspeaker. Jiancheng Tao, Qijing Jiao, Xiaojun Qiu, and Ning Han (Inst. of Acoust., Nanjing Univ., Rm. 307, Acoust. West Bldg., No.22 Hankou Rd., Nanjing, Jiangsu, China, Nanjing, Jiangsu 210093, China, jctao@nju.edu.cn)

Micro-perforated panel (MPP) backed by a rigid cavity is a widely used clean sound absorber; however, its application at low frequency is limited because a deep cavity is required to achieve good sound absorption in the low frequency range. In the present paper, a composite absorber composed by an MPP and a shunted loudspeaker is proposed. The loudspeaker is installed at the back wall of the air cavity and the acoustic impedance can be optimized by adjusting the parameters of the loudspeaker and the electronic components in the shunt to improve the sound absorption performance. The prediction model of such a finite-sized composite absorber is established based on the mode analysis solution and the equivalent circuit of the loudspeaker. Both numerical simulations and experiments show that the thickness of the proposed composite absorber can be much smaller than that of traditional MPP constructions.

4:40

2pAAb11. Active noise control in rooms. Jiri Tichy (Penn State Univ., 5552 N Citation Rd., Toledo, OH 43615, tichy@engr.psu.edu)

Maa published his ideas and achievements in the early days of the new branch of noise control based on using secondary sources to reduce the sound level of the primary sources by negative interference. He concentrated on active control of the reverberant sound in a room by a loudspeaker placed in one of the room corners and a microphone in its close vicinity. He examined the role of the complete solution of the wave equation consisting of a direct wave radiated from the sources and the reverberant field created by room modes. His work resulted in deriving a general formula for possible noise reduction that is independent of the shape, size, and the content of the room. He has shown noise reduction of 8 dB for a noise band centered at 100 Hz.

5:00

2pAAb12. A hybrid modal analysis for enclosed sound fields and its applications. Buye Xu (Signal Process. Res., Starkey Hearing Technol., 6600 Washington Ave. S, Eden Prairie, MN 55344, buye_xu@starkey.com), Scott D. Sommerfeldt, and Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

In 1939, Dr. Dah-You Maa presented a paper entitled “The distribution of eigentones in a rectangular chamber at lower frequency ranges” [J. Acoust. Soc. Am. **10**, 258 (1939)]. Since then, his interest in room acoustics has not diminished. More than six decades later, Maa proposed an idea of adding a monopole solution to the normal modal expansion to improve the accuracy in simulating the near-field sound field in rooms [D. Y. Maa, Acta Acust. (Beijing) **27**, 385–388 (2002)]. Following this idea, a hybrid model that combines the free field Green’s function and a modal expansion has been proposed by the authors based on a rigorous mathematical derivation [Xu *et al.*, J. Acoust. Soc. Am. **128**, 2857–2867 (2010)]. The hybrid modal expansion can be further extended for complex sound sources by introducing the multipole expansion to the solution. In this talk, the theoretical derivation of the hybrid modal expansion will be reviewed, followed by examples demonstrating the use of the hybrid modal expansion in real-world applications.

Session 2pAB

Animal Bioacoustics, Psychological and Physiological Acoustics, Signal Processing in Acoustics, and Noise: Listening in the Natural Environment

Cynthia F. Moss, Cochair
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Peter M. Narins, Cochair
Integrative Biol. & Physiol., UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095

Chair's Introduction—12:55

Invited Papers

1:00

2pAB1. Three directions in research on auditory scene analysis. Albert S. Bregman (Psychology, McGill Univ., 1205 Doctor Penfield Ave., Montreal, QC H3A 1B1, Canada, al.bregman@mcgill.ca)

Research on auditory scene analysis (ASA) began with some simple laboratory phenomena such as streaming and illusory continuity. Subsequently, research has gone in three directions, downwards toward underlying mechanisms (by neurophysiologists), upwards toward system organization (by computer scientists), and sideways toward other species (by neurobiologists). Each direction has its problems. The downward approach sometimes takes a phenomenon-oriented view of ASA, leading to simple explanations of a single ASA demonstration, such as streaming, with no obvious connection to any larger system. Research done by the upward approach usually takes the form of a computer program to achieve ASA in a working system, often ignoring known facts about human ASA, in favor of mathematically understood principles. The sideways approach often finds that non-human animals can respond to an important sound despite the presence of other interfering sounds. However, there is no reason to believe that a frog, a fish, and a human accomplish this by means of the same mechanisms. So finding out how some animal does this, while interesting in its own right, may shed little light on how humans do it. I will describe some properties of the human ASA system that should be borne in mind when manufacturing explanations.

1:20

2pAB2. Mechanisms of perceiving communication sounds in scenes. Sarah M. Woolley (Psychology, Columbia Univ., 406 Schermerhorn Hall, 1190 Amsterdam Ave., New York, NY 10027, sw2277@columbia.edu)

Vocal communicators must perceive the vocal signals of social partners in complex auditory scenes that include distracting background sounds. The auditory system must therefore parse auditory scenes into multiple information streams and/or accurately encode individual vocalizations despite the presence of competing sounds. Explaining mechanisms whereby neural representations of vocalizations are extracted from neural representations of scenes is an important part of understanding how auditory processing leads to perception of communication signals in complex scenes. We study how songbirds recognize individual vocalizations (songs) in scenes of conspecific choruses. We combine behavioral studies with neurophysiological studies of song and scene coding in midbrain and cortex. We find dramatic transformations in the neural coding of songs and scenes between different regions of auditory cortex. Neural representations of individual songs are dense and non-selective in the midbrain and primary cortex, but are sparse and highly selective in higher cortex. Sparse coding neurons produce background-invariant responses to individual songs in scenes, providing a potential neural mechanism for the perception of individual communication vocalizations in complex auditory scenes. Acoustic manipulations of song and pharmacological manipulations of neural coding suggest that sparse and background-invariant representations of songs in higher cortex are due to context-dependent inhibition.

1:40

2pAB3. The search for a neural basis of communication: Learning, memory, perception. and performance of vocal signals. Jonathan Prather (Dept. of Zoology and Physiol., Univ. of Wyoming, 1000 E Univ Ave. - Dept. 3166, Laramie, WY 82071, jprathe2@uwyo.edu)

Brain mechanisms for communication must establish a correspondence between sensory perception and motor performance of individual signals. A class of neurons in the swamp sparrow forebrain is well suited for that task. Recordings from awake and freely behaving birds reveal that those cells express categorical auditory responses to changes in note duration, a learned feature of their songs, and the neural response boundary accurately predicts the categorical perceptual boundary measured in field studies. Extremely precise auditory activity of those cells represents not only songs in the adult repertoire but also songs of others and tutor songs, including those imitated only very few times or perhaps not at all during development. Furthermore, recordings during singing reveal that these cells also express a temporally precise auditory-vocal correspondence, and limits on auditory responses to extremely challenging tutor songs may contribute to the emergence of a novel form of song syntax. Therefore, these forebrain neurons provide a mechanism through which sensory perception may influence motor performance to enable imitation. These cells constitute the projection from a premotor cortical-like area into the avian striatum (HVCX neurons), and data from humans implicate analogous or homologous areas in perception and performance of the sounds used in speech.

Contributed Paper

2:00

2pAB4. Call perception in mice. Erikson Neilans, David Holfoth, Kelly E. Radziwon (SUNY, Univ. at Buffalo, 207 Park Hall, Buffalo, NY 14260, eneilans@buffalo.edu), Christine V. Portfors (School of Biological Sci., Washington State Univ., Vancouver, WA), and Micheal L. Dent (SUNY, Univ. at Buffalo, Buffalo, NY)

Acoustic communication in laboratory mice is a relatively recent subject of experimental study, often yielding disparate findings. For example, researchers often manually place mouse ultrasonic vocalizations (USVs) into categories based on spectrotemporal characteristics, but the numbers and types of categories differ widely between laboratories. Here, we attempt to determine what cues CBA/CaJ mice use to discriminate between vocalizations by testing them in an operant conditioning paradigm. The mice were trained to discriminate a repeating background containing one USV from several target USVs. The targets were different call types used by Holmstrom *et al*. (2010) and manipulations of the background calls, such as removing the frequency modulation, shifting the entire call up or down in frequency, shortening or lengthening the call, or reversing the entire call. Results show that large frequency shifts were easy for the mice to discriminate, while reversing the calls and removing the frequency modulation were much more difficult. For most calls, similarity in spectrotemporal characteristics yielded poor discrimination performance. These results are the first to show that mice can discriminate between some vocalizations but not others, and that they may place different meaning to different call types, though not necessarily the call types designated by humans.

2:20–2:40 Break

Invited Papers

2:40

2pAB5. Auditory processing for contrast enhancement of salient communication vocalizations. Alex G. Dunlap, Frank G. Lin (Biomed. Eng., Georgia Inst. of Technol. and Emory Univ., Atlanta, GA), and Robert C. Liu (Biology, Emory Univ., 1510 Clifton Rd. NE, Rm. 2006, Atlanta, GA 30322, robert.liu@emory.edu)

In a natural acoustic environment, coherent representations of auditory objects and sources are streamed from the myriad sounds that enter our ears. Features of those sounds that are familiar and behaviorally salient to us are detected and discriminated into invariant precepts that inform us about our external world. Research into how this occurs is increasingly converging on the idea that there is a transformation from the auditory periphery wherein an initial acoustically faithful representation by neurons becomes progressively altered to enhance the population neural representation of perceptually relevant aspects of the sound. How this occurs may vary for sounds whose meanings are acquired in different ways, perhaps depending on what actions and decisions must be executed upon recognition. We have investigated this process in a natural social context in which mouse mothers “learn” about the meaning of pup ultrasound vocalizations through their maternal care. Here we discuss our recent studies in awake mice using electrophysiological, behavioral, immunohistochemical, and computational methods. Our results suggest that experience with natural vocalizations may alter core auditory cortical neural responses so that the contrast in activity across the neural population enhances the detection and discrimination of salient calls.

3:00

2pAB6. Influences of perceptual continuity on everyday listening. Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu), Golbarg Mehraei (Speech and Hearing Biosci. and Technol., Harvard/MIT, Cambridge, MA), Scott Bressler, and Salwa Masud (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

In the natural environment, listeners face the challenge of parsing the sound mixture reaching their ears into individual sources, and maintaining attention on a source of interest through time long enough to extract meaning. A number of studies have shown that continuity of certain acoustic features (including pitch, location, timbre, etc.) allows the brain to group sound from one acoustic sound source together through time to form an auditory object or stream. This presentation reviews results demonstrating that auditory feature continuity has important consequences on how listeners maintain attention on a stream through time. For instance, continuity of a sound feature that a listener knows is irrelevant to the task at hand nonetheless impacts the ability to maintain auditory attention based on some other sound feature. Moreover, the influence of auditory feature continuity decreases as the time between events in a given sound stream increases. Taken together, these behavioral results support the idea that auditory attention operates on auditory objects, rather than on individual sound features, and that feature continuity has an obligatory influence on the formation of auditory streams, and therefore on how selective auditory attention allows us to communicate in everyday settings.

3:20

2pAB7. Bugs and bats: Neural analysis of behaviorally relevant sounds in crickets. Gerald Pollack (Dept. of Biol., McGill Univ., 1205 Dr. Penfield Ave., Montreal, QC H3A1B1, Canada, gerald.pollack@mcgill.ca)

Hearing in crickets is specialized to serve particular behavioral functions, namely intraspecific communication and predator avoidance. Male crickets produce species-specific acoustic signals (songs) that attract distant females, promote copulation, and contribute to agonistic interactions with rivals. Crickets also hear the echolocation calls of aerially hunting bats, which evoke avoidance responses. These clear behavioral functions of hearing, combined with the relative simplicity of the cricket’s nervous system, make it possible to address questions about how behaviorally relevant sensory signals are analyzed at the level of single, uniquely identifiable nerve cells. Cricket songs and bat calls differ both in rhythm and in spectrum, and neurons throughout the auditory processing chain are specialized for processing these two sorts of signal. I will focus on specializations that are evident at early stages of auditory processing, i.e., primary sensory neurons and the first-order interneurons with which they interact.

Contributed Paper

3:40

2pAB8. New observations and modeling of an unusual spatiotemporal pattern of fish chorusing off the southern California coast. Gerald L. D'Spain, Heidi H. Batchelor, and Tyler A. Helble (Marine Physical Lab., Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu)

The purpose of this paper is to present new results on an unusual spatiotemporal pattern of fish chorusing off the southern California coast. Characteristics of this fish chorus have been reported previously; it occurs at night in the late spring and summer months in shallow, sandy bottom regions just outside the surf zone. The background sound levels increase by up to 30 dB and cycle in level with a period of 30–35 s all night long. In this paper, recent results from measurements made by a set of high spatial resolution sensor systems spanning a 50-km stretch of coastline out to 20 km offshore over a 2-month time period are presented. These data allow the spatial dependence and long-term temporal variability of the chorus to be examined at high spatial resolution. Refinements to a numerical model that predicts this chorusing behavior are required to account for some aspects of these new observations. [Work supported by the Office of Naval Research, Code 322-MMB.]

Invited Papers

4:00

2pAB9. The influence of anthropogenic noise on the evolution of communication systems. Peter M. Narins (Integrative Biol. & Physiol., UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095, pnarins@ucla.edu)

Many species of animals, including man, face the formidable task of communicating in noisy environments. In this talk, I shall discuss the effects of anthropogenic (man-made) noise on the calling behavior of anuran amphibians. Moreover, the role of spectral, temporal, and spatial separation in minimizing masking by background noise will be examined. For example, presenting high-level, periodic (or aperiodic) tones at the males co-note frequency to males of the Puerto Rican treefrog, *Eleutherodactylus coqui* results in a clear shift in their calling pattern in an attempt to minimize acoustic overlap with the interfering playback stimulus. Amphibians also have a remarkable ability to shift their call timing in response to small intensity shifts in the background noise. Males of *E. coqui* are capable of reliably detecting a change in interfering tone intensity as small as 2–4 dB. Finally, I shall present behavioral evidence that anthropogenic noise may act as a strong selective force in sculpting the acoustic communication systems of several species of Old World frogs. Some techniques for visualizing sound interference will be discussed. [Supported by grants from the NIDCD (Grant No. DC-00222), and the UCLA Academic Senate (3501).]

4:20

2pAB10. Active listening in a complex environment. Melville Wohlgemuth and Cynthia F. Moss (Psych. and ISR, Univ. of Maryland, Biol.-Psych. Bldg. 2123M, College Park, MD 20742, cynthia.moss@gmail.com)

Spatially guided behaviors in echolocating bats depend upon the dynamic interplay between auditory information processing and adaptive motor control. The bat produces ultrasonic signals and uses information contained in the returning echoes to determine the direction and distance of objects in space. With this acoustic information, the echolocating bat builds a 3-D auditory representation of the world, which it uses to guide a suite of coordinated motor behaviors, including head and pinna movements, as well as the timing, duration, frequency characteristics, and directionality of sonar signals. Adaptive echolocation behaviors shape the acoustic information available to the bat's sonar imaging system and provide a window to its perception of complex scenes. In a complex environment, an echolocating bat encounters multiple reflecting surfaces that return a cascade of echoes from each sonar transmission. The work presented here will focus on adaptive echolocation behaviors of the big brown bat as it tracks a selected prey item in the presence of multiple objects, both obstacles and other prey. Data suggest that bats can successfully segregate streams of echoes from closely spaced objects through finely tuned adaptive sonar signal control.

Contributed Papers

4:40

2pAB11. Your attention, please! Determining saliency of competing audio stimuli in natural scenarios. Francesco Tordini (Dept. of Elec. and Comput. Eng., McGill, 3480 University St., McConnell Eng. Bldg., Montreal, QC H3A 2A7, Canada, tord@cim.mcgill.ca), Albert S. Bregman (Dept. of Psych., McGill, Montreal, QC, Canada), Anupriya Ankolekar, Thomas Sandholm (Hewlett-Packard Lab., Palo Alto, CA), and Jeremy R. Cooperstock (Dept. of Elec. and Comput. Eng., McGill, Montreal, QC, Canada)

Perceptual saliency is a precursor to bottom-up attention modeling. While visual saliency models are approaching maturity, auditory models remain in their infancy. This is mainly due to the lack of robust methods to gather basic data, and oversimplifications such as an assumption of monaural signals. Here we present the rationale and initial results of a newly designed experimental paradigm, testing for auditory saliency of natural sounds in a binaural listening scenario. Our main goal is to explore the idea that the saliency of a sound depends on its relation to background sounds by using more than one sound at a time, presented against different backgrounds. An analysis of the relevant, emerging acoustical correlates together with other descriptors is performed. A review of current auditory saliency models and the deficiencies of conventional testing approaches are provided. These motivate the development of our experimental test bed and more formalized stimulus selection criteria to support more versatile and ecologically relevant saliency models. Applications for auditory scene analysis and sound synthesis are briefly discussed. Some initial conclusions are drawn about the definition of an expanded feature set to be used for auditory saliency modeling and prediction in the context of natural, everyday sounds.

5:00

2pAB12. A neuroethological analysis of the information in propagated communication calls. Frederic E. Theunissen, Solveig Mouterde (Psychology, UC Berkeley, UC Berkeley, 3210 Tolman Hall, Berkeley, CA 94720, theunissen@berkeley.edu), and Nicolas Mathevon (Univ. of Lyon/Saint-Etienne, Saint-Etienne, France)

The detection and recognition of communication signals in natural soundscapes is a difficult task that animals and birds in particular excel at. We have used a neuroethological approach to quantify the recognition

performance for propagated communication signals in the zebra finch, specifically regarding the information about individual identity. The propagated signals were analyzed using a regularized discriminant function analyses on a complete spectrographic representation of the signals. We found (1) a reduction in the informative frequency range a long distances yielding a frequency band sweet-spot, (2) that call duration and pitch are important parameters at short distances, and (3) that frequency modulation gains are important parameters at longer distances. Operant conditioning experiments showed that female songbirds were able to discriminate male calls at up to 128 m but not at 256 m. Finally, neurophysiological recordings showed a similar pattern in that high neural discrimination for calls was observed at 16 m and that this information degraded as a function of distance. We are currently analyzing the tuning properties of neurons that showed the most invariant responses to propagated sounds and hypothesized that these will be tuned to the parameters that we found were the most informative in the discriminant function analysis.

TUESDAY AFTERNOON, 4 JUNE 2013 — 519A, 1:00 P.M. TO 5:20 P.M.

Session 2pBAa

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

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

1:00

2pBAa1. Spatially and temporally resolved single bubble sonoluminescence and its entrainment in Rayleigh-Taylor jets. Jonathan R. Sukovich, Phillip A. Anderson (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jonsukovich@gmail.com), Ashwinkumar Sampathkumar (Frederic L. Lizzi Center for Biomed. Engineering, Riverside Research, New York, NY), and R. Glynn Holt (Mech. Eng., Boston Univ., Boston, MA)

Previous investigations of the temporal and spatial evolution of single bubble sonoluminescence (SBSL) have shown events to last on the order of tens to hundreds of picoseconds with spatial extents of less than 1 μm. Here we present observations of the temporal and spatial evolution of laser-nucleated SBSL events in a high-pressure spherical resonator. Using high-speed imaging, we observe large, long-lived SBSL events reaching diameters of up to 50 μm and lasting on the order of 30 ns. Observations of events entrained in Rayleigh-Taylor jets resulting from instabilities in the final stages of the bubbles collapses will also be presented. We observe the light emitting region entrained in these jets to reach velocities well in excess of Mach 1 and to travel up to 100 μm before being extinguished. The size and duration of events, and the velocity of those entrained in Rayleigh-Taylor jets, will be compared to the maximum radius and collapse velocity of the bubbles responsible for generating them to develop a better understanding of the dynamics leading to, and the mechanisms responsible for light emissions during highly energetic collapse events. [Work supported by Impulse Devices, Inc.]

1:20

2pBAa2. High pressure phase transitions in the fluid region surrounding the collapse point of large single bubbles in water. Jonathan R. Sukovich, Phillip Anderson (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jonsukovich@gmail.com), Ashwinkumar Sampathkumar (Frederic L. Lizzi Ctr. for Biomed. Eng., Riverside Res., New York, NY), and R. Glynn Holt (Mech. Eng., Boston Univ., Boston, MA)

Observations from imaging experiments will be presented, which have shown persistent, long-lived spherical objects to form in the fluid region surrounding large, single bubbles in highly over-pressured water. Objects have been observed to form in a region of fluid where pressures are first predicted to exceed 0.8 GPa, and to extend radially inward to where fluid pressures are predicted to reach 6 GPa. These pressures bound those requisite for transitions in water to the crystalline phases of Ice-VI and Ice-VII, at 1.1 GPa and 2.1 GPa, respectively. The objects have been observed to behave in a fashion more consistent with a highly viscous fluid. They support and recover from large shape deformations, as well as support fluid flows within them. While water does have phases which are known to exhibit properties of highly viscous fluids, they have only been observed to form at or near cryogenic temperatures, typically via hyperquenching or quasi-static pressurization at low temperatures. Here, we present evidence for a high pressure liquid-liquid phase transition in water surrounding collapsing bubbles at room temperature. [Work supported by Impulse Devices, Inc.]

1:40

2pBAa3. Experimental characterization of light emission during shock-driven cavity collapse. Phillip Anderson, Nicholas Hawker, Matthew Betney (Dept. of Eng. Sci., Univ. of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, phillip.anderson@eng.ox.ac.uk), Brett Tully (Oxyntix Ltd., Oxford, United Kingdom), Yiannis Ventikos (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), and Ronald A. Roy (Dept. of Mech. Eng., Boston Univ., Boston, MA)

The authors describe experimental work examining the collapse of a cavity by a strong shockwave. A millimeter size cavity is cast in Phytagel, which is then impacted by a metallic projectile accelerated by a compressed gas gun, reaching velocities up to 500 m/s. The impact generates a strong shockwave that propagates into the gel at greater than sonic velocity. Schlieren images are presented that illustrate both this process and the subsequent cavity collapse at a sub-microsecond timescale. As the shockwave reaches the cavity, it is shown to cause a rapid asymmetric collapse process characterized by the formation of a high-speed transverse jet. The pressure of the shockwave is found to be 100+ MPa as measured via a custom-built fiber-optic probe hydrophone. Previous work examining shock-driven cavity collapse observed luminescence, postulated to be due to the high-speed impact of the transverse jet on the far bubble wall; this experimental observation is replicated. Further, the light emission is characterized as a function of impact velocity and thus of shockwave pressure. This reveals that shock-driven cavity collapse shares many of the unique features that make the more widely studied SBSL-type collapse interesting.

2:00

2pBAa4. Simulation of warm dense matter in intense bubble collapse. Brett Tully (Oxyntix Ltd., Dept. of Eng. Sci., Parks Rd., Oxford OX1 3PJ, United Kingdom, brett.tully@oxyntix.com), Nicholas Hawker, Matthew Betney, and Yiannis Ventikos (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Previous work by the authors includes computational study of shock-bubble interaction. This work demonstrated strong compression and heating within the bubble, with gas reaching densities of order of magnitude 1 g/cc and temperatures of 10 eV. These conditions correspond to the warm dense matter regime. This paper addresses limitations of previous work through utilization of various equations of state (EOS) appropriate for the modeling of dense plasma. This is achieved through the design and implementation of a generic interface based on tabulated EOS data. Any EOS may be utilized through the framework, requiring only knowledge of pressure and energy as functions of density and temperature. The solutions to various issues such as table interpolation, tabulated change of variables, arbitrary calculation of entropy, and calculation of thermodynamic derivatives are presented. In addition, the trade-offs between CPU time, memory requirement and computational accuracy are discussed. Validation work is presented and a comparison of different EOS is also explored. The EOS used include but are not limited to the EOS for air utilized by Moss *et al*. (1994) to study SBSL, SESAME tabulated EOS and QEOS-type formulations. Finally, conditions attained during shock-bubble interaction are re-examined.

2:20

2pBAa5. Percussoluminescence. Nicholas Hawker and Yiannis Ventikos (Dept. of Eng. Sci., Univ. of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, nicholas.hawker@eng.ox.ac.uk)

The phenomenon of light emission from a bubble driven to collapse via ultrasound has a long history of study. Light emission is widely believed to occur due to the formation of plasma within the bubble during the final moment of collapse and one important focus of the literature has been on understanding the exact thermodynamic conditions created. There are, however, other phenomena that demonstrate a similar light emission, including but not limited to other types of bubble collapse. This paper examines light emission in seeming disparate phenomena and discovers that a common thread exists. The terminology and fundamental theory appropriate for sonoluminescence is found to be decreasingly relevant and a new term, percussoluminescence (PCL), is established. This is substantiated through the presentation of a 1D theory, detailed numerical simulation, and illustrative experimental results. The implications of this new perspective are explored.

2:40–3:00 Break

3:00

2pBAa6. Investigating the acoustic response of gold nanoparticle coated microbubbles. Mehrdad Azmin (UCL Mech. Eng., Univ. College London, London, United Kingdom, m.azmin@ucl.ac.uk), Paul Rademeyer, Graciela Mohamedi (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Mohan Edirisinghe (Mech. Eng., Univ. College London, London, United Kingdom), Luis Liz-Marzan (Biofunctional Nanomaterials, CIC biomaGUNE, San Sebastian, Spain), and Eleanor Stride (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Recent work has shown that incorporating solid nanoparticles into the coatings of contrast agent microbubbles can be used to control their stability and to provide other functional characteristics for example in multimodality imaging. The aim of this study was to investigate the influence of nanoparticle characteristics and concentration on the response of the microbubbles to ultrasound excitation. Theoretical models were first derived to simulate the effects of adsorbing different types and concentrations of nanoparticle on to the surface of a bubble in both a monolayer and a layer of finite thickness. The results indicate that the particles modify the symmetry of the microbubble oscillations and enhance their nonlinear character. Experimentally, microbubbles coated with a surfactant and varying concentrations of gold nanoparticles of different sizes and surface properties were produced using either sonication or microfluidics. The attenuation and backscattering coefficients from the bubble suspensions and the scattered response from individual bubbles were measured for a range of frequencies (1–7.5 MHz) and pressures (50–500 kPa). The nanoparticles were found to enhance the nonlinear character of the bubble response in agreement with the theoretical results. Both the degree of enhancement and stability of the microbubbles was dependent upon the nanoparticle surface chemistry.

3:20

2pBAa7. Investigating the sensitivity of microbubble acoustic response for biosensing applications. Caroline J. Harfield (Inst. of Biomed. Eng., Dept. of Eng. Sci., Oxford Univ., Old Rd. Campus Res. Bldg., Headington, Oxford OX3 7DQ, United Kingdom, caroline.harfield@stcatz.ox.ac.uk), Gianluca Memoli (Acoust. Group, National Physical Lab., London, United Kingdom), Philip Jones (Dept. of Phys. and Astronomy, Univ. College London, London, United Kingdom), Nick Ovenden (Dept. of Mathematics, Univ. College London, London, United Kingdom), and Eleanor Stride (Inst. of Biomed. Eng., Dept. of Eng. Sci., Oxford Univ., Oxford, United Kingdom)

Microbubbles are currently used as contrast agents for diagnostic imaging on account of their high scattering efficiency and non-linear response to ultrasound. The exact nature of this response depends not only upon the bubble size and imposed sound field but also the bubble environment: physical properties of the surrounding liquid, bubble surface coating, ambient temperature, pressure, and proximity to other bubbles or surfaces. This dependence can potentially be exploited for the microscale interrogation of a liquid to detect, e.g., changes in viscosity or the presence of particular chemical species. To facilitate this, the sensitivity of the microbubble acoustic response to changes in its environment must be analyzed. The aim of this study was to provide a theoretical framework for this. A modified Rayleigh-Plesset equation was derived to describe the radial bubble motion, including the effects of gas diffusion and adsorption/desorption of a surfactant coating, and coupled to an equation describing microbubble translation. The presence of a rigid boundary was also included in the simulations. A sensitivity analysis was performed for the effect of each of the physical variables upon the bubble response, which indicated high sensitivity to species altering the dynamic surface tension and proximity to a boundary.

3:40

2pBAa8. Modeling of microbubbles pushed through clots via acoustic radiation force. Ascanio Guarini and E. C. Everbach (Engineering, Swarthmore College, 500 College Ave., Swarthmore, PA 19081, ceverba1@swarthmore.edu)

Previous studies have shown that thrombi, which may completely block the blood flow in a vessel, can be dissolved by ultrasound acting on echo-contrast agent microbubbles. The presumed mechanism is acoustic cavitation, the radial oscillations of the bubbles, which can exert locally large forces on the fibrin ropes that make up the clot matrix. However, the movement of the bubbles through the clot in the absence of flow suggests that acoustic radiation force also plays an important role. Because detailed mechanistic modeling of this process is not available, we present here a heuristic study in which microbubble transit times in gels of various porosities were measured and described by a simplified percolation theory. Results suggest considerations for optimizing the penetration of active microbubbles in sonothrombolysis.

4:00

2pBAa9. Use of dolphin-like pulses to enhance target discrimination and reduce clutter. Tim Leighton, Gim-Hwa Chua, Paul R. White (Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton, Hampshire SO17 1BJ, United Kingdom, tgl@soton.ac.uk), Kenneth Tong, Hugh Griffiths (Dept. of Electron. and Elec. Eng., Univ. College London, London, United Kingdom), and David Daniels (Cobham Tech. Services, Leatherhead, United Kingdom)

Building on the earlier success of using twin inverted pulses to supress bubble scatter and so reduce clutter when detecting targets in bubbly water (proven in ship wakes in field trials), the same nonlinear processing scheme is generalized to make use of dolphin-like pulses. Their performance in reducing clutter and enhancing target discrimination is demonstrated in the laboratory, and the opportunities for using the same scheme to improve the detection of hidden electronic devices or semiconductor devices by radar are discussed.

4:20

2pBAa10. Acoustic and optical observations of methane gas seeps in the Gulf of Mexico. Thomas C. Weber, Yuri Rzhanov, Kevin Jerram, Larry Mayer (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, weber@ccom.unh.edu), and Dave Lovalvo (Eastern Oceanics/NOAA Office of Ocean Exploration, West Redding, New Hampshire)

In 2011 and 2012, measurements of acoustic backscatter from natural methane seeps were made in the northern Gulf of Mexico in water depths between 1000 and 2000 m. The measurements were made using a calibrated 18 kHz echo sounder with an 11 degree beamwidth in order to estimate the depth-dependent target strength (TS). The TS data indicate a wide variation in the rate of gas seepage from the seafloor. Several of these seeps were revisited with a remotely operated vehicle in order to optically assess the bubble size distribution and to estimate the rate at which gas bubbles were exiting the seafloor. The optical data show bubble sizes between 1 and 10 mm radius, and similar rates of gas seepage ranging from a few bubbles per second to several tens of bubbles per second. Together, these data help to suggest the requirements for acoustically estimating gas flux from the seafloor over large regions.

4:40

2pBAa11. Effect of shell thickness on sound propagation through encapsulated bubbles: A resonator approach. Craig N. Dolder and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dolder@utexas.edu)

The acoustics of elastic-shelled bubbles are of interest in applications ranging from biomedical imaging, to fisheries acoustics, to underwater noise abatement. Multiple models exist to predict the velocity and attenuation of sound propagating through water containing encapsulated bubbles, but existing measurements have yet been able to confidently verify model accuracy for void fractions above about 10^{-3}. The effect of shell thickness was studied in this work using tethered, rubber-encapsulated bubbles bearing shells of varying thickness, deployed within a one-dimensional resonator apparatus. In previous work, resonator modes below the individual bubble resonance frequency (IBRF) were exploited to extract inferences of sound speed, but this is a regime where there is little difference between competing model prediction. In the present work, an increased understanding of the modal field inside the resonator has extended this technique to just below IBRF and to well above IBRF, both regimes where model behavior diversifies, thus providing a new opportunity for model verification. Measurement-model comparisons will be shown for encapsulated bubbles with radii ranging from 13 mm to 30 mm, void fractions ranging from 8.4×10^{-4} to 1.1×10^{-2}, and shell thicknesses ranging from 0.085 to 0.16 mm. [Work supported by the Office of Naval Research.]

5:00

2pBAa12. Attenuation of sound in water through collections of very large bubbles with elastic shells. Kevin M. Lee (Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu) and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Lab., The Univ. of Texas at Austin, Austin, TX)

The ultimate goal of this work is to accurately predict the attenuation through a collection of large (on the order of 10 cm radius) tethered encapsulated bubbles used in an underwater noise abatement system. Measurements of underwater sound attenuation were performed during a set of lake experiments, where a low-frequency compact electromechanical sound source was surrounded by different arrays of encapsulated bubbles with various individual bubbles sizes and void fractions. The measurements are compared with an existing predictive model [J. Acoust. Soc. Am. **97**, 1510–1521 (1995)] of the dispersion relation for linear propagation in liquid containing encapsulated bubbles. Although the model was originally intended to describe ultrasound contrast agents, it is evaluated here for large bubbles, and hence low frequencies, as a design tool for the underwater noise abatement system, and there is fairly good quantitative agreement between the data and the model. Refinements to the model to incorporate multiple scattering effects, which may be important at high void fractions, via an effective medium approach [J. Acoust. Soc. Am. **111**, 168–173 (2002)] and comparison with the data will also be discussed. [Work supported by Shell Global Solutions.]

TUESDAY AFTERNOON, 4 JUNE 2013 518C, 1:20 P.M. TO 3:20 P.M.

Session 2pBAb

Biomedical Acoustics: Imaging and Characterization in Elastic Media

Guy Cloutier, Chair

Lab. of Biorheology and Med. Ultrasonics, Univ. of Montreal Hospital, 2099 Alexandre de Sève, Montreal, QC H2L 2W5, Canada

Contributed Papers

1:20

2pBAb1. Dynamic quantitative ultrasound imaging of mimicked breast lesions during shear wave propagation to emphasize differences in tissue statistical backscatter properties. Marzieh Alavi (Lab. of Biorheology and Med. Ultrasonics, Univ. of Montreal Hospital, Univ. of Montreal Hospital Res. Ctr., Montreal, QC, Canada, marziehalavi@gmail.com), Francois Destrempes, Emmanuel Montagnon, and Guy Cloutier (Lab. of Biorheology and Med. Ultrasonics, Univ. of Montreal Hospital, Montreal, QC, Canada)

The main motivation in this study was to increase the accuracy of the breast tissue characterization by combining quantitative ultrasound (QUS) with ultrasound (US) dynamic elastography. To demonstrate that, an agar-gelatin breast mimicking phantom with two inclusions containing the same density of agar (US scatterers) but different proportions of gelatin corresponding to different mechanical properties was made. Transient plane shear waves (SW) at 200 Hz were transmitted through the phantom while the displacement of scatterers was imaged at 5 MHz with an ultrafast imaging technique. With segmented inclusions, the reciprocal (beta parameter) of the effective density of scatterers of a general distribution model of the echo envelope and its normalized range (normalized by the mean of beta during SW propagation) were estimated for each inclusion. The results showed that the relative difference of beta between the surrounding medium and both inclusions A and B were 55.6% (A) and 0.9% (B), respectively, whereas differences (in %) of the beta normalized range were 46.2% (A) and 52.6% (B), respectively. The static value beta failed to distinguish inclusion B from the surrounding; however, the dynamic range of beta succeeded in that task for the two inclusions. Thus, dynamic QUS might add information to QUS in a static framework.

1:40

2pBAb2. Shear wave elastography for characterizing muscle tissue in myofascial pain syndrome. Diego Turo, Paul Otto (Dept. of Bioengineering, George Mason Univ., 4400 University Dr., Fairfax, VA 22030, diego-turo@gmail.com), Tadesse Gebreab (National Inst. of Health, Bethesda, MD), Katherine Armstrong, Lynn H. Gerber (Dept. of Rehabilitation Sci., George Mason Univ., Fairfax, VA), and Siddhartha Sikdar (Dept. of Elec. and Comput. Eng., George Mason Univ., Fairfax, VA)

Myofascial pain syndrome (MPS) affects 85% of chronic pain sufferers in a specialty pain center. Neck and low-back are commonly affected by MPS. Myofascial trigger points (MTrPs) are characteristic findings of MPS and are palpable tender nodules in the muscles of symptomatic subjects. Mechanical characterization of MTrPs and surrounding tissue can offer important insight about the pathophysiology of the MPS, which is currently poorly understood. In this study, we propose an inexpensive technique, based on ultrasound shear wave elastography, to objectively measure mechanical properties of MTrPs and surrounding tissue in the upper trapezius. In an ongoing clinical study, we recruited 34 subjects: 12 healthy controls, 10 with not spontaneously painful MTrPs (latent) and 12 with symptomatic chronic neck pain (>3 months) and active (spontaneously painful) MTrPs. Shear wave elastography was performed on the upper trapezius of all subjects using the Ultrasonix RP system and an external vibrator. Voigt' s model was used to estimate shear modulus G and viscosity μ of the interrogated tissue. Preliminary analysis demonstrates that symptomatic muscle tissue in subjects with neck pain is stiffer ($G = 8.40 \pm 8.31$ kPa, mean $\pm$ standard deviation) compared to muscle in control subjects ($G = 2.86 \pm 2.48$ kPa) ($p < 0.05$), and that active MTrPs are more viscous ($\mu = 17.10 \pm 9.46$ Pa*s) than surrounding tissue ($\mu = 10.59 \pm 5.96$ Pa*s). Latent MTrPs ($\mu = 24.31 \pm 12.72$ Pa*s) and surrounding tissue ($\mu = 20.09 \pm 7.48$ Pa*s) are more viscous than normal tissue ($\mu = 10.96 \pm 4.17$ Pa*s).

2:00

2pBAb3. Electromagnetic-acoustic imaging of stiffness and dielectric properties in gels. Ning Zhang, Robin Cleveland, and David J. Edwards (Eng. Sci., Univ. of Oxford, Inst. Biomed. Eng., Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, robin.cleveland@eng.ox.ac.uk)

Here we present a novel multi-modal imaging method, electromagnetic-acoustics (EMA) that combines acoustic radiation force and electromagnetic scattering. Experiments were carried out in gels using a focused megahertz ultrasound source, amplitude modulated at 160 Hz, to create an oscillating radiation force resulting in vibration of the gel in the focal region. At the same time an EM signal was transmitted into the gel and the backscattered EM signal recorded. The tissue that was in motion resulted in a Doppler shift of the EM signal which manifested itself as frequency modulation of the EM wave. The modulated component was detected by means of a demodulator and lock-in amplifier and the amplitude of the modulated signal we call the EMA signal. By steering the ultrasound beam through the sample, an EMA image can be created which has spatial resolution of the ultrasound but is sensitive to shear and dielectric properties. We show that the EMA signal is sensitive to changes in elasticity and conductivity in a gel. EMA may have utility in biomedical imaging by detecting diseases, which have contrast in dielectric properties without the cost and complexity of an magnetic resonance imaging system.

2:20

2pBAb4. Speckle tracking in multiple-scattered pressure field. Ayse Kalkan-Savoy (Biomed. Eng. and Technol., UMass-Lowell, 1 University Ave., Lowell, MA 01854, ayse.k.savoy@gmail.com) and Charles Thompson (Elec. and Comput. Eng., UMass-Lowell, Lowell, MA)

Speckle tracking using ultrasound B-scan image sequences to quantify myocardial strain has the potential to become a standard method in echocardiography. A pressure field for a modulated Gaussian pulse traveling in inhomogeneous media is constructed. This model is based on computation of pressure field using Kirchoff integral formulation in frequency domain. Each pressure term of Neumann Series is converted into time domain and used in computation of Pade approximants. Pressure field in time and space is constructed utilizing Pade approximants and includes multiple-scattering effects. The scatterers of interest are large enough in size and contrast to contribute to strong and multiple-scattering. The signals at transducer point for each time frame are analyzed and speckles are tracked utilizing an optical flow algorithm. A velocity field which can be utilized to quantify strain is computed and visualized. The efficiency of this method will be discussed.

2:40

2pBAb5. Validation of three-dimensional strain tracking by volumetric ultrasound image correlation in a pubovisceral muscle model. Anna S. Nagle, Ashok R. Nageswaren, Balakrishna Haridas, and T. Douglas Mast (SEEBME, Univ. of Cincinnati, 3940 CVC, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, nagleas@mail.uc.edu)

Little is understood about the biomechanical changes leading to pelvic floor disorders such as stress urinary incontinence. In order to measure regional biomechanical properties of the pelvic floor muscles *in vivo*, a 3D strain tracking technique employing correlation of volumetric ultrasound images has been implemented. In this technique, local 3D displacements are determined as a function of applied stress and then converted to strain maps. To validate this approach, an *in vitro* model of the pubovisceral muscle, with a hemispherical indenter emulating the downward stress caused by intra-abdominal pressure, was constructed. Volumetric B-scan images were recorded as a function of indenter displacement while muscle strain was measured independently by a sonomicrometry system (Sonometrics). Local strains were computed by ultrasound image correlation and compared with sonomicrometry-measured strains to assess strain tracking accuracy. Image correlation by maximizing an exponential likelihood function was found more reliable than the Pearson correlation coefficient. Strain accuracy was dependent on sizes of the subvolumes used for image correlation, relative to characteristic speckle length scales of the images. Decorrelation of echo signals was mapped as a function of indenter displacement and local tissue orientation. Strain measurement accuracy was weakly related to local echo decorrelation.

3:00

2pBAb6. Measurement of surface acoustic wave in soft material using swept-source optical coherence tomography. Yukako Kato, Yuji Wada, Yosuke Mizuno, and Nakamura Kentaro (Tokyo Inst. of Technol., 4259-R2-26 Nagatsudacho Midoriku, Yokohama 226-8503, Japan, ykato@sonic.pi.titech.ac.jp)

In endoscopic elastography, it is needed to observe small area with high spatial resolution. Optical coherence tomography (OCT) is one of the candidate imaging methods, which has the depth resolution of several 10 μm. In this study, we try to find the propagation velocity of surface acoustic wave (SAW) using a swept-source OCT (SS-OCT). The depth scanning rate in the SS-OCT is rather fast, which is determined by the wavelength sweep of the light source as fast as 20–100 kHz. However, on the other hand, the lateral scanning is limited up to 100 times per second, since it is performed using a mechanical moving mirror. We develop a theory to estimate SAW velocity of tissues ranging from 1 to 20 m/s from data taken by the slow lateral scanning of less than 1 m/s using the OCT. The present method is tested for agar samples with different concentrations and also for several tissue samples. Vibrations are excited on the sample surface using a small stick connected to a loudspeaker. The measurements are carried out at many frequencies from 500 to 1000 Hz. The dependence of the SAW velocity on the concentration successfully agreed the previous results.

Session 2pEAa

Engineering Acoustics: Computational Methods in Transducer Design, Modeling, Simulation, and Optimization I

Daniel M. Warren, Chair
Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60134

Chair's Introduction—12:55

Invited Paper

1:00

2pEAa1. Virtual prototyping of condenser microphones using the finite element method for detailed electric, mechanic, and acoustic characterization. Mads Jakob Herring Jensen (COMSOL A/S, Diplomvej 373, Lyngby 2800, Denmark, mads@comsol.dk) and Erling Sandermann Olsen (Brüel & Kjaer Sound and Vib. Measurement A/S, Naerum, Denmark)

Recent development and advances within numerical techniques and computers now enable the modeling, design, and optimization of many transducers using virtual prototypes. Here, we present such a virtual prototype of a Brüel & Kjær Type 4134 condenser microphone. The virtual prototype is implemented as a model using the finite element method with COMSOL Multiphysics and includes description of the electric, mechanic, and acoustic properties of the transducer. The acoustic description includes thermal and viscous losses explicitly solving the linearized continuity, Navier-Stokes, and energy equations. The mechanics of the diaphragm are modeled assuming a pre-stressed membrane, electrostatic attraction forces, and acoustic loads. The model includes electric description of the active and passive capacitances of the microphone cartridge as well as an external circuit model representing the preamplifier. Different modes of the system are studied, including the important first rocking mode of the membrane. The model has no free fitting parameters and results in the prediction of the frequency dependent sensitivity, capacitance, and mechanical impedance. The model results show good agreement with measured data.

Contributed Papers

1:20

2pEAa2. New planar nano-gauge detection microphone: Analytical and numerical acoustic modeling. Cécile Guianvarc'h, Thierry Verdot (LVA-Insa de Lyon, Villeurbanne, France), Jaroslaw Czarny (CEA-Leti, Grenoble, France), Emmanuel Redon, Kerem Ege, Jean-Louis Guyader (LVA-Insa de Lyon, 25 Bis, avenue Jean Capelle, Villeurbanne 69621, France, emmanuel.redon@insa-lyon.fr), Arnaud Walther, and Philippe Robert (CEA-Leti, Grenoble, France)

The miniaturization of microphones is of great interest for several fields, such as medical applications (audio implants), or consumer electronics (cell phones). Almost all existing miniature microphones rely on electrostatic transduction and offer good performances (sensitivity, frequency bandwidth). However, their sensitivity, proportional to the membranes area, would be dramatically reduced in case of extreme miniaturization. A new concept of microphones developed by CEA-LETI, which uses membranes moving in the plane of the substrate and inducing strain on piezoresistive Si nano-gauges (M&NEMS technology), seems promising for its miniaturization potential without significant decrease of sensitivity. The design and optimization of such planar piezo-resistive microphone require a deep understanding of its acoustic and vibroacoustic behavior. Regarding the small dimensions of the slits (1–100μm) and the sharp discontinuities in the microphones structure, viscous and thermal effects in the boundary layers and turbulent perturbations are of great importance, and must then be taken into account with high accuracy in device modeling. The aim of the present work is to provide accurate analytical and numerical (FEM) models able to gather all these effects in a consistent manner, and to suggest an experimental method to check their validity.

1:40

2pEAa3. Noise minimization in micromachined piezoelectric microphones. Robert Littrell (Baker-Calling, Inc., 1810 14th St., Ste. 210, Santa Monica, CA 90404, rlittrell@bakercalling.com) and Karl Grosh (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Piezoelectric MEMS microphones have been built for more than 30 years and they offer some advantages over other technologies such as improved linearity, simple construction, no need for a bias voltage or charge, and the ability to withstand high temperatures. Despite these advantages, a relatively high noise floor has always limited their utilization. Traditionally, the noise of these sensors has been minimized by viewing the preamplifier or amplifier as a black box with fixed gain and noise. This leads the designer to minimize noise by maximizing microphone sensitivity. By viewing the microphone-amplifier system together, we develop a different method of optimization, leading to lower noise. Further, by including the back cavity compliance of the package in the optimization, we can determine absolute limits on the minimum achievable noise floor with very few assumptions. To date, we have built piezoelectric MEMS microphones utilizing aluminum nitride with a 32 dBA noise floor. We can compute a minimum achievable noise floor of 24 dBA for the same sensing structure with a 2 mm^3 back cavity volume.

2:00

2pEAa4. Directional robustness of an *in-situ*, dual dipole omni microphone array for hearing instruments. Thomas Burns (Starkey Hearing Technol., 6600 Washington Ave. S, Eden Prairie, MN 55344, tburns@starkey.com)

The three critical factors for providing stable directional performance for typical microphone arrays used in hearing aids include the relative sensitivity and phase between the microphones in addition to the placement of the hearing instrument behind the user's ear. A directional system is robust if these factors can operate over a wide range of levels without degrading the directional performance. In this study, dual dipole microphones were arranged symmetrically around an omnidirectional microphone such that all inlets were collinear. Compared to an endfire array, whether it be a delay-and-sum or Blumlein configuration, this dual-dipole-omni array is remarkably more robust, yielding very little degradation in the Directivity Index for the aforementioned critical factors varying as much as +/− 3 dB, +/− 30 ms, and the directional axis of the hearing instrument varying +/− 20 degrees on the ear.

2:20

2pEAa5. The efficiency of receivers used in hearing aid devices. Michael Salameh (Research, Starkey Hearing Technol., 6600 Washington Ave. S., Eden Prairie, MN 55344, michael_salameh@starkey.com)

Current drain is one of the most important considerations in the hearing aid design. It is determined mainly by the receiver impedance. Recently, many additional features and technologies such as wireless are used in the hearing aid devices. These capabilities increase the current drain further. Efficiency evaluation becomes an important step in the receiver selection for hearing aid devices. Balanced armature receivers are widely used in hearing aids. In comparison with the moving-coil loudspeakers, these receivers are designed for better efficiency in closed acoustical loads found in different hearing aid styles. However, the receiver efficiency for hearing aid applications is not well defined, evaluated, or reported. In this paper, the receiver efficiency including the effect of the receiver size, impedance, and acoustical load is discussed. The evaluation of the receiver efficiency is revisited and a new approach is suggested. The simulation and the measurement results are presented.

2:40

2pEAa6. Cantilever mode piston transducer array. John Butler and Alexander L. Butler (Image Acoust., Inc., 97 Elm St., Cohasset, MA 02025, jbutler@imageacoustics.com)

The underwater sound cantilever mode transducer uses a piston, tail mass, dual piezoelectric cantilever benders, and mechanical arms for translating lateral bending cantilever motion to piston motion in a direction normal to the cantilever motion. This compact design provides a wide band response and exceptionally low resonance, especially under array loaded conditions. We present the operation of the cantilever mode transducer along with its unique response under array loading conditions where the resonance frequency decreases as the array gets larger. These results are demonstrated through a number of finite element array models and measured results on a 2x2 array. The single crystal PIN-PMN-PT results show a significant source level improvement over ceramic PZT-8 active material when driven up to 5 V/mil.

3:00

2pEAa7. Applications of network synthesis and zero-pole analysis in transducer modeling. Daniel Warren (R&D, Knowles Electron., 1151 Maplewood Dr., Itasca, IL 60143, daniel.warren@knowles.com)

When a transducer is an immutable component of a larger system being simulated, it is sufficient that the transducer model correctly reproduce behavior only at the available ports of the transducer. The behavior of two-port electroacoustic transducer should be completely characterized by three transfer functions related to the electrical and acoustic termination impedance and the transfer impedance. To the extent that the transducer could be represented as a analog circuit of passive linear elements, the same circuit could be exactly represented by rational polynomials of the Laplace *s*, embodied as a set of poles and zeros of the transfer functions. This invites the reverse process of identifying poles and zeros by nonlinear curve fit of rational polynomials to measured transfer data, perhaps even synthesizing a circuit directly from the identified poles and zeros. Measured transducer transfer data have been fit demonstrating both the utility and the pitfalls of this method. Curve fit transfer functions can be a compact and faithful representation of complex data over frequency, but have no predictive value outside the given data. Judicious selection of the number of poles and zeros, initial values, proper constraints, and some physical insight are necessary for stable curve fits. Further investigation into the relationship among the transfer functions of a physical system may lead to a meaningful model derived from curve fits.

3:20

2pEAa8. Improved estimation of direction of arrival of sound sources for hearing aids using gyroscopic information. Alan W. Boyd (Dept. of Electron. and Elec. Eng., Univ. of Strathclyde, 204 George St., Glasgow G1 1XW, United Kingdom, alan.boyd@strath.ac.uk), William M. Whitmer, W. Owen Brimijoin, and Michael A. Akeroyd (Inst. of Hearing Res., Med. Res. Council, Glasgow, United Kingdom)

Determining the direction of arrival (DOA) of a sound source is important in spatial audio signal processing, as it can lead to substantial improvement in noise reduction performance. Techniques such as generalized cross correlation with phase transform (GCC-PHAT) and adaptive eigenvalue decomposition (AED) perform optimally when the measurement microphones are fixed in place. However, hearing-aid microphones move with the listener's head movements, which can result in momentarily inaccurate directional estimates and noise artifacts in the output signal. Techniques such as GCC-PHAT experience degraded short-term performance in the presence of multiple signals and noise. The system presented measures instantaneous head movement velocity using a micro-electromechanical systems (MEMS) gyroscope attached to binaurally communicating hearing aids. Estimates of DOA for physically stationary sources are shifted based on the gyroscope's head-movement information. Using GCC-PHAT with gyroscopic input can produce robust *in situ* DOA estimates for several sources in reverberant environments. In addition, the gyroscope allows an adaptive beamformer to be steered to a target direction, compensating for head movements on a very short timescale during DOA estimation. Results show improved localization performance over a standard GCC-PHAT system during head movements.

3:40

2pEAa9. Miniaturized electrostatic receiver with small-sized backing electrode. Alexey Podkovskiy, Petr Honzík, Stéphane Durand, Nicolas Joly, and Michel Bruneau (LUNAM Université, LAUM (Laboratoire d'acoustique de l'université du Maine), UMR CNRS 6613, Avenue Olivier Messiaen, Le Mans F-72085, France, petr.honzik@gmail.com)

A miniaturized electrostatic receiver design, having a central cylindrical backing electrode of small radius surrounded by a flat annular cavity behind the circular membrane, can lead to both a higher sensitivity and a larger frequency bandwidth compared to the ones achieved with other designs, while bringing a geometrical simplicity which is advantageous from the point of view of microfabrication. An appropriate computational method, relying on a specific 2-D axisymmetrical simulation using an adaptive mesh and accounting for both viscous and thermal boundary layer effects, provides results against which analytical results can be tested. An analytical approach, which leads to solutions based on the eigenmode expansion of the membrane displacement, the acoustic pressure field depending on the radial coordinate in the central fluid gap but being assumed quasi-uniform in the annular cavity, is much faster in terms of running time and appears to be sufficiently accurate to achieve final optimization of this kind of devices.

4:00

2pEAa10. How a hearing aid transducer works. Noori Kim and Jont B. Allen (UIUC, 1085 Baytowner df 11, Champaign, IL 61822, nkim13@illinois.edu)

The oldest magnetic earphone, the balanced armature receiver (BAR), is the most widely used receivers in modern hearing-aid instruments, where the efficiency of the power (battery life) and the size of the device, as well as the larger frequency bandwidth, are critical parameters. Since these miniature loudspeakers remain one of the most expensive components of the hearing-aids, a detailed studying of them is a cornerstone of understanding the hearing-aid system, and we believe that the appropriate and rigorous analysis of this transducer is critical. The motivation of this study started from the modeling of a widely used commercial hearing-aid receiver ED series, manufactured by Knowles Electronics, Inc. Our proposed model includes a semi-inductor and a gyrator along with the two-port network glue which enables us an intuitive design of the electromagnetic transducer. Based on the BAR model, we will investigate and discuss the roles of each physical component in the BAR such as a coil, magnets, an armature, a diaphragm, and the rear volume of the receiver. Ultimately, this work will deliver a fundamental and innate answer for the question, "How does a hearing-aid transducer work?"

4:20

2pEAa11. Numerical study of the cross-talk effects in acoustical transducer arrays and correction. Abdelmajid Bybi, Jamal Assaad (OAE Dept., CNRS UMR 8520, IEMN, Univ. of Valenciennes and Hainaut Cambrésis, Valenciennes, France), Anne-Christine Hladky-Hennion (ISEN Dept., UMR CNRS 8520, IEMN, Lille, France), Farouk Benmeddour, Sébastien Grondel, and Frederic Rivart (OAE Dept., CNRS UMR 8520, IEMN, Univ. of Valenciennes and Hainaut Cambrésis, Campus Mont Houy, Valenciennes 59313, France, farouk.benmeddour@univ-valenciennes.fr)

Cross-talk in acoustical transducer arrays is an undesirable phenomenon which decreases seriously the performances of these sensors. Indeed, when one element of the array is driven, it generates parasitic displacement fields at the radiating surfaces of the neighboring elements, which changes the directivity of the antenna. To well understand this phenomenon a transducer array similar to those used in medical imaging and NDT applications is modeled by finite element method. The research work, investigated systematically the effects of the cross-talk. First, it inspected the acoustical and mechanical cross-talk throughout the propagating medium and the filling material. Second, it studied the influence of the matching layer and the backing on the acoustical performances of the transducer. It is shown that the filling material and the matching layer are the major factor contributing to this phenomenon. In order to cancel the cross-talk a correction method previously developed by the author has been used. This solution consisted in applying adapted electrical voltages on each neighboring element of the active one in the purpose to reduce the displacement field on their active surface. This method was tested numerically and the obtained results clearly demonstrated its ability to reduce the cross-talk.

4:40

2pEAa12. Development and performance evaluation of virtual auditory display system to synthesize sound from multiple sound sources using graphics processing unit. Kanji Watanabe (Faculty of Systems Sci. and Technol., Akita Prefectural Univ., 84-4 Ebinokuchi, Tsuchiya, Yuri-Honjo, Akita 015-0055, Japan, kwatanabe@akita-pu.ac.jp), Yusuke Oikawa (Grad. School of Systems Sci. and Technol., Akita Prefectural Univ., Yuri-Honjo, Japan), Sojun Sato, Shouichi Takane, and Koji Abe (Faculty of Systems Sci. and Technol., Akita Prefectural Univ., Yuri-Honjo, Japan)

Head-related transfer function (HRTF) is characterized as sound transmission from sound source to listener's eardrum. When a listener hears a sound that is filtered with the HRTFs, the listener can localize a virtual target (sound image) as if the sound had come from the position corresponding to that at which the HRTFs were measured. A moving sound image can be generated to switch HRTFs of successive direction in real-time. While many virtual auditory displays (VADs) based on synthesis of HRTFs have been proposed, most of them can synthesize only a few sound images due to lack of computation power. In this article, the VAD system implemented based on graphics processing unit (GPU) was introduced. In our system, the convolution of HRTFs is parallelized on GPU to realize a high-speed processing. In addition, the multiple HRTFs each of which is corresponding to sound sources at different position are processed in parallel to control multiple sound image simultaneously. In this article, the performance of our system was evaluated not only objectively but also subjectively. The results showed that our current system can present at least 40 sound images simultaneously in real-time.

TUESDAY AFTERNOON, 4 JUNE 2013 512BF, 1:00 P.M. TO 5:00 P.M.

Session 2pEAb

Engineering Acoustics: Controlling Sound Quality

Stephen Butler, Chair
NUWC, Newport, RI 02841

Contributed Papers

1:00

2pEAb1. Subjective and objective evaluation of sound quality of radio programs transmitted via Digital Audio Broadcast (DAB+) System. Andrzej B. Dobrucki and Maurycy J. Kin (Chair of Acoust., Wroclaw Univ. of Technol., Wybrzeze Wyspianskiego 27, Wroclaw 50-370, Poland, andrzej.dobrucki@pwr.wroc.pl)

The work presents results of research on the sound quality of different radio programs transmitted via Digital Audio Broadcasting (DAB+). This assessment has been provided with a use of psychoacoustic model as well as standard listening tests, using an Absolute Category Rating (ACR) method of scaling, and Comparison Category Rating (CCR) method. Results have shown that sound quality gets worse when bit-stream is of the lowest values (48 kbit/s or 24 kbit/s). Application of the Spectral Band Replication processor significantly improves the perceived quality, which is satisfying for bit streams higher than 64 kbit/s, particularly for jazz and popular music. The assessment with ACR method (recommended for broadcast by International Telecommunication Union) showed better notes than CCR one. It means that recommended method is less critical. Also results obtained with psychoacoustic model are more similar to obtained with CCR method. The attributes of spatial impression change in different ways. The greatest distortion has been observed for the perspective and spaciousness of sound image, while the sound color as well as localization stability and accuracy of phantom sources remained almost the same.

1:20

2pEAb2. Objective evaluation of sound quality for attacks on robust audio watermarking. Akira Nishimura (Media and Cultural Studies, Tokyo Univ. of Information Sci., 4-1, Onaridai, Wakaba-ku, Chiba 2658501, Japan, akira@rsch.tuis.ac.jp), Masashi Unoki (School of Information Sci., Japan Adv. Inst. of Sci. and Technol., Nomi, Japan), Kazuhiro Kondo (Grad. School of Sci. and Eng., Yamagata Univ., Yamagata, Japan), and Akio Ogihara (Faculty of Eng., Kinki Univ., Higashihiroshima, Japan)

Various attacks on robust audio watermarking have been proposed. Reversible signal processing attacks, such as sampling frequency conversion, degrade sound quality of the distributed watermarked audio (stego audio)

and disturb extraction of hidden data so that copyright detection systems using automated crawling are invalidated. Reversible signal processing of the attack can recover sound quality of the degraded audio data. In order to prove validity and security of audio watermarking system, analysis of the presumed attacks or reversible signal processing on stego audio, is required. However, these attacks on audio signal also degrade sound quality of commercial music where such pieces of music are considered to be not suitable for appreciation. Therefore, degradation of sound quality induced by various attacks should be taken into account to decide if the intensity of the attacks are realistic. In this study, objective audio quality measurement (PEAQ) was applied to the audio signals including typical perceptual coding, MP3, tandem MP3, MPEG4AAC, and reversible signal processing of sampling frequency conversion, noise addition, frequency shift, bandpass filtering, and echo addition. The results indicate requirements for robustness and criteria of the attacks on high quality and robust audio watermarking technology.

1:40

2pEAb3. Measurement of acoustic transmission properties of a handset with a piezoelectric vibrator using a head and torso simulator. Toshiharu Horiuchi and Tsuneo Kato (User Interface Lab., KDDI R&D Lab., Inc., 2-1-15 Ohara, Fujimino, Saitama 356-8502, Japan, to-horiuchi@kddilabs.jp)

This paper presents equalizing an acoustic transmission property from a handset with a piezoelectric vibrator at a pinna to an eardrum with that of a normal headphone by hearing the measured sound through a head and torso simulator (HATS). Recently, a piezoelectric vibrator that vibrates a pinna to produce sounds was adopted as a receiver of smartphones to improve perceived quality within noisy environments. The HATS, used for handset testings in accordance with ITU-T recommendations, has a silicon-rubber pinna simulator to reproduce realistic acoustic properties with its human-like shape and stiffness. However, the handset with the built-in piezoelectric vibrator is beyond the scope and was not tested on the HATS before. This paper clarifies the difference of frequency responses between the pinna simulator and the real pinna based on a subjective assessment that adjusts the loudness of pure tones through the pinna simulator to be equalized auditorily to those through the real pinna. We used B&K HATS Type 4128-D. The results indicated a flat response for the pinna simulator, while a low-pass-like response for the real pinna with a cutoff at 1.5 kHz. Thereby, the actual sound can be simulated from the sound measured by the HATS with the responses.

2:00

2pEAb4. The effect of firefighting protective equipment on head related transfer functions. Joelle I. Suits, Theodore F. Argo (Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E Dean Keaton St., Austin, TX 78712, jsuits@utexas.edu), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Lab., The Univ. of Texas at Austin, Austin, TX), Ofodike A. Ezekoye (Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Craig A. Champlin (Austin, Texas)

Personal Alert Safety System (PASS) devices are used in the fire service to locate trapped or injured personnel. When a firefighter becomes incapacitated, the device emits an audible alarm to help rescue teams locate the downed firefighter. These devices have been successful, but there are still cases in which PASS is not effective, and the present project seeks to provide science-based guidance for improvements to PASS. One part of this complex problem is the effect of the protective equipment (helmet, eye protection, hood, coat) that is worn by firefighters on hearing. Since this has not previously been studied, it has not been accounted for in the current design of the PASS signal. To address this deficiency, head related transfer function (HRTF) measurements have been taken with a KEMAR acoustic mannequin wearing various combinations of the aforementioned equipment. Results indicate a reduced received level at the ear when the full complement of gear is worn, as might be expected, potentially causing a reduced detection range. In addition, the helmet and eye protection devices cause significant disruption of the normal HRTF patterns, which could potentially interfere with localization. [Work supported by U.S. Department of Homeland Security Assistance to Firefighters Grants Program.]

2:20

2pEAb5. An escape guiding system utilizing the precedence effect for evacuation signal. Takahiro Fujikawa and Shigeaki Aoki (Electron. Information and Commun. Eng., Kanazawa Inst. of Technol., 7-1 Ohgigaoka, Nonoichi, Ishikawa 921 - 8501, Japan, aoki_s@neptune.kanazawa-it.ac.jp)

This research aims at building an escape guiding system using audio signals. The system utilizes the precedence effect for the listener to perceive easily the direction of an emergency exit. In one of the ordinary escape guiding systems, two or more loudspeakers are set on the ceiling of a passageway. Since the precedence effect is generated by delaying suitably the audio guidance signal radiated from the loudspeakers, the proper direction of the emergency exit is recognized. However, the effective listening area is limited. That is, the ordinary escape guiding systems is not effective in areas under loudspeakers. In this paper, a new method that the loudspeakers are set beside in a passageway is proposed. The generation and disappearance of the precedence effect of an audio signal in the new configuration are investigated. In the listening tests, the time delay and the difference of intensity between loudspeakers are parameters. The installation angle of the loudspeaker is another parameter. Male voice and female voices are used as emergency guidance. The guidance effect of the configuration in setting beside the loudspeaker is confirmed and the test results are discussed.

2:40

2pEAb6. Reduction of sound leakage in handheld devices using open loop control. GunWoo Lee, Aran Cha, SeoungHun Kim, YoungTae Kim (Samsung Electron. Co., Ltd., 416, Martan 3-dong, Yeongtong-gu, Suwon-si 443-742, South Korea, gw325.lee@samsung.com), and JungWoo Choi (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea)

Recently, handheld devices with sound functionality are popular. A problem encountered by using such devices is sound leakage due to inappropriate volume setting, which should be reduced not to disturb people around the user. In previous study, we presented an active control (ANC) technique to control such sound leakage in handheld devices. The least squares (LS) optimum control under various positions of error sensors was investigated. Based on the results, desirable and feasible microphone-loudspeaker setups were suggested. In this paper, we present a novel open loop leakage reduction scheme, and it is compared with adaptive noise control method. To achieve this, control available frequency bands and spatial ranges are studied. And the influence of leakage reduction performance by the receiver and control speaker's radiation pattern are analyzed. Also the effects of the physical environment including the user's hand for the leakage control performance are studied. Finally, the proposed method is implemented on the mobile phone mock-up, and the performance of actual measured leakage reduction is investigated.

3:00

2pEAb7. Active control applied to simplified wind musical instrument. Thibaut Meurisse, Adrien Mamou-Mani, René Caussé (Instrumental Acoust., IRCAM, 1 place Stravinsky, Paris, France, thibaut.meurisse@ircam.fr), and David Sharp (Acoust. Res. Group, The Open Univ., Milton Keynes, United Kingdom)

Musicians have always been interested in the evolution of their instruments. This evolution might be done either to adapt an instrument's quality to musicians' and composers' needs, or to enable it to produce new sounds. In this study, we want to control the sound quality and playability of wind instruments, using active control. The active control makes it possible to modify the input impedance (frequency, gain, and damping) of these instruments and to modify the instrument's quality. Simulations and first experiments on a simplified reed instrument are presented. We simulate a control of the modes (frequency, damping) of a cylinder using two different approaches: classic feedback and modal active control. Then, we apply these control methods on a simplified reed instrument with embedded microphone and speaker. Finally, the effects on sound and playability of the instrument is studied.

3:20

2pEAb8. Predicting speech transmissibility using ray tracing and statistical energy analysis. Sascha Merz, Vincent Cotoni, and Phil Shorter (ESI Group, 12555 High Bluff Drv, Ste. 250, San Diego, CA 92130, sascha.merz@esi-group.com)

Statistical Energy Analysis (SEA) is widely used for predicting interior noise across many different industries. SEA typically describes steady-state reverberant responses, for example the response at interior passenger locations in transportation applications due to steady-state exterior sources. However, many applications exist where the direct field is also important. This includes prediction of Speech Transmissibility for audio systems in automotive applications or public address systems in train and aircraft applications. For predicting Speech Transmissiblity, the transient response at a given receiving location due to transient excitation applied at a particular source location must be known. Geometrical methods such as ray tracing are often used for describing the first few reflections of the direct field; however, they are not well suited for describing the reverberant field and typically include approximate statistical assumptions about late time reflections. These assumptions often do not consider the detailed distribution of sound package in interior spaces. An alternative approach is to use ray tracing for predicting the direct field and low order reflections and to use SEA to predict late time reflections and background noise. This approach is computationally efficient and can use information contained in existing SEA models. The method is discussed and validation examples are presented.

3:40

2pEAb9. Comparison of precedence effect behavior in anechoic chamber with that in ordinary room. Koji Abe, Shouichi Takane, Sojun Sato, and Kanji Watanabe (Electron. and Information Systems, Akita Prefectural Univ., 84-4 Ebinokuchi Tsutiya, Yuri-Honjyo 015-0055, Japan, koji@akita-pu.ac.jp)

The precedence effect is well known as one of auditory illusions occurred by using multiple sound sources with similar sound output. When a sound is followed by similar sound separated with relatively short time delay, a single fused sound image is localized at the source position corresponding to the first-arriving sound. This feature is applicable to public address systems, which make audience perceive the sound image different from the actual sound source positions prepared for the system, with some sound reinforcement achieved. In spite of many studies in this phenomenon, the behavior of the precedence effect has been investigated for limited sound source arrangements in laboratory environments like anechoic chamber. On the other hand, this behavior in the ordinary room is not obvious, and it is effective to clarify the difference of the behavior of the precedence effect in anechoic chamber from that in the ordinary room for the application of the precedence effect to the public address system. In this study, the similar sound sources were installed both in the lecture room and in the anechoic chamber, and the behavior of the precedence effect was compared each other with the given time and level difference among sound sources.

4:00

2pEAb10. Vertical sound image control using level differences between parametric speakers. Kumi Maeda, Takanori Nishino, and Hiroshi Naruse (Grad. School of Eng., Mie Univ., 1577 Kurimamachiya-cho, Tsu 5148507, Japan, nishino@pa.info.mie-u.ac.jp)

Horizontal sound image can be controlled by using level difference between two loudspeakers; however, vertical sound localization is difficult. In this report, we propose a method of controlling a sound image with sound level differences between two parametric speakers that have a super-directivity. Our proposed system uses sounds that are reflected on a wall. In our experiments, two parametric speakers were arranged 56.6 and 226.0 cm high, respectively. Vertical sound localization was evaluated by subjective tests. Subjects were seven males and one female. Test signals were a white noise whose duration was 0.5 s. Sound level differences between parametric speakers were $-\infty$, -9, -6, -3, 0, 3, 6, 9, and ∞ dB. Test signals were presented three times to each subject in a random order. Both parametric speakers were arranged at angles of $0°$, $\pm 5°$, and $\pm 10°$ from the horizontal plane, respectively. Answers were examined by the Wilcoxon rank-sum test. From the results, good performances were obtained when parametric speakers were arranged at angles of $\pm 10°$. [Work supported by a Grant-in-Aid for Scientific Research (24500203).]

4:20

2pEAb11. Characteristics of whistle noise from mufflers with perforated pipes. Tatsuya Yamada, Takehiko Seo, Masato Mikami (Grad. School of Sci. and Eng., Yamaguchi Univ.,2-16-1 Tokiwadai, Ube, Yamaguchi, Japan, 201, 3-44 Wakamatsucho, Ube, Yamaguchi, Japan, Ube, Yamaguchi 755-8611, Japan, s045ve@yamaguchi-u.ac.jp), and Takashi Esaki (Sango Co., Miyoshi, Aichi, Japan)

As a countermeasure to reduce exhaust gas pulsation noise in a wide frequency range, the exhaust system employs expansion, resonance and sound absorbing structures with perforated pipes. However, whistle noise is generated near holes under some conditions. In straight-through-type mufflers, sound absorbing materials covering punching holes can suppress whistle noise generation. On the other hand, in the expansion-cavity-type muffler, which has inlet and outlet insertion pipes, effective countermeasures have not been clarified yet. The purpose of this study is to improve understanding of whistle noise generation mechanism in mufflers with perforated pipes. First, we measured sound pressure of whistle noise radiating from a straight-through-type sub-muffler with a perforated pipe with steady flow. Results show that the frequency of predominant whistle noise became higher stepwisely with increasing the flow velocity and was higher with smaller hole diameter. Strouhal number based on the hole diameter, the frequency of predominant whistle noise and flow velocity existed within a certain range while the hole diameter and flow velocity were varied. Next, we measured the sound pressure of whistle noise radiating from an expansion-cavity-type main muffler with a perforated pipe. The whistle noise generation is discussed in comparison with that for the sub-muffler.

4:40

2pEAb12. Back scattering attenuators (silencers). Giora Rosenhouse (89 Hagalil St., Haifa 32684, Israel, fwamtech@bezeqint.net)

Back scattering silencers are a kind of "sonic crystals." The frequency range of "sonic crystals" modeling includes the whole audio range and ultrasound, up to phonon waves. Here we concentrate on applications in the audio domain. However, the present investigation is applicable at the higher frequencies as well¸ because of the high scalability of the system. The paper defines and analyses 2-D and 3-D back scattering silencers, made of arrays of rigid or soft obstacles (cylinders, spheres or prolate/oblate spheroids for example), in order to attenuate plane waves by multiple scattering along a wave guide. This effect is strong especially at certain band-gaps along the frequency domain. Each obstacle reflects secondary waves that are partially reflected. Thus, along each row of the array, the sound waves lose a certain amount of the energy that adds to the total amount of attenuation. Specifically, the paper analyses back scattering silencers built of meshes of either cylinders or prolate spheroids, where each obstacle is located symmetrically within a fluid cell and each cell is identical to the others.

Session 2pED

Education in Acoustics: Teaching Methods in Acoustics

Preston S. Wilson, Chair
Appl. Res. Lab., Univ. of Texas at Austin, 1 University Station, TX 78712-0292

Contributed Papers

1:20

2pED1. A design of an impedance tube for teaching acoustic material properties and laboratory techniques. Chelsea E. Good, Aldo A. Glean, Joseph F. Vignola, John A. Judge, Teresa J. Ryan, Nicole Bull (Mech. Eng., Catholic Univ. of America, 620 Michgan Ave. NE, Washington, DC 20064, 26good@cardinalmail.cua.edu), and Diego Turo (Bioengineering, George Mason Univ., Fairfax, VA)

The design and implementation of an adaptable acoustic impedance tube for instructional use will be presented. This system, which was designed for use in a graduate-level laboratory environment, is in accordance with ASTM standard E1050-98. The system is configurable for both horizontal and vertical measurements. Vertical use enables acoustic property measurement of granular materials, allowing characterization of a wide variety of media. The academic content is delivered by engaging students in measurements using the impedance tube. Specific acoustics content objectives include modeling of sound propagation in porous materials and the role of acoustic parameters such as porosity, tortuosity, and flow resistivity in quantifying acoustic behavior of materials. Instructional goals also include mastery of computer based data acquisition and processing. This aspect of modern laboratory practice is a critical part of our graduate acoustics curriculum. Students participated actively in the development of the software used to collect data and calibrate the impedance tube. The system design required the students to solve practical problems such as cutting and positioning of the sample as well as more demanding tasks such as design of a multilayer porous medium with desired acoustic properties.

1:40

2pED2. Popular papers with short case stories on acoustics and vibration for practical engineers and students. Roman Vinokur (Engineering, ResMed Motor Technol., 9540 De Soto Ave., Chatsworth, CA 91311, romanv@resmed.com)

One of the reasons for using foam wedges or cones in hemi-anechoic rooms is a gradual change of the acoustical impedance in order to reduce the reflection of incident sound waves from the sound absorbing walls. By analogy, popular papers on science (in particular, in acoustics and vibration) facilitate a smooth introduction to new theories because of their small cognitive "impedance" to understanding the written information. Such papers are relevant mostly for extramural reading but they help engineers and students to promptly perceive important effects and applications via interesting case stories and simplified physics and mathematics. Generally speaking, this approach is not new: in particular, it was successfully applied by Perelman in his book "Physics for Entertainment." But in author's opinion, for better effectiveness such texts should be limited in size and include 3–4 related short case stories from actual engineering or consulting practice, history, news, or literature. To illustrate this method, several one-page papers published in the "Sound and Vibration" magazine will be briefly discussed: "Vibroacoustic Measurements without Transducers," "A Common Myth about Mechanical Resonance," "Only the Best Will Do," and "Haunted Buildings and Other Acoustical Challenges."

2:00

2pED3. Mechanical bent-type models of the human vocal tract consisting of blocks. Takayuki Arai (Dept. of Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp)

In our previous work, we developed several physical models of the human vocal tract and reported that they are intuitive and helpful for students studying acoustics and speech science. Models with a bent vocal tract can achieve relatively realistic tongue movements. These bent-type models had either a flexible tongue or a sliding tongue. In the former case, the tongue was made of a flexible gel-type material so that we could form arbitrary tongue shapes. However, this flexibility meant that training is needed to achieve target sounds. In the latter case, the tongue was made of an acrylic resin, and only a limited number of vowel sounds can be achieved because so few sliding parts are available to change the tongue shape. Therefore, in this study, we redesigned the mechanical bent-type models so that they now consist of blocks. By placing the blocks at the proper positions, the block-type model can produce five intelligible Japanese vowels. We also designed a single bent-type model with sliding blocks that can produce several vowel sounds. [This work was partially supported by a Grant-in-Aid for Scientific Research (24501063) from the Japan Society for the Promotion of Science.]

2:20

2pED4. Teaching acoustics at architect students using digital tools. Delphine Bard (Eng. Acoust., Lund Univ., John Ericssonväg 1, Lund 221oo, Sweden, delphine.bard@construction.lth.se), Tina-Henriette Kristiansen (Architecture, Lund Univ., Lund, Sweden), and Eva Frühwald Hansson (Solid Mechanics, Lund Univ., Lund, Sweden)

At the School of Architecture, Lund University Sweden, courses are taught in different ways. A large part of the education during year one and two is held as "studios," doing creative (individual) project work. Usually acoustics courses rather correspond to the traditional engineering education style, using lectures, exercises, small project works, and final written examination. The problem is that many students do not know how to use the gathered information in their creative works. The aim of the study covered by this paper was to improve upon the existing teaching/learning of the fundamental acoustics principles scheme by introducing new methodologies. In order to achieve our goal, we gave the students two different assignments: In the first assignment, the students had to produce short educational movies to explain and teach acoustics principles to their peers. In the second assignment, they should implement the new knowledge gathered in the first assignment into their individual creative projects.

2:40

2pED5. Black box measurements – Using a family of electrical circuits as a tool for self-guided learning in acoustical engineering. Bernardo H. Murta, Sergio Aguirre, Jessica Lins, Stephan Paul, Eric Brandao (Departamento de Estruturas e Construção Civil, Universidade Federal de Santa Maria, Rua Dezenove de Novembro, 289, 302, Santa Maria, Rio Grande do Sul 97060160, Brazil, be.murta@gmail.com), and Pascal Dietrich (Inst. of Tech. Acoust., RWTH Aachen Univ., Santa Maria, Brazil)

A partnership between Brazil's first undergraduate program in Acoustical Engineering and the Institute of Technical Acoustics of RWTH Aachen University yielded in a didactic project that uses the engineering software MATLAB with the ITA-Toolbox to teach acoustic measurements. Simple electrical circuits are used to mimic typical behavior of acoustical systems. This low-cost solution has proven to be didactically very effective since it helps students to identify themselves with the measurement tasks. Two hardware solutions were developed-a simple oscillator circuit integrated into connectors of audio cables and a desktop box containing seven different transfer characteristics ranging from ideal linear and time-invariant to nonlinear and time-varying behavior. Undergraduate students of Acoustical Engineering used both devices in classroom experiments for self-guided learning by comparing their results to published results. Students were able to learn the fundamental concepts of acoustical measurements and to handle measurement tasks. Besides the practical experiences and the learning effect, the students were also encouraged to step into the open source routines of the software, understand the signal processing steps, adapt routines, and even write their own ones, e.g., a GUI that provides effective control of the measurement via touch-screens.

3:00

2pED6. A for play! Daniel Butko (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu)

The task was straightforward; design and build a playhouse to be raffled for a charitable organization. A team consisting of students, volunteers, and faculty banded together to not only meet the requirements but to exceed the typical preconceived ideas of a totally enclosed miniature home. Based upon the needs of juvenile clients, the design team focused more on "play" than on "house" when working out conceptual ideas. The playful design was based upon the enclosure being partially open to allow air flow, sunlight, and the ability for the inhabitants to have an aural connection to the outside. The idea of having partially open space on the lower level, a mere 5'-0" x 5'-0" footprint, flanked by stepped bands of cedar and cypress yielded a particular acoustical presence. The space is not only visually unique, but the selection of materials, how they were cut and assembled, and the scale in relation to a seated child enhance the fun factor by creating an enveloping and somewhat amplified acoustic. This project provided pedagogical opportunities within an atypical learning environment. The final inhabitable playhouse exceeded our visual and acoustical expectations of a small space and prove acoustics "plays" an intrinsic role despite occupant age.

TUESDAY AFTERNOON, 4 JUNE 2013 512DH, 1:00 P.M. TO 5:40 P.M.

Session 2pMU

Musical Acoustics: Musical Preference, Perception, and Processing

Jean-François Petiot, Cochair
IRCCyN, Ecole Centrale de Nantes, 1 rue de la noe, BP92101, NANTES 44321, France

Richard L. King, Cochair
Music Res., McGIll Univ., 555 Sherbrooke West, Montreal, QC H3A 1E3, Canada

Contributed Papers

1:00

2pMU1. Modeling of the subjective quality of saxophone reeds. Jean-François Petiot, Pierric Kersaudy (IRCCyN, Ecole Centrale de Nantes, 1 rue de la noe, BP92101, NANTES 44321, France, jean-francois.petiot@irccyn.ec-nantes.fr), Gary Scavone, Stephen McAdams (CIRMMT, Schulich School of Music, McGill Univ., Montreal, QC, Canada), and Bruno Gazengel (LAUM, Université du Maine, Le Mans, France)

The subjective quality of cane reeds used on saxophones or clarinets may be very different from one reed to another even though the reeds have the same shape and strength. The aim of this work is to understand the differences in the subjective quality of reeds and to explain them with objective measurements. A subjective study, involving a panel of 10 musicians, was first conducted on a set of 20 reeds of the same strength. Second, signal recordings during saxophone playing (*in vivo* measurements) were made of the pressures in the player's mouth, in the mouthpiece and at the bell of the instrument. These measurements enable us to deduce specific parameters, such as the threshold pressure or the spectral centroid of the notes. After an analysis of the subjective and objective data (assessment of the agreement between the assessors and the main consensual differences between the reeds), correlations between the subjective and objective data were performed. To propose a model of the subjective quality, a machine learning approach was proposed using partial least-squares (PLS) regression and PLS discriminant analysis. Results show interesting performance of the model in cross validation and open the potential for an objectification of the perceived quality.

1:20

2pMU2. Perceptual evaluation of violins: A comparison of intra-individual agreement in playing vs. listening tasks for the case of richness. Charalampos Saitis, Gary P. Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., 555 Sherbrooke Str. West, Montreal, QC H3A 1E3, Canada, charalampos.saitis@mail.mcgill.ca), Claudia Fritz (Lutheries-Acoustique-Musique, Institut Jean Le Rond d'Alembert, Université Pierre et Marie Curie, Paris, France), and Bruno L. Giordano (Inst. of Neurosci. and Psychol., Univ. of Glasgow, Glasgow, United Kingdom)

In previous studies by the authors, it was shown that there is a significant lack of agreement between violinists when evaluating different instruments in terms of perceived richness in free-playing tasks. A new experiment was designed to further investigate the perceptual evaluation of richness using both a constrained-playing task, which was recorded, and a subsequent listening task (using the previously recorded sounds). The goal was to compare the evaluation of richness from playing vs. listening tasks in order to better understand whether they are based more on auditory feedback or tactile and

proprioceptive cues in the wider context of correlating audio features extracted from the recordings with richness judgments. Skilled violinists were asked to rank five different instruments by playing only certain notes on the G-string. Subsequently, the players were asked to listen to their recordings and rank the violins. Results appeared to show a higher inter-individual agreement relative to previous studies. Furthermore, the rankings in the playing task were generally different from those in the listening task, indicating that the evaluation of richness is based on different criteria in the two cases. Results from matching the tristimulus and spectral centroid of recorded tones to richness judgments will be presented at the conference.

1:40

2pMU3. Music of the body: An investigation of skull resonance and its influence on musical preferences. Jitwipar Suwangbutra, Rachelle Tobias, and Michael S. Gordon (Dept. of Psych., William Paterson Univ., 300 Pompton Rd., William Paterson U., Wayne, NJ 07470, jitwipar@gmail.com)

Musical preferences can be attributed to environmental and biological factors. This research analyzes the specific influence of body resonance, and in particular, how the resonant properties of the skull might contribute to auditory perception of music and musical preferences. To examine this issue resonances were sampled from a set of participants and analyzed using FFTs. The fundamental frequencies of each participant's head was correlated against their preference amongst a set of novel melodies presented in each of the 12 major keys. Using this method the spectral properties of the melody could be directly related to the resonant properties of a listener's skull to evaluate their influence. While results were subtle, participants were found to be influenced in their judgments of loudness and musical preference for the melodies. Conclusions from this research support speculation on an embodied model of cognition for musical interactions.

2:00

2pMU4. Preferences for melodic contours transcend pitch. Jackson Graves, Christophe Micheyl, and Andrew J. Oxenham (Psychology, Univ. of Minnesota, 1849 Washington Ave. S, Apt 435, Minneapolis, MN 55454, grave276@umn.edu)

The question of what makes a good melody has interested composers, music theorists, and psychologists alike. Many of the observed principles of good "melodic continuation" involve the melodic contour—the pattern of rising and falling pitch within a sequence. Recent work has shown that contour perception can extend beyond pitch to other auditory dimensions, such as timbre and loudness. Here we show that the generalization of contour perception to non-traditional dimensions also extends to melodic preferences and expectations. We find that subjective continuation ratings for sequences that vary in brightness or loudness generally conform to the same contour-based expectations as pitch sequences. The results support the hypothesis that contour perception is a general auditory phenomenon, and suggest that the well-known preference for narrow ranges and small intervals in melodies is not unique to the dimension of pitch. [Work supported by NIH Grant R01DC05126 and by the Undergraduate Research Opportunities Program of the University of Minnesota.]

2:20

2pMU5. Development of a new series of tests to assess the effectiveness of hearing aids for the perception of music. Martin Kirchberger (Health Sci. & Technol., ETH Zürich, Universitätsstrasse 2, Zürich 8092, Switzerland, martin.kirchberger@phonak.com), Frank A. Russo (Psychology, Ryerson Univ., Toronto, ON, Canada), Peter Derleth, and Markus Hofbauer (Sci. & Technol., Phonak AG, Stäfa, Switzerland)

A new series of tests has been designed to assess the effectiveness of hearing aids for the perception of music. Within each subtest, discrimination thresholds for low-level acoustic dimensions are determined adaptively using a 2AFC method within the context of a musical judgment regarding melody, harmony, timbre or meter. The presented test stimuli are synthesized and either unprocessed or processed by different hearing aid signal processing algorithms before being played back via loudspeaker. The battery will be used to evaluate different hearing aid algorithms with regard to their benefit for functional hearing in music. A group of six normal hearing control participants (6.7 dB HL) and five hearing impaired participants (34 dB HL) each performed the melody subtest and the harmony subtest twice. The hearing impaired subjects had higher discrimination thresholds than the control group. A comparison of the results from both administrations suggests that these two subtests have good test-retest reliability.

2:40

2pMU6. The examination of the performance motion and emotional valence by a pianist. Yuki Mito, Hiroshi Kawakami (College of Art, Nihon Univ., 2-42-1 Asahigaoka Nerima-Ku, Tokyo 176-8525, Japan, mitotic@hotmail.com), Masanobu Miura (Faculty of Sci. and Technol., Ryukoku Univ., Shiga, Japan), and Yukitaka Shinoda (College of Sci. and Technol., Nihon Univ., Tokyo, Japan)

Until now, we examined the performance motion by the snare drum. Particularly, we analyzed an association between emotion and motion using motion capture. We analyzed total of six emotions [tenderness, happiness, sadness, fear, anger, and non-emotion from Juslin (2001)]. The analysis method averaged the performance motion of six emotions. We named the method as "Motion Averaging Method (MAM)." We calculated the difference of the each emotion by the MAM and revealed the characteristic of motion. From the result, we achieved results in snare drum motion. Therefore, in this study, we considered the association between emotion and motion in the keyboard. The subject was a professional pianist. A musical piece was an etude of the music dictation. Motion capture system is the MAC 3D System of Motion Analysis Corp., which is an optical motion capture system. The measurement marker bonded 33 points to the upper body. As a result, we understood the difference in emotional valence by the center of gravity of the head, an arm, the body, and the hand. Particularly, the non-emotion was move less than other five emotions. Then, we were able to express an association between five emotions of Juslin and motion as a figure.

3:00

2pMU7. Human ability of counting the number of instruments in polyphonic music. Fabian-Robert Stöter, Michael Schoeffler, Bernd Edler, and Jürgen Herre (Int. Audio Lab. Erlangen, Am Wolfsmantel 33, Erlangen 91058, Germany, fabian-robert.stoeter@audiolabs-erlangen.de)

There are indications that humans are only able to correctly count up to three voices in polyphonic music pieces of homogeneous timbre, where each voice is played by the same instrument. A more general case, where voices are played by instruments of inhomogeneous timbre, has not been fully addressed so far. In order to approach this question we conducted a listening experiment with 62 participants to find out whether both scenarios — instrumentation by inhomogeneous or homogeneous timbre — share the same outcome. This paper describes the design of the experiment including an analysis of the results, which show that both scenarios are related. Furthermore, a detailed analysis of the error rates in correctly counting the number of instruments reveals that there are significant differences between non-musician and musician listeners, in particular regarding the upper auditory limit of the number of correctly counted instruments. Based on these results, models for the perception of instruments in auditory streams can be developed.

3:20–3:40 Break

3:40

2pMU8. Loudspeakers and headphones: The effects of playback systems on listening test subjects. Richard L. King, Brett Leonard (Music Res., Schulich School of Music, McGill Univ., 555 Sherbrooke West, Montreal, QC H3A 1E3, Canada, richard.king@mcgill.ca), and Grzegorz Sikora (Automotive Audio, Bang & Olufsen, Pullach, Germany)

Many modern listening test designers disagree on the best playback system to be used when conducting tests. The preference often tends toward headphone-based monitoring in order to minimize the possibility of undesirable acoustical interactions with less than ideal testing environments. On the other hand, most recording and mixing engineers prefer to monitor on loudspeakers, citing a greater ability to make critical decisions on level balances and effects. While anecdotal evidence suggests that differences exist between systems, there is little quantified, perceptually based data to guide both listening test designers and engineers in what differences to expect

when alternating between monitoring systems. Controlled tests are conducted with highly trained subjects manipulating the level of solo musical elements against a backing track, using both headphones and loudspeakers. This task serves to make the results equally applicable to critical mixing tasks and rigorous listening tests. The results from both playback systems are compared, showing a defined difference in the mean levels set on the two different monitoring systems. Likewise, the variance seen across subjects is larger when monitoring on headphones than on loudspeakers, lending credence to the hypothesis that tests conducted on one playback system may not be equally applicable to the other.

4:00

2pMU9. Modeling listener distraction resulting from audio-on-audio interference. Jon Francombe, Russell Mason, Martin Dewhirst (Dept. of Music and Sound Recording, Inst. of Sound Recording, Univ. of Surrey, Guildford, Surrey GU2 7XH, United Kingdom, j.francombe@surrey.ac.uk), and Søren Bech (Bang & Olufsen, Struer, Denmark)

As devices that produce audio become more commonplace and increasingly portable, situations in which two competing audio programs are present occur more regularly. In order to support the design of systems intended to mitigate the effects of interfering audio (including sound field control, noise cancelation or source separation systems), it is desirable to model the perceived distraction in such situations. Distraction ratings were collected for a range of audio-on-audio interference situations including various target and interferer programs at three interferer levels, with and without road noise. Time-frequency target-to-interferer ratio (TIR) maps of the stimuli were created using a simple auditory model. A number of feature sets were extracted from the TIR maps, including combinations of mean, standard deviation, minimum and maximum TIR taken across the duration of the program item. In order to predict distraction ratings from the features, linear regression models were produced. The models were evaluated for goodness-of-fit (RMSE) and generalizability (using a K-fold cross-validation procedure). The best model performed well, with almost all predictions falling within the 95% confidence intervals of the perceptual data. A validation data set was used to test the model, suggesting areas for future improvement.

4:20

2pMU10. Effects of audio latency in a disc jockey interface. Laurent S. Simon, Arthur Vimond (INRIA, INRIA Rennes, Bretagne Atlantique, Campus Universitaire de Beaulieu, Rennes Cedex 35042, France, laurent.s.simon@inria.fr), and Emmanuel Vincent (INRIA Nancy, Rennes, France)

This study presents an evaluation of the disturbance caused by audio latency in a DJ-ing task. An experiment was conducted, during which subjects were asked to synchronize one piece of music to a reference piece of music using a common DJ-ing interface. Synchronization was performed by adjusting the speed of one of the music pieces to that of a reference piece of music and time-aligning both pieces. Latency was introduced between the interface and the audio output, varying between 0 and 550 ms. The average synchronization time was estimated as a function of subjects, beat-per-minute difference between the pieces of music, and latency. Results showed that for trained DJs, synchronization time increased significantly above 130 ms of audio latency, whereas for naive subjects, latency had no influence on the synchronization time.

4:40

2pMU11. Evaluating the absolute volume of digital sound source measurement and standard measuring unit. Doo-Heon Kyon and Myung-Jin Bae (Electron. Eng. Dept., Soongsil Univ., Hyeongnam Eng. Hall #1212, Sangdo-dong, Seoul 156-030, Democratic People's Republic of Korea, kdhforce@gmail.com)

Listeners do not know the actual volume of sound before playing a sound source, so they have to adjust the volume through trials and errors. Moreover, they have to change the volume repetitively because each sound source has different volume. If we can identify the absolute volume of a sound source at the perspective of listener, the volume of all sound sources can be effectively standardized. This study evaluated a method to measure the absolute volume of a digital sound source and suggested the dB(N) as a measuring unit. The pink noise was used as a reference sound source, to be used for measuring the absolute volume. The pink noise was set as 60 dB(N), which is equal to sound output of 60 dB(A). The volume was adjusted until the pink noise and the target sound matches into a recognizable volume by reducing the pink noise or target sound source under the given environment. Subsequently, the difference is reflected to 60 dB(N) to determine the absolute volume. The accuracy of measured results was confirmed through a music listening test and suggested how to develop the volume system using the absolute volume.

5:00

2pMU12. Refining the stereo technique for augmented ambience gradient: Improvements in stereo image, spatial envelopment, and mixing flexibility. David J. Tagg and Kevin Fallis (Sound Recording, McGill Univ., 74 Cameron Cres, Toronto, Ontario M4G 2A3, Canada, kevin.fallis@mail.mcgill.ca)

While working on location, recording engineers are often challenged by insufficient monitoring. Poor (temporary control room) acoustics and/or mandatory headphone monitoring can make judgments regarding microphone choice and placement difficult. This compromised monitoring often leads to timbral, phase, and stereo image problems. We are often forced to choose between the improved spatial imaging of near-coincident techniques and the attractive acoustic envelopment from spaced omni-directional mics. This research reviews a new technique: Stereo Technique for Augmented Ambience Gradient (STAAG), which aims to improve stereo imaging, ambient envelopment, and flexibility in the mix. Building on a preliminary study, this research realizes ideal microphone angle/spacing combinations to promote spatial accuracy, investigates the quality of the ambient envelopment compared to omnidirectional-based techniques, and the ability of STAAG to allow an engineer to manipulate the direct to reverberant energy ratio during post-production without corrupting the stability of the stereo image.

5:20

2pMU13. Real-time concatenative synthesis for networked musical interactions. Chrisoula Alexandraki (Dept. of Music Technol. and Acoust., Technol. Educational Inst. of Crete, E. Daskalaki Perivolia, Rethymnon, Crete 74100, Greece, chrisoula@staff.teicrete.gr) and Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Hamburg, Germany)

The recent proliferation of Networked Music Performances has led to the investigation of low-latency, low-bitrate musical encoding schemes, including audio codecs and control protocols that specifically address the requirements of live musical interactions across the Internet. This work presents an alternative perspective inspired by the "synthesis by analysis" approach, tightly constrained in terms of processing latencies and rendering quality. The entire process is fully automated and involves an offline processing phase (that takes place prior to performance) and an online real-time analysis-synthesis phase. The offline phase involves processing a solo recording of each musician's part so as to acquire (a) audio segments corresponding to each note in the performance, and (b) a trained Hidden Markov Model to be later used for online analysis. During live performance, the online analysis process encodes the position of the performance on a music score and re-synthesizes the audio waveform by concatenating the audio segments of the offline phase. Although the synthesized waveform originates from an offline solo recording, it is synchronized to the live performance at note level, so as to allow for rendering a wide range of musical tempi as well as their expressive variations. The paper presents the complete methodology and reports on implementation details and preliminary evaluation results.

Session 2pNSa

Noise: Community Response to Low-Amplitude Sonic Booms

Alexandra Loubeau, Cochair
NASA Langley Res. Ctr., MS 463, Hampton, VA 23681

Juliet A. Page, Cochair
Wyle, 200 12th St. South Ste. 900, Arlington, VA 22202

Chair's Introduction—12:55

Invited Papers

1:00

2pNSa1. Overview of the waveform and sonicboom perception and response program. Juliet A. Page (Wyle, 200 12th St. South Ste. 900, Arlington, VA 22202, juliet.page@wyle.com)

The Waveform and Sonicboom Perception and Response (WSPR) Program conducted in California in November 2011 was designed to test and demonstrate the applicability and effectiveness of techniques to gather data relating human subjective response to multiple low-amplitude sonic booms. It was a practice session for future wider scale testing of communities, eventually using a purpose built low-boom demonstrator aircraft. The WSPR program addressed the following: design and development of an experimental design to expose people to low-amplitude sonic booms; development and implementation of methods for collecting acoustical measures of the sonic booms in the neighborhoods where people live; design and administration of social surveys to measure people's reactions to sonic booms; and assessment of the effectiveness of various elements of the experimental design and execution to inform future, wider-scale testing. The low boom community response pilot experiment acquired sufficient data to assess and evaluate the effectiveness of the various physical and psychological data gathering techniques and analysis methods. Results include a comparison of survey modes, techniques for correlating subjective and objective data, assessment of single event and cumulative daily sonic boom subjective and percent highly annoyed results, and methods for analysis of empirical boom data.

1:20

2pNSa2. Low amplitude sonic boom noise exposure and social survey design. Kathleen K. Hodgdon (Appl. Res. Lab., Penn State, ARL North Atherton St., P.O. Box 30, State College, PA 16804-0030, kkh2@psu.edu) and Juliet Page (Wyle, Arlington, VA)

The Waveform and Sonicboom Perception and Response (WSPR) Program conducted a field study relating subjective response to noise from multiple low-amplitude sonic booms. The team was led by Wyle and included researchers from Penn State, Tetra Tech, and Gulfstream Aerospace Corp. The test exposed residents in the Edwards Air Force Base (EAFB) Housing area to two weeks of low-amplitude sonic booms while recording their responses via surveys. The noise exposure design balanced DNL across test days, the number of low, medium, and high booms, and the separation of booms between AM and PM flight sequences. Survey instruments consisted of a Baseline survey, a Single Event survey, and a Daily Summary survey. The WSPR low boom survey included a question on strength of annoyance, followed by questions on the strength of perception of five additional variables that contribute to the annoyance response. Three modes of administration were utilized for both the single event and daily summary surveys: paper/pen, web-based, and Mobile (Apple) device. The survey followed recommendations published by The International Commission on the Biological Effects of Noise (ICBEN). The data from the low boom field test provide a measure of the acceptance of low booms in an acclimated community.

1:40

2pNSa3. A flight research overview of the Waveforms and Sonicboom Perception and Response Project, the National Aeronautics and Space Administration's pilot program for sonic boom community response research. Larry J. Cliatt, Edward A. Haering, Michael D. Holtz, Thomas P. Jones, Erin R. Waggoner (NASA Dryden Flight Res. Ctr., P.O. Box 273, M.S.2228, Edwards, CA 93523, larry.j.cliatt@nasa.gov), Scott L. Wiley (Tybrin Corp., Edwards, CA), Ashley K. Parham (NASA Dryden Flight Res. Ctr., Edwards, CA), and Franzeska F. Houtas (Tybrin Corp., Edwards, CA)

To support the National Aeronautics and Space Administration's (NASA) ongoing effort to bring supersonic commercial travel to the aerospace industry NASA Dryden, in cooperation with other government and industry organizations, conducted a flight research experiment to identify the methods, tools, and best practices for a large-scale sonic boom community human response test. The name of the project was Waveforms and Sonicboom Perception and Response (WSPR). Such tests go toward building a dataset that governing agencies like the Federal Aviation Administration and International Civil Aviation Organization will use to establish regulations for acceptable sound levels of overland sonic booms. This paper focuses on NASA's role in the project on essential elements of community response testing including recruitment, survey methods, instrumentation systems, flight planning, and operations. Objectives of the testing included exposing a residential community with sonic boom doses designed to simulate those produced by the next generation of commercial supersonic aircraft. The sonic booms were recorded with an instrumentation array that spanned the community. Human

response data was collected using multiple survey methods, and was correlated to acoustic metrics from the sonic booms. The project resulted in lessons-learned and the findings of appropriate methods necessary to implement a successful large-scale test.

2:00

2pNSa4. Objective data collection and analysis for the waveform and sonic boom perception and response program. Brian Cook, Joe Salamone (Acoust. and Vib., Gulfstream Aerosp., 3 Innovation Dr., M/S R-4P, Savannah, GA 31408, brian.cook@gulfstream.com), Chris Hobbs, and Juliet Page (Wyle, Arlington, VA)

The Waveform and Sonic boom Perception and Response (WSPR) program experiment was conducted in November 2011. Low-amplitude sonic booms were created by planned NASA F-18 supersonic flights executing a unique dive maneuver. The WSPR program was designed to simultaneously collect objective sonic boom acoustic data and subjective response data from residents in the Edwards Air Force Base residential community. Sonic Boom field kits were developed for the WSPR program consisting of a digital data acquisition system with networked nodes, deployable for extended periods of time. The Sonic Boom Unattended Data Acquisition System (SBUDAS) purposely developed for sonic boom community noise testing was deployed and details of the measurement system and all aspects of the objective data collection process are described. Data analysis during testing provided vital information to the flight planners for experimental execution. This paper also explains the post-experimental analysis of the objective data achieved by creation of a measurement data archive, predictions of sonic boom exposure at subject household locations, an automated algorithm to locate sonic booms within the recorded data and computation of a variety of indoor and outdoor metrics.

2:20

2pNSa5. A comparison of survey implementation methods. Peg Krecker, Carrie Koenig (TetraTech, Madison, WI), Clifton Wilmer, and Juliet A. Page (Wyle, 200 12th St. South Ste. 900, Arlington, VA 22202, juliet.page@wyle.com)

As part of a pilot program to measure subjective response to low level sonic booms, 52 residents at Edwards Air Force Base were recruited to answer questions about their reactions to low-amplitude sonic booms. Over a two-week period, participants completed brief surveys (12 items) each time they heard a sonic boom and a short summary form at the end of each day. The study used three modes of survey administration—paper, Web, and Apple mobile device—to support analysis of the effectiveness of different approaches. Previous research on subjective response to sonic booms or other impulsive noise with similar measurement objectives has used in-person surveys, telephone surveys, or computer-assisted self-administered methods similar to a Web survey, but study designs prevented direct comparisons of the methods on the same sample of participants. We examine data quality across the three modes and the paper will present results on completion rates by survey mode, variation in completion rates over time, and differences in the timeliness of response for web and Apple participants. Qualitative interviews with a subset of participants yield further insights into each approach.

2:40

2pNSa6. Statistical analysis of community response to low amplitude sonic boom noise. Kathleen K. Hodgdon (Appl. Res. Lab., Penn State Univ., ARL North Atherton St., P.O. Box 30, State College, PA 16804-0030, kkh2@psu.edu), Juliet Page (Wyle, Arlington, VA), Trent Gaugler, Daisy Phillips, Durland Shumway, and James Rosenberger (Dept. of Stat., Penn State Univ., University Park, PA)

The Waveform and Sonicboom Perception and Response (WSPR) Program conducted a field study of subjective response to noise from multiple low-amplitude sonic booms. The team was led by Wyle and included researchers from Penn State, Tetra Tech and Gulfstream Aerospace Corp. The test exposed residents in the Edwards Air Force Base (EAFB) Housing area to two weeks of low-amplitude sonic booms while recording their responses via surveys. There were 52 participants divided across three response modes. The response instruments included Baseline Surveys, Single Event Surveys submitted each time a participant heard a boom, and Daily Surveys submitted at the end of each day. The analysis included assessments of single events and cumulative daily ratings of annoyance and categorical variables including loudness, interference, startle, vibration, and rattle. The WSPR daily annoyance data was analyzed by computing percent highly annoyed (%HA) and relating it to the cumulative noise exposure and by relating the subjective annoyance rating directly to the daily noise exposure. The WSPR design was established to cover the full range of noise exposures and annoyance factors so that sufficient data would be gathered to facilitate analyses of %HA and noise metrics. The statistical analyses examining these relationships will be presented.

3:00

2pNSa7. Relationships among near-real time and end-of-day judgments of the annoyance of sonic booms. Sanford Fidell (Fidell Assoc., Inc., 23139 Erwin St., Woodland Hills, CA 91367, sf@fidellassociates.com), Richard Horonjeff (Consultant in Acoust. and Noise Control, Boxborough, MA), and Linda Fidell (California State Univ., Emerita, Morro Bay, CA)

A recent social survey of the annoyance of low amplitude sonic booms included both prompt and delayed-response questions about the annoyance of sonic booms heard by respondents in the home over the course of two weeks. Interviews were conducted via smartphone with 49 voluntary test participants. Most of the prompt annoyance judgments were made within about a minute of notice of a sonic boom. The delayed response judgments were solicited in the evening, at a time of the respondent's choosing. Prior analyses showed that dosage-response relationships between the prevalence of high annoyance and sonic boom amplitude were well predicted by CTL analysis. The current analyses investigated how individual, within-day, prompt annoyance judgments were related to end-of-day judgments of the annoyance of sonic booms. Preliminary analyses suggest that end-of-day annoyance judgments are not simply a linear sum or averaging of the annoyances of individual sonic booms.

3:20

2pNSa8. Community response to low-amplitude sonic booms. Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov)

Sonic boom research conducted at NASA is oriented toward understanding the potential impact of sonic boom noise on communities from new low-boom supersonic aircraft designs. This research contributes to knowledge in key areas needed to support development of a new noise-based standard for supersonic aircraft certification. Partnerships with several industry, government, and academic

institutions have enabled the execution of a pilot low boom community response test to develop and assess experimental methodologies, including sonic boom data acquisition, subjective data collection, and data analysis. Areas of additional research are identified and a prioritization of issues is performed to guide design of a potential follow-on pilot test. Lessons learned from these community response tests will facilitate future community testing with actual low-boom aircraft in communities not familiar with sonic booms.

Contributed Papers

3:40

2pNSa9. The impact of including diffraction when predicting the effect of listener environment on the perceived loudness of outdoor sonic booms. Amanda B. Lind and Victor Sparrow (Grad. Prog. in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, State College, PA 16801, Amanda.Blair.Lind@gmail.com)

The human impact of sonic booms varies with listening environment. Given the incident sonic boom waveform, the specular field around a realistic geometry has been predicted via a c++ implementation of image source method (ISM) tailored to outdoor applications. This work explores the necessity of including the diffracted field when predicting time series and PLdB, both in and out of shadow zones. The impulsive nature of the excitation, and the sensitivity of the PLdB to temporal details, constrains appropriate diffraction modeling techniques to those capable of time domain accuracy. Uniform Theory of Diffraction (UTD) and Biot Tolstoy Medwin (BTM) approaches are considered. The benefits and challenges of each approach are explored, particularly with regards to scalability and bandwidth. The importance of accurately predicting diffraction in this application is evaluated through comparison with booms recorded around a building corresponding to the simulated geometry. [Work supported by the FAA/NASA/Transport Canada PARTNER Center Excellence and the Applied Research Laboratory. Experimental data courtesy of NASA.]

4:00

2pNSa10. Effect of room characteristics on perception of low-amplitude sonic booms heard indoors. Clothilde Giacomoni and Patricia Davies (Purdue Univ., 2911 Horizon Dr., Apt. 4, West Lafayette, IN 47906, cgiacomo@purdue.edu)

Supersonic flight over inhabited territories of the United States has been banned by the Federal Aviation Association. While research has been conducted to determine the effects of sonic booms on the general population when heard outdoors, little work has been done on people's perception of sonic booms as heard indoors. A sound's waveform will change in its transmission from outdoors to indoors due to several factors, one of which is the indoor acoustic environment. This can be changed using different room sizes, shapes or which materials are covering each surface (wall, ceiling, or floor). A subjective test, designed to determine which of these room characteristics has an effect on people's ratings of annoyance, has been completed. It was found that smaller rooms and square rooms are rated as more annoying than larger rooms or rooms with a corridor-like or rectangular shape, and that rooms with lower reverberation times were rated as less annoying than rooms with higher reverberation times.

4:20–5:20 Panel Discussion

TUESDAY AFTERNOON, 4 JUNE 2013 — 511CF, 12:55 P.M. TO 5:00 P.M.

Session 2pNSb

Noise and Architectural Acoustics: Soundscape and its Application

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

Bennett M. Brooks, Cochair
Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066

Chair's Introduction—12:55

Invited Papers

1:00

2pNSb1. Soundscape workshop report: Perception Lexicon. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com) and Brigitte Schulte-Fortkamp (Tech. Univ. Berlin, Berlin, Germany)

A workshop was held at the 164th Meeting of the ASA in Kansas City on methodology standardization for the advancement of the developing field of soundscape measurement, analysis, and design. The workshop focused on the terminology lexicon used for soundscape subject interviews. Interviews of local experts, residents, and other users and inhabitants of the sonic environment can yield insights into both personal reactions and universal observations. The specific terminology used in this process may significantly affect the outcomes. As the success of a new research or development project can depend on the lessons learned from previous projects, the standardization of interview techniques becomes increasingly important. Workshop participants were invited to develop a standardized lexicon of descriptors for field use in interview questionnaires. After a review of soundscape objectives and procedures, the participants reviewed key issues and assessed available lexicon term types. The group then proposed, developed, and prioritized terms which could describe a soundscape as it is contextually perceived by an interview subject. Two key conditions were considered for each proposed lexicon term: would the term be commonly understood, and would the term describe a reaction to the soundscape of sufficiently notable intensity. The results of the workshop are presented.

1:20

2pNSb2. Characterizing the soundscape of tranquil urban spaces. Bert De Coensel, Michiel Boes, Damiano Oldoni, and Dick Botteldooren (Dept. of Information Technol., Ghent Univ., St.-Pietersnieuwstraat 41, Ghent B-9000, Belgium, bert.decoensel@intec.ugent.be)

Tranquil spaces provide restorative environments for urban residents and visitors and are therefore essential for health and quality of life. Tranquil spaces may be characterized through a combination of acoustical criteria, such as relatively low (percentile) sound levels and the relative absence of non-fitting sounds, and non-acoustical criteria, such as the presence of natural elements within the visual scene. Public urban parks and courtyards as well as private urban backyards are typically considered to be the most tranquil spots within a city. Current state-of-the-art in distributed measurement technology allows for long-term sound monitoring at these places. In this paper, the soundscape at a number of urban parks and backyards in the cities of Ghent and Antwerp is investigated through a detailed analysis of sound measurements performed over an extended period of time. An analysis of percentile sound levels, noise events and indicators for temporal and spectral structure is presented, and novel computational methods are applied to estimate the relative occurrence of sounds arising from various sources.

1:40

2pNSb3. What will be the influence of e-mobility on soundscape? Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath-Kohlscheid 52134, Germany, klaus.genuit@head-acoustics.de)

The increasing electrification of the powertrain after 125 years of continuous development of the internal combustion engine will not only lead to a sound pressure level reduction of vehicle exterior noises but to a complete change of sound quality. With this expected development road traffic noise affected persons hope for quiet cities and a better quality of life. The creation and successful preservation of quiet zones in cities and to avoid harmful effects of noise exposure are special focuses in European noise policy. However, different surveys have shown the increased risk of accidents for pedestrians and cyclists with respect to collisions with quiet vehicles, which caused a lively discussion about acoustical warning systems for the prevention of crashes. But it is obvious, that major conflicts between quietness and safety arise. Consequently, to address this issue, on the one hand, sustainable concepts must be developed for the successful avoidance of accidents and on the other hand the general traffic noise must be minimized. The sound of electric vehicles will influence in a significant way our soundscape at places and cities in the future. This is a special challenge for psychoacoustics to provide helpful contribution besides the A-weighted sound pressure level.

2:00

2pNSb4. Plan for research to better understand visitor response to the soundscape in national parks. James H. Boyle (Univ. HS, Univ. of Illinois, 3805 Deerfield Dr., Champaign, IL 61822, jhhboyle@gmail.com) and Paul D. Schomer (Schomer and Assoc., Inc., Champaign, IL)

Since 91% of national park visitors come to a park, at least in part, for the natural sounds, analysis of the perceptions visitors have of sound in national parks is extremely significant. In the summer of 2011, several pilot surveys were tested at Rocky Mountain National Park in Colorado for their abilities to provide an accurate reflection of visitors' perceptions of the soundscape. The 2011 surveys were modified in various ways to create the current 2013 survey, which uses "pleasantness" of park sound as the main visitor response metric. The testing plan for the 2013 survey is non-labor intensive for respondents and researchers; rather than closely monitoring activities of respondents, and having them closely monitor the sounds they hear, this survey requires respondents to complete a short questionnaire every 30 to 60 min and an end-of-hike survey. Much like how day-night sound level (DNL) has traditionally been predicted for a cluster of houses around airports or highways, global sound measurements conducted along the trail will be correlated with visitor survey results in order to develop the means to predict how an average park visitor will respond to the soundscape.

2:20

2pNSb5. When do we judge sounds? Relevant everyday situations for the estimation of ecological validity of indoor soundscape experiments. Jochen Steffens (FH Duesseldorf, Josef-Gockeln-Strasse 9, Duesseldorf 40476, Germany, jochen.steffens@fh-duesseldorf.de)

This paper introduces a model of indoor soundscape evaluation, which is based on the results of listening experiments on household appliances. The model reveals, among other things, permanently changing action and attention processes and learning effects which occur in everyday settings. However, what are the relevant situations which we want to reconstruct in our experiments? When do people perceive and evaluate sounds in their everyday life like they do under test conditions? This theoretical knowledge is essential for the estimation of ecological validity of soundscape experiments in general. Hence, this contribution deals with approaches to determine meaningful real-life situations. The interviews held in the course of the studies show that many subjects construct extremely critical context situations while they are assessing sounds. These contexts are often not representative for these peoples' reality. Many participants, in fact, state that they no longer consciously perceive and evaluate the sounds of the appliances in their personal life. Decreasing attention over time can lead to a great influence of the first impression on subsequent perceptions and cognitions (primacy effect). Furthermore, the effect of memory processes on retrospective sound evaluations will be discussed.

2:40

2pNSb6. The practical SoundLab for architects: Sound parameters as a design tool. Juergen Bauer (Dept. of Architecture, Waterford Inst. of Technol., Granary, Hanover St., Waterford, Co Waterford, Ireland, jbauer@wit.ie)

The experience gained from exploring acoustics with architectural students in Waterford Institute of Technology (Ireland) suggests that a studio situation can be exploited and used as a case study for young designers to become familiar with basic sound phenomena. From phase 1 of previous research, the student "audiovisual design workshop" concluded that class rooms that perform poorly

acoustically can help to strengthen the students' understanding of the sonic environment, in a way that the students were exposed day after day to high reverberation, low speech intelligibility, and the "Lombard" effect in their own studio. However, becoming aware of poor acoustics necessitates a creative follow-up: how to design and create good acoustics, or how to develop the environment as a "soundscape"? Consequently, phase 2 of our audiovisual design workshop introduced the practical sound lab: our students were required to design and to build acoustic panels for their own studio environment. These prototypes would be analyzed as part of an Acoustic Laboratory, with specialist advise from an Acoustic Consultant. The paper summarizes the findings from the audiovisual design laboratory. It also reflects on how to compile international case studies of buildings with good acoustics that can be used as references for designing architects.

3:00–3:20 Break

Contributed Papers

3:20

2pNSb7. Evaluation of noise climate in a campus environment using geospatial technology. Rajasekar Elangovan (Ctr. for Excellence and Futuristic Developments, Larsen & Toubro Construction, L&T ECC, Manapakkam, Chennai, TamilNadu 600089, India, erajas@gmail.com), Pari YadavaMudaliar (GeoSpatial Technol., EDRC, Larsen & Toubro Construction, Chennai, India), Venkateswaran Rajan, Khaleel Elur Rahaman Anser Basha (Ctr. for Excellence and Futuristic Developments, Larsen & Toubro Construction, Chennai, India), and Ravi Muttavarapu (GeoSpatial Technol., EDRC, Larsen & Toubro Construction, Chennai, India)

This study focuses on experimental evaluation of outdoor noise climate in an office campus, which is spread across 27 acres and investigation of its impact on the adjoining office buildings. For this purpose, a base map of the campus was prepared from the recent high spatial resolution satellite image using ESRI Arc/GIS and the same was used for planning appropriate locations for capturing noise levels and its spectral characteristics. Mapping grade Trimble GPS was used to stake out the measurement locations and noise levels were recorded using 01 dB solo and Norsonic type 118 sound level meters. Captured noise levels were plotted in GIS environment and appropriate spatial interpolation was carried out in order to give a continuous graphical representation of sound levels. A wide variation of noise levels was observed across the campus with LAeq ranging from 50 dB(A) to 80 dB(A). Low frequency noise was found to be predominant compared to mid and high frequency noise. Major noise sources and the propagation pattern were determined through the mapping. The data thus obtained is used to investigate the noise impact on the office buildings. The measurements were also accompanied by a subjective evaluation of the outdoor noise annoyance.

3:40

2pNSb8. On the noise generated by park visitors along hiking trails. Lucas Newman-Johnson (Univ. HS, Univ. of Illinois, 808 w. White St., Champaign, IL 61821, newmanj1@uni.illinois.edu) and Paul D. Schomer (Schomer and Assoc., Champaign, IL)

In assessing the sound environment of a park, visitor noise along trails may be important because of its affect upon: (1) the un-altered park environment, (2) wildlife, and (3) other park visitors. Just as noise in residential communities has been correlated with population density we set out to see if the noise level along trails would correlate with visitor density. In the summer of 2011, measurements were made using two sites, one on each of two trails. These measurements included one second Leq measurements at an array of four trailside microphones, and recording of the number of park visitors entering the measurement zone during each minute. Examination of the data revealed little correlation between a 5 min measurement of the Leq and density of trail visitors. The problem was that the smallest time increment in which we could accurately portray the number of park visitors was about 5 min, using the one minute totals, and in a 5 min period visitor noise would rarely equal or exceed the measurement ambient. Thus, no relation between visitor density and trailside noise could be developed. Additional analysis was done with data to better understand the limiting factors to the measurement.

4:00

2pNSb9. Investigating soundscape affordances through activity appropriateness. Frederik L. Nielbo (Ctr. for Semiotics, Aarhus Univ., Jens Chr. Skous Vej 2, Bldg. 1485, Rm. 525, Aarhus DK-2200, Denmark, norfln@hum.au.dk), Daniel Steele (Ctr. for Interdisciplinary Res. in Music Media and Technol., McGill Univ., Montréal, QC, Canada), and Catherine Guastavino (School of Information Studies, McGill Univ., Montréal, QC, Canada)

Central to the concept of soundscape is the understanding of the acoustic environment in context. Previous research indicates that people understand soundscapes through their potential for activities. One way to look at activities is through the concept of affordances—defined as the actionable properties of an object. In this study, the object is a location and time in the city. Fifteen participants listened to stereo recordings of eight outdoor sites in Paris and Montreal. In each trial, they evaluated on a continuous scale how appropriate the soundscapes were for a given activity. Four activities were considered and presented in random order: studying for an exam, meeting up with a friend, riding a bike and relaxing. Participants justified their ratings in free-format comments. A 8(Soundscapes) x 4(Activities) factorial ANOVAs revealed significant effects of Soundscape and Activities and Soundscape*Activities on appropriateness ratings. Certain soundscapes were found to accommodate specific activities only while others were found to potentially accommodate all activities or none (prominent mechanical/traffic noise). We also analyzed comments to further understand how participants envision utilizing the soundscape/environment and attribute meanings to the various sounds present.

4:20–5:00 Panel Discussion

Session 2pPA

Physical Acoustics: Material Characterization

Noureddine Atalla, Cochair
GAUS Mech. Eng., Univ. of Sherbrooke, Sherbrooke, QC J1K 2R1, Canada

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

Contributed Papers

1:00

2pPA1. Probing acoustics of liquid foams by optical diffusive wave spectroscopy. Benjamin Dollet, Marion Erpelding, Reine-Marie Guillermic, Caitlin Sample, Juliette Pierre, Arnaud Saint-Jalmes, and Jérôme Crassous (Institut de Physique de Rennes, CNRS/Université Rennes 1, Campus Beaulieu, Bâtiment 11A, Rennes 35042, France, benjamin.dollet@univ-rennes1.fr)

Sound propagation through liquid foams, which are dispersions of gas bubbles in a continuous liquid phase, is not well known yet. To characterize foam acoustics at the local scale, we have studied the effect of an external acoustic wave on bubble displacements inside an aqueous foam. We quantify these displacements by using a technique based on optical diffusive wave spectroscopy, that we specially developed to resolve tiny deformations in materials. Bubble displacements induce a modulation on the photon correlation curve. Measurements for various sound frequencies and amplitudes are interpreted using a light diffusion model. It allows us to unravel a nontrivial acoustic displacement profile inside the foam; in particular, we find that the acoustic wave creates a localized shear in the vicinity of the solid walls holding the foam, as a consequence of inertial contributions. This study of how bubbles "dance" inside a foam as a response to sound turns out to provide new insights on foam acoustics and sound transmission into a foam, foam deformation at high frequencies, and analysis of light scattering data in samples undergoing nonhomogeneous deformations.

1:20

2pPA2. Reduction of ultrasonic multiple scattering applied to flaw detection with array probes in polycrystalline materials. Sharfine Shahjahan (Site des Renardieres, EDF R&D, Moret-sur-Loing, France), Alexandre Aubry (Institut Langevin - ESPCI ParisTech & CNRS, Universite Paris Diderot, 1 rue Jussieu, Paris 75005, France), Fabienne Rupin, Bertrand Chassignole (Site des Renardieres, EDF R&D, Moret-sur-Loing, France), and Arnaud Derode (Institut Langevin - ESPCI ParisTech & CNRS, Universite Paris Diderot, Paris, France, arnaud.derode@espci.fr)

Flaw detection using ultrasonic evaluation of coarse-grain steels is perturbed by a high structural noise due to scattering. This leads to a decrease of the detection capabilities, particularly at high frequencies and large depths for which multiple scattering dominates. Recent academic studies have shown that the contribution of multiple scattering could be dramatically reduced. These results were obtained on a model random medium made of parallel steel rods immersed in water. The ability to detect a target could be significantly increased using a specific filtering method, based on the full matrix capture (F.M.C.) combined with a smart post-treatment based on random matrix theory, in supplement with the DORT method (i.e., decomposition of the time-reversal operator). Here, the same technique to separate simple and multiple scattering contributions is now applied to a real material. Experimental results were obtained on a nickel-based alloy (Inconel600®) with a thermically induced coarse grain structure and manufactured flaws (side drilled holes) at different depths. The experimental set-up used a multi-element ultrasonic array. Results are presented and compared to other detection techniques, at various depths and frequencies. Despite a dominant multiple scattering noise, a significant improvement of the detection performances is observed.

1:40

2pPA3. Negative and density-near-zero acoustic metamaterials based on quasi-two-dimensional phononic crystals. Victor Manuel Garcia-Chocano, Rogelio Graciá-Salgado, Daniel Torrent, and José Sánchez-Dehesa (Dept. of Electron. Eng., Polytechnic Univ. of Valencia, C/Camino de Vera S/N, Departamento de Ingeniería Electrónica, Valencia 46022, Spain, vicgarch@upvnet.upv.es)

A phononic crystal consisting of an array of cylindrical boreholes in a two dimensional waveguide has been fabricated and characterized. Reflectance and transmittance have been measured in a slab with seven layers of scatterers. The acoustic bands as well as the effective parameters have been extracted from experimental data, showing that the proposed structure behaves as an acoustic metamaterial with negative bulk modulus. In addition it is shown that the inclusion of anisotropic effects through angular-dependent structures inside the boreholes leads to metamaterials with double negative parameters. This feature allows the observation of interesting phenomena such as an acoustic tunneling, which has been predicted through full wave simulations in a density-near-zero metamaterial.

2:00

2pPA4. Impulse response of a medium in a three layered media. Ambika Bhatta (Elec. Eng., Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, ambika_bhatta@student.uml.edu), Charles Thompson, Kavitha Chandra (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, Lowell, MA), and Vineet Mehta (MIT, Lincoln Lab., Lexington, MA)

This paper investigates the point source response of a layered medium separating two semi-infinite impedance medium. A detailed numerical method to evaluate Laplace transform of the reflection coefficient for each boundary is presented. A description of the branch integral dominantly contributing to the reflection coefficient in Laplace domain for single and multiple reflections is given. The analysis utilizes the image theory to evaluate the order of the reflection and complex pressure amplitude at each image source location. The impulse response is also evaluated numerically from the Green's function obtained from the reflection coefficient and complex pressure due to image sources. Fresnel's plane wave reflection coefficient validates the solution when the height of layer containing the source and the observation position is relatively large. The obtained expression for reflection coefficient for large wave number and observation position is verified numerically.

2:20

2pPA5. Experimental demonstration of a three-dimensional acoustic cloak based on a cancelation effect. Jose Sanchez-Dehesa, Victor M. García-Chocano, Alfonso Climente, Francisco Cervera (Dept. of Electron. Eng., Universitat Politecnica de Valencia, Camino de vera s.n., Edificio 7F, Valencia, Valencia ES-46022, Spain, jsdehesa@upvnet.upv.es), Vicente Gomez-Lozano (Centro de Tecnologías Físicas, Universitat Politecnica de Valencia, Valencia, Spain), Lorenzo Sanchis (UMDO (Unidad Asociada al CSIC-IMM), Universidad de Valencia, Valencia, Spain), Rafael LLopis-Pontiveros, and Martínez-Pastor Juan (UMDO (Unidad Asociada al CSIC-IMM), Universidad de Valencia, Valencia, Spain)

A three-dimensional acoustic cloak has been designed, fabricated, and experimentally characterized. The cloak is made of 60 concentric tori acoustically rigid that surround a sphere of radius 4 cm, which is considered as the cloaked object. The major radii and positions of the tori along the symmetry axis are determined by a cancelation condition; i.e., the scattering cross section by the sphere and the tori must be zero. An optimization algorithm that combines a genetic algorithm and simulated annealing is employed to satisfy such a condition. The operational frequency of the one-directional cloak is 8.67 kHz with a bandwidth of about 120 Hz.

2:40

2pPA6. Ice thickness determination with audio sound. Anurupa Shaw (Georgia Tech Lorraine, 133 Rue Du Fort de Queuleu, D107, Residence Lafayette, Metz 57070, France, shaw.anurupa@gmail.com) and Nico F. Declercq (Georgia Inst. of Technol., George W. Woodruff School of Mech. Eng., Lab. for Ultrasonic Destructive Evaluation "LUNE", Georgia Tech-CNRS UMI2958, Georgia Tech Lorraine, Metz-Technopole, France)

The objective of this study is to determine the thickness of Ice by analyzing the sound spectrum generated by dispersion of lamb waves propagating in ice. In winters when the lakes and rivers freeze, it is important to know the thickness of the ice layer before one intends to walk on it. When we throw a stone on the ice layer, we can hear a fluting noise. This is recorded for different thicknesses of ice and the sound spectrum is compared with the results simulated using a parameterized model. This model is created using a combination of plane waves for different incident angles and frequencies to generate dispersion curves for different thicknesses of ice. The frequencies of the reflected sound are then compared with the frequencies of musical notes in order to assign different musical notes to different thicknesses of ice.

3:00

2pPA7. Shaving foam: A complex system for acoustic wave propagation. Juliette Pierre (Laboratoire Matière et Systèmes Complexes, UMR CNRS 7057 - Univ. Paris Diderot, Université Rennes 1 - CNRS UMR 6251 263 av. Général Leclerc, Rennes 35042 Cedex, France, juliet.pierre@gmail.com), Valentin Leroy (Laboratoire Matière et Systèmes Complexes, UMR CNRS 7057 - Univ. Paris Diderot, Paris, France), Arnaud Saint-Jalmes, Benjamin Dollet, Imen Ben Salem, Jérôme Crassous, Reine-Marie Guillermic (Institut de Physique de Rennes, UMR CNRS 6251 - University Rennes 1, Rennes, France), Wiebke Drenckhan (Laboratoire de Physique du Solide, UMR CNRS 8502 - Univ. Paris-Sud, Orsay, France), and Florence Elias (Laboratoire Matière et Systèmes Complexes, UMR CNRS 7057 - Univ. Paris Diderot, Paris, France)

While liquid foams have applications in an increasing number of industrial areas (food, cosmetic, or petroleum industry), it remains difficult to non-invasively probe their structure and/or composition. Since the propagation of acoustic waves is very sensitive to parameters such that the liquid fraction, the bubble size distribution, or even the nature of the liquid phase, acoustic spectroscopy could be a very powerful tool to determine the structure and/or composition of liquid foams. In this context, we present an investigation of the acoustic properties of a useful and common foam, often considered as a model system: shaving foam. Phase velocity and attenuation of acoustic waves in a commercial shaving foam (Gillette) were measured over a broad frequency range (0.5 to 600 kHz), using four different experimental setups: an impedance tube (0.5–6 kHz), an acousto-optic setup based on diffusive wave spectroscopy (1–10 kHz), and two transmission setups with narrow-band (40 kHz) and broad-band (60–600 kHz) transducers. We present the results and discuss the advantages and shortcomings of each setup in terms of a potential spectroscopy technique.

3:20

2pPA8. Lamb modes and acoustic microscopy for the characterization of bonded structures. Alaoui Ismaili Naima, De Mello Da Silva Camilla (Institut Electronique du Sud (IES), UMR CNRS 5214, University of Montpellier 2, Place Eugène Bataillon, Montpellier 34095, France, naima.alaoui-ismaili@insa-lyon.fr), Ech-Cherif El-Kettani Mounsif (Laboratoire Ondes et Milieux Complexes (LOMC), UMR CNRS 6294, Univ. of Le Havre, Le Havre, France), Despaux Gilles (Institut Electronique du Sud (IES), UMR CNRS 5214, Univ. of Montpellier 2, Montpellier, France), Rousseau Martine (Institut Jean Le Rond d'Alembert UMR CNRS 7190, Univ. of Pierre et Marie Curie, Paris, France), and Izbicki Jean-Louis (Laboratoire Ondes et Milieux Complexes (LOMC), UMR CNRS 6294, Univ. of Le Havre, Le Havre, France)

This paper is a contribution to the evaluation of the bonding by means of guided waves, when the quality of the glue is degraded by addition of grease. Experimental dispersion curves are compared to the theoretical ones obtained either from a perfect welded tri-layer model or from a second model using delaminated conditions. It is shown that the second model is more convenient when the quality of the glue is highly degraded. On another hand, the parameters of Jones are extracted from experimental data by inverse problem. It is shown that the values of these parameters strongly diminish and the variation is of logarithmic type when the quality of the glue is progressively degraded. Acoustic microscopy is also performed, and it provides images of the inner structure giving a qualitative explanation of the values of the parameters of Jones.

3:40

2pPA9. Experimental measurements of the coherent field resulting from the interaction of an ultrasonic shock wave with a multiple scattering medium. Nicolas Viard, Bruno Gianmarinaro, Arnaud Derode, and Christophe Barrière (Institut Langevin, 1, rue Jussieu, Paris 75005, France, nicolas.viard@espci.fr)

Whereas multiple scattering and shock wave formation are known to be antagonistic phenomena, this work concentrates on the interaction of an ultrasonic shock wave with a random multiple scattering medium. The shock wave is generated by long distance propagation of a short pulse (four periods at a 3.5 MHz central frequency) in water before it encounters the scattering medium (a slab-shaped random set of parallel metallic rods). Transmitted waves are recorded over hundreds of positions along the lateral dimension of the slab to estimate the ensemble-averaged transmitted field $\langle \Phi(t) \rangle$, also known as the coherent wave. Experiments are repeated for different thicknesses L of the slab and different emission amplitudes. The elastic mean free path l_e (i.e., the typical distance for the decreasing of the coherent intensity $|\langle \Phi(t) \rangle|^2$ due to scattering) is determined as well as the harmonic rate of the averaged transmitted wave. Experimental results are discussed and compared to the linear case.

4:00

2pPA10. Frequency-resolved measurements of the diffusion constant for ultrasonic waves in resonant multiple scattering media. Nicolas Viard and Arnaud Derode (Institut Langevin, 1, rue Jussieu, Paris 75005, France, nicolas.viard@espci.fr)

Experimental measurements of the diffusion constant for ultrasonic waves (around 3 MHz) propagating in water through a random set of scatterers (parallel metallic rods arranged as a slab) are presented. The slab thickness is around 10 times the transport mean free path. Transmitted waves are recorded over hundreds of emitting/receiving positions in order to estimate the ensemble-averaged transmitted intensity $\langle I(x,t) \rangle$. Focused beamforming is performed on both faces of the sample in order to mimic a set of point-like sources and receivers. In theory, under the diffusion approximation, the ratio of the off-axis intensity $\langle I(x,t) \rangle$ to the on-axis intensity $\langle I(0,t) \rangle$ shows a simple gaussian dependence on the lateral dimension x, independently from absorption or boundary conditions. This yields a simple way to estimate the diffusion constant D and therefore characterize the scattering medium. Based on that method, broadband as well as frequency-resolved measurements of the diffusion constant are presented in controllable model media, such as these forests of steel rods. Experimental results and difficulties for measuring a reliable value for D on a real sample are discussed.

4:20

2pPA11. Wavetrain-long waves interaction in a non-homogeneous, non-stationary medium. Alexander Voronovich (Physical Sci. Div., NOAA/ESRL, 325 Broadway, Boulder, CO 80305, alexander.voronovich@noaa.gov)

A weakly nonlinear wavetrain of the acoustic waves apart from higher harmonics generates also a low-frequency, large-scale motion which in turn affects both the wavetrain itself and other wavetrains, thus leading to an effective wavetrains interaction. A closed set of equations describing interaction of a wavetrain with inhomogeneous medium including a recoil effect on the background motion is derived using Hamiltonian form of the equations of motion (viscosity is neglected). Corresponding energy and momentum conservation equations are derived which account for the energy and pseudo-momentum exchange between the wavetrain and a background, large-scale motion (the latter includes the nonlinearly generated component due to the wavetrains). A large-scale motion due to a single wavetrain propagating in an initially homogeneous medium is calculated. Far from the wavetrain nonlinearly generated large-scale current is confined by an expanding sphere with the velocity aligned with the wavetrain propagation direction; the velocity appears to be constant in the planes perpendicular to this direction. It is demonstrated that interaction between acoustic wavetrains lead to their effective repulsion which in a complex way depends on mutual location and velocities of the wavetrains. Generalization of the approach to the case of integral gravity waves will be also briefly mentioned.

4:40

2pPA12. Gaussian closure technique for Bouc's hysteretic model under white noise excitation. Holger Waubke (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14, Vienna 1040, Austria, holger.waubke@oeaw.ac.at)

Bouc developed a hysteretic model for materials like rubber under dynamic excitation. The response of a hysteretic system under white noise excitation is normally estimated by means of the statistical linearization or a related method. Disadvantages of this method are the assumption of a Gaussian nature of the random distributions, the high computational efforts caused by the iterations needed, and the instability of the iteration in certain parameter regions. Using the assumption of Gaussian random distributions, the Gaussian closure technique can be applied. Analytic solutions of the integrals occurring in this approximation were found and are presented. These solution allow for an explicit time step procedure for the random moments in the transient case. For the stationary case, a fast and stable iteration about a set of non linear equations is needed. Both procedures allow to calculate the moments in a fast manner and allow to solve problems with more than one degree of freedom with limited computational efforts.

5:00

2pPA13. Numerical model of a causal and fractional all-frequency wave equation for lossy media. Margaret Wismer (Bloomsburg, 400 E Second St., Bloomsburg, PA 17815, mwismer@bloomu.edu)

A numerical algorithm, to simulate a lossy acoustic wave equation with fractional time derivative terms, is presented. The inclusion of fractional derivatives yields a causal acoustic wave equation which can model and predict power law attenuation for which the level of absorption is proportional to frequency raised to a non-integer power. The fractional Zener wave equation is derived from the fractional Zener stress-strain constitutive relation and contains two absorption terms. One term which includes the Laplacian, similar to the traditional wave equation in viscous fluids, has a coefficient, which is proportional to a relaxation time. The other term is a fractional time derivative, higher than second order, and is proportional to the creep or retardation time. Both relaxation and retardation parameters will affect the frequency behavior of the attenuation. The inclusion of two terms enables the modeling of power-law attenuation in all frequency ranges. It results in more stable numerical simulations as both the overall mass and stiffness matrices of the finite element algorithm are changed according to the level of absorption. Results show the animation of diffraction patterns of focused and planar acoustic waves by inclusions with different values for the two different parameters, retardation time and relaxation time.

TUESDAY AFTERNOON, 4 JUNE 2013 — 514ABC, 12:55 P.M. TO 4:20 P.M.

Session 2pPPa

Psychological and Physiological Acoustics: Celebrating a "Long" Career: Explorations of Auditory Physiology and Psychoacoustics

Jungmee Lee, Cochair
Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208

Elizabeth A. Strickland, Cochair
Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907

Chair's Introduction—12:55

Invited Papers

1:00

2pPPa1. Glenis Long's contribution to animal psychoacoustics. Richard Fay (MBL, 179 Woods Hole Rd., Falmouth, MA 02540, rfay@luc.edu)

Glenis and I were psychology graduate students together in 1970–1971. She was interested in auditory psychophysics, but soon developed an interest in comparative psychophysics which she pursued in a post-doc and a faculty position in Germany, where she obtained a behavioral audiogram for the horseshoe bat and later studied masking in the same species. This audiogram was one of the very first psychophysical investigations of hearing in any bat and the very first for the horseshoe bat. It confirmed a rather complex frequency response function with sensitivity peaks at about 20 kHz and 60 kHz, and a very sharp peak at about 80 kHz. The masked thresholds indicated rather normal critical masking ratios (CR) except in a narrow region at about 80 kHz where the CRs are up to 15 dB

lower than expected, suggesting a critical band of about 300 Hz at 80 kHz that is presumably used in Doppler-shift compensation. In 1981–1983, Glenis went on to investigate frequency and rate modulation discrimination in the chinchilla, for the first time, and to study tone-on-tone masking in the chinchilla. Since these early works, Glenis has maintained her comparative interests with studies on birds, kangaroo rats, frogs, and fleas on cats.

1:20

2pPPa2. Understanding subtle changes in auditory function with otoacoustic emissions. Linda J. Hood (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, MCE South, 8310, Nashville, TN 37232, linda.j.hood@vanderbilt.edu), Shanda Brashears (A. I. du-Pont Hospital for Children, Wilmington, DE), Glenis Long (The Grad. Ctr., City Univ. of New York, New York, NY), and Carrick Talmadge (Univ. of Mississippi, Oxford, MS)

Otoacoustic emissions (OAEs), a sensitive measure of cochlear processing, may be altered by subtle changes in auditory function that are not measurable by usual clinical methods. We studied auditory function in carriers of genetic mutations related to recessive hereditary hearing loss where we hypothesized that carrying a single mutation copy may compromise auditory function and be reflected when sensitive assays are used. Parents and siblings who were confirmed carriers of recessive mutations associated with GJB2 (connexin 26) or Usher syndrome, as well as obligate carriers of mutations related to recessive hearing loss of unknown genetic origin, were compared to age and gender matched control subjects. All participants had normal pure tone and middle ear responses. Metrics included transient OAEs, distortion product OAEs, and OAE fine structure. DPOAE fine structure was specifically explored based on the ability to isolate components that could be differentially affected by genetic mutations. The results support the hypothesis that carriers of gene mutations related to hearing loss display subtle auditory abnormalities that can be observed in OAEs. These findings will be related to other studies of subtle changes in OAEs in disorders affecting auditory function. [Work supported by NIH NIDCD R01-DC03679 and VU Development Funds.]

1:40

2pPPa3. Brain activity and perception of gaze-modulated tinnitus. Pim Van Dijk, Margriet J. Van Gendt, Kris Boyen, Emile De Kleine (Otorhinolaryngology, Univ. Med. Ctr. Groningen, P.O. Box 30001, Groningen 9700 RB, Netherlands, p.van.dijk@umcg.nl), and Dave R. Langers (NIHR Nottingham Hearing Biomed. Res. Unit, Univ. of Nottingham, Nottingham, United Kingdom)

We studied the correspondence between brain activity and tinnitus in subjects with gaze-modulated tinnitus. These subjects are able to modulate their tinnitus by peripheral gaze of the eyes. This is a rare form of tinnitus that primarily occurs in subjects that underwent acoustic schwannoma surgery. The voluntary control of the tinnitus allows for a controlled experiment to study the perceptual characteristics of tinnitus and the corresponding brain activity as assessed by functional MRI. Eighteen subjects with gaze-modulated tinnitus participated in the study. The effect of gaze on tinnitus was diverse. Most commonly, the largest effect on tinnitus was observed for horizontal gaze toward the surgery side. When the loudness of tinnitus changed, it was usually an increase. In addition, changes of the pitch and apparent bandwidth of the tinnitus were reported. Peripheral gaze corresponded to increase of activity in the cochlear nucleus and inferior colliculus, a decrease of activity in the medial geniculate body, and a reduction of deactivation in the auditory cortex. The inhibition of the medial geniculate body in the thalamus contrasts with the excitation that is typically observed in response to external sound stimuli. It suggests that abnormal functioning of the thalamus plays a role in tinnitus.

2:00

2pPPa4. Improving the usability of the distortion product otoacoustic emisssions-sweep method: An alternative artifact rejection and noise-floor estimation. Manfred Mauermann (Med. Phys., Univ. of Oldenburg, Carl-von-Ossietzky-Straße 9-11, Oldenburg 26111, Germany, manfred.mauermann@uni-oldenburg.de)

The DPOAE-sweep method combined with a specific least-squares-fit (LSF) analysis provides a fast method to measure and analyze distortion product otoacoustic emissions (DPOAE) with a high frequency resolution. In studies using this technique the noise reduction, artifact rejection and noise estimation are typically realized in a “classical” way, i.e., as temporal averaging with preceding elimination of time epochs exceeding an artifact threshold level and a noise estimation based upon the analysis of the difference of two buffers of epoch averages. However, the choice of an artifact threshold is arbitrary to some extent and different choices can lead to differences in the estimation of DPOAE levels. The two-buffer technique is ambiguous as well since a different grouping of the epochs into the buffers leads to rather different noise estimates. Therefore, the present study proposes an alternative approach, which provides unique noise estimators for a given set of data, a robust artifact rejection without the need to select an arbitrary rejection threshold, as well as estimators including confidence intervals for the DPOAE levels and phases. The “classical” and suggested estimators for DPOAE levels, DPOAE phases, and noise levels are compared based on Monte Carlo simulations and real measured data sets.

2:20–2:40 Break

2:40

2pPPa5. The relevance of otoacoustic emission fine structure. Sumitrajit Dhar (Roxelyn and Richard Pepper Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, s-dhar@northwestern.edu)

The quasi-periodic fluctuations in otoacoustic emission (OAE) amplitude and phase, known as fine structure, provide critical insight into the mechanisms of OAE generation and propagation. Whether fine structure is relevant from the perspective of hearing health remains an open question. Starting with work done under the tutelage of Professor Long, our group has conducted a decade of work investigating the clinical relevance of fine structure in general, and distortion product (DP) OAE components in particular. These studies cover a significant portion of the human lifespan (0 to ~80 yr) and the collective results provide insights into the avenues of application that hold promise. Drawing from the results of several studies conducted in our laboratory and those of collaborators and colleagues we will make the case for focused clinical application of the DPOAE components that constitute fine structure. These constituent elements appear to be ideal for monitoring of cochlear status and expose physiological changes prior to their evidence in other measures of

auditory function. However, harnessing the information encapsulated in fine structure directly is not without challenge and the examination of the constituent elements of fine structure may prove more profitable for clinical applications.

3:00

2pPPa6. Noise-induced hearing loss and strategies for its prevention in the New Zealand population: The Kiwi connection. Peter R. Thorne, Gavin Coad, Ravi Reddy, and David Welch (Audiology, Univ. of Auckland, School of Population Health, Private Bag 92019, Auckland 1142, New Zealand, pr.thorne@auckland.ac.nz)

Celebrating Glenis Long's outstanding contribution to auditory physiology and psychoacoustics, this presentation covers the New Zealand connection and particularly her involvement in our research program into the monitoring and prevention of noise-induced hearing loss (NIHL) in the country. A large multidisciplinary project is being undertaken to investigate the nature of occupational hearing loss in New Zealand and establish a national approach to prevent NIHL. This includes estimates of NIHL prevalence and the design and evaluation of education and prevention programs to reduce the impact of noise. Using a modeling approach, we have estimated that NIHL contributes to 17–25% of cases of hearing impairment in New Zealand and is therefore a significant modifiable risk factor. A key component of our project is monitoring of noise injury and we have also studied distortion product otoacoustic emissions (DPOAE) as a measure of early injury. To assess DPOAEs as a measure of injury, we recorded them using swept pure tones and extracted DPOAE components using a least-squares fit approach in noise and non-noise exposed individuals. OAE findings were compared with measures of auditory function. We found that the generator component correlated more strongly with auditory threshold and thus may be a better physiological index of noise injury. Overall, these findings have informed a national strategy involving government and community agencies to mitigate the effects of noise.

3:20

2pPPa7. Demonstration of distributed distortion-product otoacoustic emission components using onset-latency techniques. Brenda L. Lonsbury-Martin (Otolaryngology, Loma Linda Univ. Med. Ctr., Res. Service (151), 11201 Benton St., Loma Linda, CA 92357, blonsbury-martin@llu.edu), Glen K. Martin, and Bart B. Stagner (Res. Service, VA Loma Linda Healthcare System, Loma Linda, CA)

An oversimplified notion is that DPOAEs originate from a restricted region on the basilar membrane (BM). In actuality, DPOAEs are a distributed process involving the interaction of many wavelets, most likely generated over a broad region at, and basal to the overlap place of the primary tones. In the present study, DPOAEs were measured in rabbits as time waveforms by using phase rotation to cancel all components in the final average, except the 2f1-f2 DPOAE. At times, f2 was turned off for 6 ms producing a gap so that the DPOAE was no longer generated. These procedures allowed the DPOAE onset as well as the decay during the gap to be observed in the time domain. Results showed that complexities emerged near the onset of the DPOAE time waveform as the f2/f1 ratio decreased, and at the beginning of the gap when f2 was turned off. Such complexities were unaffected by interference tones (ITs) near the DPOAE. However, these complexities were removed by ITs presented above f2, which can be explained by the interactions of distributed DPOAE components with different phase relationships.

3:40

2pPPa8. Distortion-product otoacoustic emission generator and reflection components in newborns, infants, and adults. Beth Prieve (Commun. Sci. and Disord., Syracuse Univ., 805 S. Crouse Ave., Syracuse, NY 13244, baprieve@syr.edu), Glenis Long (Speech-Lang.-Hearing Program, City Univ. of New York Grad. Ctr., New York, NY), and Carrick Talmadge (National Ctr. for Physical Acoust., University, MS)

Glenis Long and colleagues (Talmadge *et al.*, J. Acoust. Soc. Am. **105**, 275) were among the first to model and describe the characteristics of distortion-product otoacoustic emissions (DPOAEs) using two sub-components from different, cochlear sources. It is now accepted that there is a generator component that results from inter-modulation distortion created by nonlinearity in the outer hair cell near the f2 place. The reflection component predominantly arises from coherent reflection near the characteristic place corresponding to the frequency of the distortion product. Because the two components are generated through different mechanisms, it has been hypothesized that they may be differentially affected by human development. The goal of this presentation is to discuss ongoing research of DPOAE components in newborns, infants and adults. Analysis of the components indicates that the relationship between growth rates for generator and reflection components are significantly different between infants and adults. Furthermore, the phase functions for both components are different among groups. Possible sources for these differences will be discussed. [Funded by the March of Dimes Birth Defects Foundation].

4:00–4:20 Panel Discussion

Session 2pPPb

Psychological and Physiological Acoustics: Speech, Attention, and Impairment (Poster Session)

Jayaganesh Swaminathan, Chair
Boston Univ., 635 Commonwealth Ave., Rm. 320, Boston, MA 02215

Contributed Papers

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

2pPPb1. Factors affecting frequency discrimination in school-aged children and adults. Crystal N. Taylor (Allied Health Sci., UNC Chapel Hill, Dept. of Allied Health Sci., CB 7190, Chapel Hill, NC 27599, Crystal_Taylor@med.unc.edu), Emily Buss (Otolaryngol.—Head and Neck Surgery, UNC Chapel Hill, Chapel Hill, NC), and Lori J. Leibold (Allied Health Sci., UNC Chapel Hill, Chapel Hill, NC)

Auditory frequency discrimination is a basic ability that may limit the maturation of speech and language skills in some listeners. Despite its importance, the factors affecting frequency discrimination in school-aged children are poorly understood. The goal of the present study was to evaluate effects related to memory for pitch, musical training, and the utilization of temporal fine-structure cues. Listeners were normal-hearing children, 5.1–13.6 years old, and adults. One subgroup of children had musical training (>150 h) and the other did not. The standard stimulus was either a 500- or a 5000-Hz pure tone, and the target stimulus was either a tone of higher frequency or a frequency-modulated tone (2- or 20-Hz rate) centered on the standard frequency. As commonly observed, mean frequency discrimination thresholds tended to be elevated in younger listeners. This developmental effect was smaller for FM detection than for pure-tone frequency discrimination, consistent with an effect of memory for pitch. The child/adult difference tended to be smaller for musically trained than untrained children. Children were not particularly poor at 2-Hz FM detection for the 500-Hz standard, a condition thought to rely on temporal fine-structure cues. [Work supported by NIDCD R03DC008389.]

2pPPb2. Cognitive and auditory influences on speech recognition by younger, middle-aged, and older listeners. Karen S. Helfer and Angela Costanzi (Commun. Disord., Univ. of Massachusetts Amherst, 358 N. Pleasant St., Amherst, MA 01002, khelfer@comdis.umass.edu)

In this study, we measured hearing thresholds and cognitive abilities (working memory, processing speed, executive function, and inhibition) in our participants to determine how these factors relate to speech understanding in the presence of competing speech. Participants were younger, middle-aged, and older adults, with the older adults having hearing thresholds that ranged from normal to a moderate hearing loss. The target stimuli for this study were recordings of TVM sentences [Helfer and Freyman (2009)] presented with various types of maskers (one or two other TVM sentences spoken by same-sex maskers, samples of running speech spoken by one or two same-sex talkers, and single-channel signal-envelope-modulated noise) at several signal-to-noise ratios. This poster will discuss speech recognition performance in the framework of multimasker penalties and will detail connections among hearing thresholds, cognitive abilities, and speech understanding. [Work supported by NIH DC012057.]

2pPPb3. Effects of reverberation and spatial diffuseness on the speech intelligibility of public address sounds in subway platform for young and aged people. Yong Hee Kim and Yoshiharu Soeta (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol. (AIST), 1-8-31 Midorigaoka, Ikeda, Osaka 563-8577, Japan, yh.kim@aist.go.jp)

This paper investigated how young and aged people response differently on the public address sounds in subway platform according to various reverberation and diffuseness. Both word intelligibility (WI) and listening difficulty (LD) tests were adopted as experimental method. Twelve simulated sound fields at the same position were prepared according to the combination of four different reverberation times (RT) and three different interaural cross-correlation coefficients (IACC). Auralized sounds with anechoic test words of actual station names were presented to young and aged subjects at the fixed sound pressure level. As results, LD results showed significant correlation with RT for both young and aged subjects, whereas significant correlation of WI results and RT was found only in aged subjects. Aged subjects showed worse speech intelligibility performances on public address sounds than young subjects due to their worse hearing level. RT was found as the most important factor to determine speech intelligibility for both young and aged subject, whereas aged subject showed better speech intelligibility performances with lower IACC. From the regression analysis, LD rating was estimated from the measured RT and IACC. Additionally, the effects of word familiarity, individual noise sensitivity and hearing level on the speech intelligibility were discussed.

2pPPb4. Auditory influence on tactile perception changes with age. Simon P. Landry (École d'orthophonie et d'audiologie, Université de Montréal, 7077 avenue du Parc, Montreal, QC H3N 1X7, Canada, simon.landry.4@umontreal.ca), Jean-Paul Guillemot (Département de kinanthropologie, Université du Québec à Montréal, Montreal, QC, Canada), and François Champoux (École d'orthophonie et d'audiologie, Université de Montréal, Montreal, QC, Canada)

Characteristics of auditory interaction with vision have been extensively studied. However, auditory system integration with other sensory modalities, such as the tactile system, lacks such thorough investigations. The objective of this study was to examine the effects of age on audiotactile integration in humans. Thirty-one participants between the ages of 20 and 65 were divided into three groups according to their age. Audiotactile integration was assessed using the "auditory flash illusion" in which 1, 2, 3, or 4 tactile stimuli were accompanied with 0, 1, 2, 3, or 4 auditory stimuli. Participants were asked to ignore auditory stimulations and report the number of tactile stimulations perceived. All participants were tested with task relevant auditory and tactile stimuli as a control measures and were shown to have similar abilities. However, groups differed during the experimental conditions. The youngest group reported a greater number of tactile stimuli than actually presented during the illusory experimental conditions. Participants in the middle and older age groups did not report this illusory tactile perception. These results suggest that age reduces predisposition to audiotactile integration. These results are consistent with developmental studies for multisensory integration in other sensory modalities.

2pPPb5. The influence of auditory training on measures of temporal resolution in younger and older adults. Meital Avivi-Reich (Psychology, Univ. of Toronto Mississauga, 1909-35 Charles st.W, Toronto, ON m4y 1r6, Canada, me_avv@yahoo.com), Stephen R. Arnott (Rotman Res. Inst., Baycrest, Toronto, ON, Canada), Tamara Tavares, and Bruce A. Schneider (Psychology, Univ. of Toronto Mississauga, Toronto, ON, Canada)

Deterioration in the ability to perceived rapid changes in auditory input is thought to contribute to the difficulties older adults experience when communicating in noise. Studies have demonstrated that the performance of

young adults on auditory tasks improves with training. However, few studies have tested the degree to which practice improves auditory performance in older adults. A previous study examined the extent to which younger and older adults benefited from training when the task was to detect a gap in a narrow-band noise centered at 1 kHz. Significant improvements occurred in both age groups, indicating that auditory learning can still occur later in life. The present study examines if training improves performance when more than one auditory filter is activated. Twenty-four younger and older participants were trained for 10 days to detect a gap between a 2-kHz and a 1-kHz noise. Performance was assessed one day and one month after the last training session along with the extent to which the benefits generalized to other frequencies and the untrained ear. In addition, event-related potentials (ERPs) were obtained pre- and post-training to assess cortical changes in the response to temporal gaps. Initial results show improvement throughout training in both age groups.

2pPPb6. Intelligibility of voiced and whispered speech in noise in listeners with and without musical training. Dorea Ruggles, Ariane Riddell (Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, druggles@umn.edu), Richard Freyman (Univ. of Massachusetts-Amherst, Amherst, MA), and Andrew J. Oxenham (Univ. of Minnesota, Minneapolis, MN)

Musicians have been shown to exhibit more robust neural coding of periodicity than non-musicians; they have also been reported to exhibit an advantage in understanding speech in noise. This study tested the hypothesis that the musicians' speech intelligibility advantage arises from more efficient coding of voiced (periodic) speech. This was tested by comparing intelligibility of normal speech in noise with that of whispered (unvoiced) speech in musicians and non-musicians. Listeners with less than 2 years of formal musical training were categorized as non-musicians; listeners who began musical training before age 10 and who currently play more than 10 h/wk were included as musicians. Listeners heard grammatically correct nonsense sentences that were either (1) voiced, (2) whispered, or (3) whispered with subband amplitude distributions matched to the voiced speech. Masking noise was either continuous or gated with a 16 Hz, 50% duty cycle. In contrast to the earlier study, preliminary data suggest no advantage for musicians over non-musicians in understanding voiced or whispered speech in either continuous- or gated-noise conditions. The results suggest that more investigation is needed to fully understand the nature of auditory and speech processing advantages imparted by musical training. [Wok supported by NIH Grant No. R01DC05216.]

2pPPb7. Level-dependent effects on speech intelligibility. Patricia Pérez-González and Enrique A. Lopez-Poveda (Neurosci. Inst. of Castilla y Leon, Univ. of Salamanca, Pintor Fernando Gallego 1, Salamanca, Salamanca 37007, Spain, patripg@usal.es)

The effect of noise on speech intelligibility is typically measured using fixed-level speech (or noise) and varying the speech-to-noise ratio (SNR). An assumption of this procedure is that intelligibility mostly depends on the SNR and barely depends on speech level. The effective SNR, however, (i.e., the SNR in the internal stimulus representation), possibly depends on peripheral compression. Indeed, compression could facilitate or hinder intelligibility for negative and positive SNRs, respectively. Insofar as compression varies with level, speech intelligibility might also vary with speech level. Here, we tested these hypotheses by measuring percent correct digit triplet identification as a function of speech level for fixed SNRs. Measurements were carried out for normal-hearing subjects and for hearing-impaired subjects with linear cochlear responses, as assessed using the temporal masking curve method. Results for both groups suggest that the detrimental effect of the noise on intelligibility is larger for speech levels near threshold, particularly for negative SNRs, a result that cannot easily be explained by compression. Alternative explanations for the result are discussed.

2pPPb8. Talker effects in speech band importance functions. Eric W. Healy, Sarah E. Yoho, Carla L. Youngdahl, and Frederic Apoux (Speech and Hearing Sci., The Ohio State Univ., Pressey Hall Rm. 110, 1070 Carmack Rd., Columbus, OH 43210, healy.66@osu.edu)

The literature is somewhat mixed with regard to the influence of (a) the particular speech material (e.g., sentences or words) versus (b) the particular talker used to create the recordings, on band-importance function (BIF) shape. One possibility is that previous techniques for creating BIFs are not sensitive enough to reveal these influences. In the current investigation, the role of talkers was examined using the compound technique for creating BIFs. This technique was developed to account for the multitude of synergistic and redundant interactions that take place among various speech frequencies. The resulting functions display a complex microstructure, in which the importance of adjacent bands can differ substantially. It was found that the microstructure could be traced to acoustic aspects of the particular talkers employed. Further, BIFs for IEEE sentences based on ten-talker recordings displayed less microstructure and were therefore smoother than BIFs based on one such talker. These results together suggest that the compound technique is sensitive enough to reveal acoustic aspects of the particular talker employed. It is further suggested that multiple talkers, rather than smoothing of the functions, be used if the goal is to describe speech more generally.

2pPPb9. Psychometric effects of adding realism to a speech-in-noise test. Virginia Best, Gitte Keidser, Jörg M. Buchholz, and Katrina Freeston (National Acoust. Lab. and the HEARing Cooperative Res. Ctr., 126 Greville St., Chatswood, NSW 2067, Australia, virginia.best@nal.gov.au)

The speech reception threshold (SRT) is routinely measured in the laboratory to assess speech understanding in noise, but is often reported to be a poor predictor of performance in real world listening situations. The overall goal of this work is to determine whether introducing realistic aspects to speech tests can better capture individual differences and ultimately produce more relevant performance measures. We examined the psychometric effects of (a) transplanting a standard sentence-in-noise test into a simulated reverberant cafeteria environment, and (b) moving from sentence recall to a new ongoing speech comprehension task. Participants included normal hearers and hearing-impaired listeners (who were tested with and without their hearing aids). SRTs in the cafeteria environment were significantly correlated with standard SRTs, but were poorer overall and more sensitive to hearing loss. The comprehension task, despite having very different demands to sentence recall, produced similar SRTs under these conditions. The benefit of hearing aids was weakly correlated across the two listening environments and the two listening tasks. These manipulations promise to be useful for the creation of realistic laboratory tests that are engaging and challenging, yet controlled enough to be useful for psychophysical experiments.

2pPPb10. The Glasgow Monitoring of Uninterrupted Speech Task: A naturalistic measure of speech intelligibility in noise. Alexandra MacPherson and Michael A. Akeroyd (MRC Inst. of Hearing Res. (Scottish section), Queen Elizabeth Bldg., 16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, alex@ihr.gla.ac.uk)

When listening to speech in noisy environments parts of the speech signal are often missed due to masking, degradation, or inattention. Sometimes the message can be reconstructed, but reallocating resources to recover the missed information can affect the efficiency and speed at which the message is understood. A slowing of processing will be particularly detrimental if there are few opportunities for understanding to catch up, e.g., when listening to the radio. It is likely then that the monitoring of uninterrupted flows of continuous speech will be especially sensitive to hearing or listening impairments. We developed a new task to test this. The Glasgow Monitoring of Uninterrupted Speech Task (GMUST) requires participants to listen to a 10-min segment of continuous speech while monitoring a scrolling transcript of the speech on the screen in front of them for any word substitutions. The proportion of word substitutions listeners are able to correctly identify is then used as a measure of speech intelligibility. When compared to speech reception thresholds (SRTs) given by a standard speech-in-noise test pilot results indicate higher SRTs for the GMUST. This suggests that the GMUST could be a more sensitive measure of deficits in speech-in-noise understanding than standard speech-in-noise tests.

2pPPb11. Magnitude of speech-reception-threshold manipulators for a spatial speech-in-speech test that takes signal-to-noise ratio confounds and ecological validity into account. Filip M. Rønne, Søren Laugesen, Niels S. Jensen, Renskje K. Hietkamp, and Julie H. Pedersen (Eriksholm Res. Ctr., Rørtangvej 20, Snekkersten 3070, Denmark, fmr@eriksholm.com)

Measuring speech-reception threshold (SRT) using adaptive procedures is popular, as testing yield results with desirable statistical properties. However, SRT measures have drawbacks related to the unbounded nature of the signal-to-noise

ratio (SNR) at which the SRT is achieved. Often the SRT will be a double-digit negative number, which compromises the ecological validity of the result. If testing involves hearing aids, it means that these devices and the signal-processing algorithms in them may be operating in conditions for which they were not intended. Further, the commonly observed large spread in SRT (both between- and within-group) has the possibility to cause SNR confounds that may lead to faulty conclusions. One way to address these issues is to provide the experimenter with SRT manipulators, to control the SNR at which testing takes place for the individual listener. The present work aims at developing a spatial speech-in-speech test with a selection of SRT manipulators for the experimenter to choose from. The manipulators investigated in this study are as follows: the spatial separation between target and maskers, the number of spatially separated maskers, changing the masker gender, and scoring in words versus sentences. The magnitudes of the SRT manipulators were investigated using 20 hearing-aid users as listeners.

2pPPb12. Binaural speech recognition in continuous and intermittent noises in people with hearing loss. Chantal Laroche, Jean-Grégoire Roveda, Julie Levionnois, Christian Giguère, and Véronique Vaillancourt (Audiology/SLP Program, Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H 8M5, Canada, claroche@uottawa.ca)

Multiple tests have been developed to quantify speech recognition in noise, but the characteristics of the masking noise vary significantly across tests and can considerably impact performance and clinical interpretation. Using the HINT, speech perception in 24 young adults with normal hearing was assessed using both the standard masker (a continuous speech-spectrum noise) and an intermittent version of the masker at an ON-OFF rate of 16 Hz. Intermittency helps in extracting speech from the "quiet" segments of the noise. Speech recognition thresholds for frontal speech were better in the intermittent than the continuous noise by an amount of 13 and 10 dB for noise maskers located in front or side, respectively, The average difference in thresholds between the noise front and side conditions, called the binaural advantage, was 4.7 and 8.4 dB for the intermittent and continuous noises, respectively. Data collected with people presenting different hearing loss profiles also show binaural and intermittency advantages, but to a lesser degree. Considering that people encounter a wide a range of fluctuating noises in daily life, these results motivate adding an intermittent noise condition to the HINT protocol to better reflect the challenges faced by individuals with hearing loss.

2pPPb13. Spatial release from masking for noise-vocoded speech. Jayaganesh Swaminathan, Christine R. Mason, Timothy M. Streeter (Boston University, 635 Commonwealth Avenue, Room 320, Boston, MA 02215, jswamy@bu.edu), Virginia Best (National Acoust. Lab., Chatswood, NSW, Australia), and Gerald Kidd, Jr (Boston Univ., Boston, MA)

Spatially separating a speech target from interfering masker(s) generally improves target intelligibility; an effect known as spatial release from masking (SRM). This study assessed the contribution of envelope cues to SRM. Target speech was presented from the front (0° azimuth) and speech maskers were either colocated or symmetrically separated from the target in azimuth (±15°, ±30°, ±45° and ±90°) using KEMAR head-related transfer functions. The target and maskers were presented either as natural speech or as noise-vocoded speech. For the vocoded speech, intelligibility was conveyed only by the envelopes from M frequency bands. Experiment 1 examined the effects of varying the number of frequency bands for the vocoder, and the degree of target-masker spatial separation, on SRM. Experiment 2 examined the effects of lowpass filtering the envelopes of the vocoded speech bands on SRM. Preliminary results for experiment 1 indicated that SRM improved as the number of spectral channels providing independent envelope cues increased for all spatial separations. Preliminary results for experiment 2 showed no difference in SRM between low and high envelope-frequency cutoffs. Potential implications for studying hearing-impaired and cochlear-implant subjects will be discussed. [Work supported by NIH-NIDCD and AFOSR.]

2pPPb14. Can envelope recovery account for speech recognition based on temporal fine structure? Frederic Apoux, Carla L. Youngdahl, Sarah E. Yoho, and Eric W. Healy (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, fred.apoux@gmail.com)

Over the past decade, several studies have demonstrated that normal-hearing listeners can achieve high levels of speech recognition when presented with only the temporal fine structure (TFS) of speech stimuli. Initial suggestions to explain these findings were that they were the result of the auditory system's ability to recover envelope information from the TFS (envelope recovery; ER). A number of studies have since showed decreasing ER with increasing numbers of analysis filters (the filters used to decompose the signal) while intelligibility from speech-TFS remains almost unaffected. Accordingly, it is now assumed that speech information is present in the TFS. A recent psychophysical study, however, showed that envelope information remains in the TFS after decomposition, suggesting a possible role of ER in speech-TFS understanding. The present study investigated this potential role. In contrast to previous work, a clear influence of analysis filter bandwidth on speech-TFS understanding was established. In addition, it was shown that near perfect speech recognition from recovered envelopes can be achieved with as many as 15 analysis filters. Finally, the relationship between analysis and auditory filter bandwidths was explored in ER. Taken together, the present findings suggest that a role of ER in speech-TFS understanding cannot be excluded.

2pPPb15. Complementary correlation may offer a new approach to better understand temporal fine structure coding. Adrian KC Lee (Speech & Hearing Sci. and Inst. for Learning & Brain Sci., Univ. of Washington, Box 357988, Seattle, WA 98195, akclee@uw.edu), Les E. Atlas, and Xing Li (Elec. Eng., Univ. of Washington, Seattle, WA)

Historically, sound has been separated into a slow-varying envelope and a component that rapidly varies, called the temporal fine structure (TFS). Perceptual studies suggest that users of cochlear implants can benefit from better TFS coding strategies, as they lead to improved speech understanding in noise. Yet given the standard signal processing methodology used in the auditory neuroscience community, namely the Hilbert transform, findings in such prior studies have been limited by their use of the heavily distorted estimates of the TFS information (through the Hilbert phase of the related analytic signal). Complementary correlation is a new mathematical tool with potential to advance our understanding of temporal coding in neuroscience. This new mathematical formulation can be defined simply by dropping the usual complex conjugation operation from the canonical definitions of correlation, coherence, variance, magnitude-squared estimators of power, and other similar common second-order statistical quantities and their estimators. We will show that a complementary correlation approach, which provides a more complete characterization of TFS information, will provide measurable perceptual benefits by reducing distortion in the original speech signal.

2pPPb16. The roles of temporal envelope and temporal fine structure in speech synthesis for cochlear implants for tonal language speakers. Nantaporn Saimai, Charturong Tantibundhit (Elec. and Comput. Eng., Thammasat Univ., 99 Moo 18 Phaholyothin Rd., Khlongluang, Pathumthani 12120, Thailand, 5310030076@student.tu.ac.th), Chutamanee Onsuwan (Linguistics, Thammasat Univ., Khlongluang, Pathumthani, Thailand), and Chai Wutiwiwatchai (National Electron. and Comput. Technol. Ctr. (NECTEC), Khlongluang, Pathumthani, Thailand)

Since most cochlear implants (CIs) have been developed for non-tonal languages, their level of hearing improvement is significantly decreased when used by speakers of tonal language, e.g., Thai. Temporal envelope (TE) and temporal fine structure (TFS) are important acoustic cues for languages with lexical tones. Specifically, TE is shown to carry manner and voicing cues for consonants, while TFS correlates with vowel formant transitions. Therefore, TFS and TE are expected to enhance intelligibility of lexical tones for CI patients. We proposed the use of six-channel bandpass filters to extract spectral information. Then, TE is extracted by half-wave rectification and smoothed by lowpass filler at 500 Hz cutoff frequency. TFS is extracted by the Hilbert transform to construct carrier signals. TE from each channel is modulated with its corresponding carrier signal and then combined to generate synthesized speech. Synthesized speech tokens from this study and two others [Fu *et al*. (1998) and Chen and Zhang (2008)] are evaluated by 16 Thais with normal hearing. The results showed that the intelligibility scores from the proposed algorithm are significantly higher than the other two for initials (by 32.2%) and final consonants (by 16.7%) and significantly higher for tones (by 48.8%) than Fu *et al*.

2pPPb17. The role of peripheral spectro-temporal coding in congenital amusia. Marion Cousineau (Département de Psychologie, Université de Montréal, Pavillon 1420 boul. Mont Royal, Entrance #1430 Ste. 0-120, Outremont, QC H2V 4P3, Canada, marioncousineau@gmail.com), Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN), and Isabelle Peretz (Département de Psychologie, Université de Montréal, Montreal, QC, Canada)

Congenital amusia, a neurogenetic disorder, affects primarily pitch and melody perception. Here we test the hypothesis that amusics suffer from impaired access to spectro-temporal fine-structure cues associated with low-order resolved harmonics. The hypothesis is motivated by the fact that tones containing only unresolved harmonics result in poorer pitch sensitivity in normal-hearing listeners. F0DLs were measured in amusics and matched controls for harmonic complexes containing either resolved or unresolved harmonics. Sensitivity to temporal-fine-structure was assessed via interaural-time-difference (ITD) thresholds, intensity resolution was probed via interaural-level-difference (ILD) thresholds and intensity difference limens, and spectral resolution was estimated using the notched-noise method. As expected, F0DLs were elevated in amusics for resolved harmonics; however, no difference between amusics and controls was found for F0DLs using unresolved harmonics. The deficit appears unlikely to be due to temporal-fine-structure coding, as ITD thresholds were unimpaired in the amusic group. In addition, no differences were found between the two groups in ILD thresholds, intensity difference limens, or auditory-filter bandwidths. Overall the results suggest a pitch-specific deficit in fine spectro-temporal information processing in amusia that cannot be ascribed to defective temporal-fine-structure or spectral encoding in the auditory periphery. [Work supported by Fyssen Foundation, Erasmus Mundus, CIHR, and NIH grant R01DC05216.]

2pPPb18. Comodulation masking release and monaural envelope correlation perception in listeners with cochlear hearing loss. Heather Porter, John H. Grose, Joseph W. Hall, and Emily Buss (Otolaryngol. - Head & Neck Surgery, Univ. of North Carolina at Chapel Hill, 170 Manning Dr., Chapel Hill, NC 27599-7070, heather_porter@med.unc.edu)

This study investigated the comparative dependence of comodulation masking release (CMR) and monaural envelope correlation perception (MECP) on the degree of envelope similarity in pre-senescent adult listeners with normal hearing (NH) or mild-to-moderate cochlear hearing loss (CL). A 1600-Hz pure-tone signal was used to measure CMR as a function of degree of envelope correlation in 100-Hz-wide noise bands centered at 727, 1093, 1600, 2300, and 3268 Hz. The same noise band configuration was used to measure MECP thresholds for both comodulated and independent standards. Envelope correlation was adjusted by mixing comodulated and independent maskers at variable intensity ratios. The five-band complex was 85 dB SPL. Signal thresholds improved monotonically (i.e., CMR increased) with increasing degrees of envelope correlation for all listeners. Results for CL listeners were most similar to data from previous NH listeners at a 72 dB SPL masker level. For MECP, performance patterns for the two conditions were uniform across NH listeners, whereas those for CL listeners exhibited greater individual differences. Finally, CMR and MECP performance appeared to be related in listeners with CL. The pattern of results will be discussed in terms of the effects of CL on sensitivity to envelope similarity. [Work supported by NIDCD R01DC001507.]

2pPPb19. Attentional switching when listeners respond to semantic meaning expressed by multiple talkers. Ervin R. Hafter (Dept. of Psych., Univ. of California, Berkeley, CA, hafter@berkeley.edu), Jing Xia, Sridhar Kalluri (Starkey Hearing Res. Ctr., Berkeley, CA), Rosa Poggesi, Claes Hansen, and Kelly Whiteford (Psychology, Univ. of California, Berkeley, Berkeley, CA)

The "cocktail party problem" asks how we know what one person says when others are speaking at the same time. With interest in the difficulty faced by older and hearing-impaired listeners in multi-talker environments, the present experiment looks at the speed of shifting attention between talkers. In our simulated cocktail party, a subject sits among multiple talkers who are each telling a different story. A sequence of questions are drawn from the various stories and presented visually for subjects to answer with manual responses. Pay is based on correct answers, and attention is assessed by comparing response accuracy on questions from the talker identified visually as the primary and from the other talkers. Attention is manipulated by varying the primary talker at random, from question to question. The speed of shifting attention is measured by varying the time from when the new primary is identified to the moment when the relevant information appears in that story. In a related study, bilingual subjects must shift attention to talkers speaking either English or Spanish. This allows determination of the additional time needed to switch languages within a multi-talker environment.

2pPPb20. Dividing attention between two segregated tone streams. Laurent Demany and Catherine Semal (INCIA, CNRS and Universite de Bordeaux, BP 63, 146 rue Leo Saignat, Bordeaux F-33076, France, laurent.demany@u-bordeaux2.fr)

Listeners were presented with pure-tone sequences which had a high speed (25–50 tones/s) and consisted of two interleaved melodies, M1 and M2, spanning separate frequency ranges (624–786 Hz and 1483–1869 Hz). M1 and M2 were renewed from sequence to sequence and could be either (1) perfectly correlated, or (2) perfectly anticorrelated, or (3) independent of each other. The main task, performed in a 2I-2AFC paradigm, was to discriminate sequences of type 1 (perfect correlation) from sequences of type 2 or 3. This appeared to be relatively easy when, in the type-1 sequences, each note of M2 immediately preceded or followed the corresponding note of M1. In that case, however, listeners were unable to tell whether M2 preceded or followed M1, which shows that the two melodies were perceptually segregated. In another series of experimental sessions, listeners knew that M2 would always follow M1 after a fixed delay, corresponding to one tone in some sessions and three tones in other sessions. The main discrimination task was performed better for the one-tone delay, but performance was still well above chance for the three-tone delay. Overall, the data suggest that two segregated tone streams can be attended to simultaneously.

2pPPb21. Selective auditory or visual attention reduces physiological noise in the ear canals of human subjects. Kyle P. Walsh, Edward G. Pasanen, and Dennis McFadden (Psychology, Univ. of Minnesota, 75 East River Rd., Minneapolis, Texas 55455, kpwalsh@umn.edu)

A nonlinear version of the stimulus-frequency OAE (SFOAE), called the nSFOAE, was used to measure cochlear responses from human subjects while they simultaneously performed behavioral tasks requiring selective auditory attention (dichotic or diotic listening), selective visual attention, or relatively little attention. The auditory- and visual-attention tasks both were digit-recall tasks, where the nSFOAE-stimuli were interleaved with seven spoken (or displayed) digits. Unlike many previous studies, the required motor behavior always was the same across all tasks, including the inattention tasks. A 30-ms recording in the quiet followed every nSFOAE-eliciting stimulus to provide an estimate of the magnitude of each subject's physiological noise in each experimental condition. For every subject, physiological-noise magnitudes were higher (noisier) in the inattention tasks, and lower (quieter) in the selective auditory- and visual-attention tasks. The differences in noise levels were about 3–6 dB, on average, and the effect sizes for those differences all were greater than 2.5. Our interpretation is that the efferent innervation of the cochlea is activated maximally during selective attention (be it auditory or visual), potentially to the benefit of the observer. [Work supported by NIDCD grant DC00153.]

2pPPb22. Build-up of auditory stream segregation induced by tone sequences of constant or alternating frequency and the resetting effects of single deviants. Nicholas R. Haywood and Brian Roberts (Psychology, School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk)

Three experiments investigated the dynamics of auditory stream segregation. Experiment 1 used a 2.0-s constant-frequency inducer (10 repetitions of a low-frequency pure tone) to promote segregation in a subsequent, 1.2-s test sequence of alternating low- and high-frequency tones. Replacing the final inducer tone with silence reduced reported test-sequence segregation substantially. This reduction did not occur when either the 4th or 7th inducer was replaced with silence. This suggests that a change at the induction/test-sequence boundary actively resets build-up, rather than less segregation occurring simply because fewer inducer tones were presented. Furthermore, experiment 2 found that a constant-frequency inducer produced its

maximum segregation-promoting effect after only three tone cycles—this contrasts with the more gradual build-up typically observed for alternating sequences. Experiment 3 required listeners to judge continuously the grouping of 20-s test sequences. Constant-frequency inducers were considerably more effective at promoting segregation than alternating ones; this difference persisted for ~10 s. In addition, resetting arising from a single deviant (longer tone) was associated only with constant-frequency inducers. Overall, the results suggest that constant-frequency inducers promote segregation by capturing one subset of test-sequence tones into an on-going, pre-established stream and that a deviant tone may reduce segregation by disrupting this capture.

2pPPb23. Annoyance perception for hearing impaired listeners: A revisit. Susie Valentine, Martin McKinney, and Tao Zhang (Starkey Hearing Technol., 6600 Washington Ave. S., Eden Prairie, MN 55344, susie_valentine@starkey.com)

For hearing impaired (HI) listeners, it is well known that certain sounds are much more annoying than others even though they may have similar spectral shape and level. For example, HI listeners often report paper rustling noise as highly annoying. A common approach to deal with this complaint is to reduce high frequency gain. While this approach may mitigate the complaint, it can create audibility issues for speech. A more effective approach is to determine the underlying cause of annoyance and then design an algorithm to selectively reduce it. While existing literature on annoyance perception for HI listeners is scant, a previous attempt was made to investigate this perception using real-world recordings [Vishnubhotla *et al.* (2012)]. The study showed a large variability of annoyance ratings across listeners that may have been due to subjective associations with the sound sources. In this study, we use abstract psychoacoustic stimuli designed carefully to avoid possible confounding subjective associations. A magnitude estimation method was used to measure the annoyance of each stimulus in hearing impaired listeners. All stimuli were presented over headphones in a sound treated room. Results will be presented along with implications for hearing aid applications.

2pPPb24. Acoustic correlates of tinnitus-like sounds. Jennifer Lentz and Yuan He (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jjlentz@indiana.edu)

Although many people describe their tinnitus using complex terms (such a tea-kettle, crickets, and roaring), past studies of tinnitus have focused using pure tones and noises as stimuli. Therefore, this study was developed to begin to address the usefulness of using complex, dynamic sounds in the assessment of tinnitus. In a previous study, a free-classification task was used to ascertain the perceptual dimensions of tinnitus-like sounds in normally hearing listeners. Sounds were representative of those commonly used to describe tinnitus (e.g., ringing, tonal, noisy, pulsing, and clicking sounds). Listeners placed icons associated with each sound on a grid and placed similar sounds in clusters. Multi-dimensional scaling conducted on the classification data revealed three different perceptual dimensions. This study evaluated the acoustics of the stimuli to determine the nature of the perceptual dimensions. These analyses estimated a variety of temporal and spectral stimulus properties (e.g., autocorrelation statistics, spectral statistics, envelope characteristics, etc.). The acoustic characteristics were then correlated with the ordering along the three perceptual dimensions. Results suggest a noisy versus tonal dimension, an envelope-based dimension stimulus (choppy versus smooth), and a dimension related to dynamic stimulus characteristics.

2pPPb25. Intervention for restricted dynamic range and reduced sound tolerance: Clinical trial using a Tinnitus Retraining Therapy protocol for hyperacusis. Craig Formby (Commun. Disord., Univ. of Alabama, 700 University Boulevard East, Tuscaloosa, AL 35487, cformby@as.ua.edu), Monica Hawley, LaGuinn P. Sherlock, Susan Gold (Otorhinolaryngology, Univ. of Maryland School of Medicine, Baltimore, MD), Jason Parton, Rebecca Brooks, and JoAnne Payne (Commun. Disord., Univ. of Alabama, Tuscaloosa, AL)

Hyperacusis is the intolerance to sound levels that normally are judged acceptable to others. The presence of hyperacusis (diagnosed or undiagnosed) can be an important reason that some persons reject their hearing aids. Tinnitus Retraining Therapy (TRT), a treatment approach for debilitating tinnitus and hyperacusis, routinely gives rise to increased loudness discomfort levels (LDLs) and improved sound tolerance. TRT involves both counseling and the daily exposure to soft sound from bilateral noise generator devices (NGs). We implemented a randomized, double-blind, placebo-controlled clinical trial to assess the efficacy of TRT as an intervention for reduced sound tolerance in hearing-aid eligible persons with hyperacusis and/or restricted dynamic ranges. Subjects were assigned to one of four treatment groups (2x2): Devices: NGs or placebo NGs and Counseling: Yes or No. They were evaluated at least monthly on a variety of audiometric tests, including LDLs, the Contour Test for Loudness for tones and speech, and word recognition measured at each session's comfortable and loud levels. Eighty percentage of the participants who received full treatment benefited significantly; whereas the other treatment groups demonstrated $\leq$ 45% treatment efficacy. Treatment dynamics and examples of improved word recognition post-treatment will be described. [Work supported by NIH R01 DC04678.]

2pPPb26. Relationship between distortion and working memory for digital noise-reduction processing in hearing aids. Kathryn Arehart (Speech, Lang., Hearing Sci., Univ. of Colorado, UCB 409, Boulder, CO 80309, kathryn.arehart@colorado.edu), Pamela Souza (Dept. of Commun. Sci. and Disord. and Knowles Hearing Ctr., Northwestern Univ., Evanston, IL), Thomas Lunner (Eriksholm Res. Ctr., Oticon A/S, Linköping, Sweden), Michael Syskind Pedersen (Oticon A/S, Smørum, Denmark), and James M. Kates (Speech, Lang., Hearing Sci., Univ. of Colorado, Boulder, CO)

Several recent studies have shown a relationship between working memory and the ability of older adults to benefit from specific advanced signal processing algorithms in hearing aids. In this study, we quantify tradeoffs between benefit due to noise reduction and the perceptual costs associated with distortion caused by the noise reduction algorithm. We also investigate the relationship between these tradeoffs and working memory abilities. Speech intelligibility, speech quality, and perceived listening effort were measured in a cohort of elderly adults with hearing loss. Test materials were low-context sentences presented in fluctuating noise conditions at several signal-to-noise ratios. Speech stimuli were processed with a binary mask noise-reduction strategy. The amount of distortion produced by the noise reduction algorithm was parametrically varied by manipulating two binary mask parameters, error rate, and attenuation rate. Working memory was assessed with a reading span test. Results will be discussed in terms of the extent to which intelligibility, quality, and effort ratings are explained by the amount of distortion and/or noise and by working memory ability. [Funded by NIH, Oticon, and GN ReSound.]

2pPPb27. Evaluation of a binaurally synchronized dynamic-range compression algorithms for hearing aids. Stephan Ernst, Giso Grimm, and Birger Kollmeier (Med. Phys., Univ. of Oldenburg, Carl von Ossietzky Universität Oldenburg, Oldenburg 26111, Germany, stephan.ernst2@uni-oldenburg.de)

Binaural cues such as interaural level differences (ILD) are used, among other cues, to organize auditory perception and to segregate sound sources in complex acoustical environments. Dynamic-range compression working independently at each ear in a bilateral hearing aid, however, can alter these ILDs, potentially affecting sound source segregation. Binaural synchronization of compression algorithms might thus be necessary to preserve potentially beneficial spatial cues. This study presents a binaurally linked model-based fast-acting dynamic compression algorithm designed to approximate the normal-hearing basilar membrane input-output function in hearing-impaired listeners. Aim of the evaluation was to assess the effect of binaural synchronization on speech intelligibility and listening effort in spatial masking conditions in comparison to bilateral fitting. Spatially symmetric and asymmetric masking conditions were used. A conventional multiband dynamic compression algorithm both implemented in a bilaterally independent and in a binaurally linked version, was tested as a reference. Hearing impaired listeners were aided individually with the algorithms for both experiments. Results indicate a small preference toward the model-based algorithm in challenging masking conditions. However, no benefit of binaural-synchronization could be found even for the fast-acting compressor, suggesting a dominant role of the better ear in all experimental conditions. [Work funded by BMBF 01EZ0741 and DFG FOR1732.]

2pPPb28. Assessing the contribution of spectral cues to recognition of frequency-lowered consonants. Kelly Fitz (Signal Process. Res., Starkey Hearing Technol., 6600 Washington Ave. South, Eden Prairie, MN 55344, kelly_fitz@starkey.com), Christophe Micheyl (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN), Susie Valentine (Audiology Res., Starkey Hearing Technol., Eden Prairie, MN), and Tao Zhang (Signal Process. Res., Starkey Hearing Technol., Eden Prairie, MN)

Commercially available strategies for restoring audibility of critical high frequency cues to patients with severe high frequency hearing loss translate information from high-frequency regions with unaidable hearing to lower-frequency regions with aidable hearing. Methods for synthesizing lowered spectral features, rather than generating them from the signal, have been proposed, though no commercially available hearing aid uses such a method. We assessed consonant discrimination under three configurations of a spectral feature synthesis method intended for use in frequency lowering. Lowered consonants were rendered using one or two narrowband noise components presented in a low frequency region with aidable hearing. Different configurations conveyed different spectral cues intended to distinguish among lowered consonants. In a short pilot study, preliminary analysis found no significant difference in consonant matching accuracy between the different configurations, suggesting that listeners were not making use of additional spectral cues when they were available. However, there were some observable trends in the data, as well as open questions about the possible impact of training and acclimatization on listener performance. We will present the findings of a more extensive study to determine whether listeners with training can make use of enhanced spectral cues to distinguish among frequency-lowered consonants.

2pPPb29. Spatial release from masking in simulations of cochlear implants for single-sided deafness. Joshua G. Bernstein (Audiology and Speech Ctr., Walter Reed National Military Med. Ctr., 8901 Wisconsin Ave., Bethesda, MD 20889, joshua.g.bernstein.civ@health.mil), Nandini Iyer (Battlespace Acoust. Branch, Air Force Res. Lab., Wright Patterson Air Force Base, OH), and Douglas S. Brungart (Audiology and Speech Ctr., Walter Reed National Military Med. Ctr., Bethesda, MD)

Previous studies have shown that single-sided-deaf (SSD) patients implanted with cochlear implants (CIs) can receive spatial release-from-masking (SRM), likely due to head-shadowing effects. This study investigated the possibility that SSD-CI patients might also obtain SRM benefits from binaural-integration cues when the target speech is masked by a spatially separated fluctuating masker. Experiment 1 measured psychometric functions for word-recognition performance in the presence of stationary noise, modulated noise, and one or two same- or opposite-gender interfering talkers. The first ear received an unprocessed mixture containing the target and masker(s). The second ear received no signal (SSD), an unprocessed mixture containing just the maskers (NH-Binaural), or a mixture containing just the maskers that was processed with an eight-channel noise vocoder (SSD+CI). The results show that SRM occurs in the NH-Binaural condition for all masker types, but that it only occurs in the SSD+CI condition with same-gender interfering talkers. Experiment 2 revealed that SRM occurred in the SSD+CI condition with as few as two vocoder channels, and that maximum performance occurred with six or more channels. These results suggest that CIs for SSD have the potential to produce SRM in situations where monaural cues are insufficient for concurrent speech-stream segregation.

2pPPb30. Speech intelligibility of hearing impaired participants in long-term training of bone-conducted ultrasonic hearing aid. Toshie Matsui, Ryota Shimokura, Tadashi Nishimura, Hiroshi Hosoi (Dept. of Otorhinolaryngol. - Head and Neck Surgery, Nara Med. Univ., Shijo-cho 840, Kashihara City 634-8522, Japan, tomatsui@naramed-u.ac.jp), and Seiji Nakagawa (Living Informatics Res. Group, National Inst. of Adv. Industrial Sci. and Technol., Ikeda City, Hyogo, Japan)

Bone-conducted ultrasonic hearing aid (BCUHA) system is the unique device to provide the auditory sensation to profoundly hearing-impaired persons without any surgical operations. To clarify effects of long-term hearing training with this device, two deaf participants engaged the BCUHA training for 9 months. They were trained to use BCUHA through repetition of sentences read aloud and free conversation, and then they took part in word recognition tests and monosyllabic identification tests. Both participants showed that they could recognize words above chance using auditory sensation only provided by BCUHA device if alternatives or context were presented to them during the trials. Besides, it was observed that monosyllabic intelligibility score with both of auditory and visual cue had much increased with the day of training than the score with auditory cue only and that with visual cue only. The result suggests that the long-term training with BCUHA achieves efficient integration of auditory and visual cue of speech such as cochlear implant users showed in previous studies.

2pPPb31. Development of a novel hearing-aid for the profoundly deaf using bone-conducted ultrasonic perception: Assessments of the modulation type with regard to articulation, intelligibility, and sound quality. Seiji Nakagawa, Chika Fujiyuki, Yuko Okubo, Takayuki Kagomiya, and Takuya Hotehama (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol. (AIST), 1-8-31 Midorigaoka, Ikeda, Osaka 563-8577, Japan, s-nakagawa@aist.go.jp)

Bone-conducted ultrasound (BCU) is perceived even by the profoundly sensorineural deaf. We have developed a novel hearing-aid using BCU perception (BCU hearing aid: BCUHA) for the profoundly deaf. In the BCUHA, ultrasonic sinusoids of about 30 kHz are amplitude-modulated by speech and presented to the mastoid. Generally, two sounds are perceived: one is a high-pitched tone due to the ultrasonic carrier, with a pitch corresponding to a 8–16 kHz air-conducted (AC) sinusoid, and the other is the envelope of the modulated signal. As a method of amplitude modulation (AM), double-sideband with transmitted carrier (DSB-TC) modulation had been used; however, the DSB-TC is accompanied by a strong high-pitched tone. In this study, two new AM methods, double-sideband with suppressed carrier (DSB-SC) and transposed modulation (TM), that can be expected to reduce the high-pitched tone were newly employed in the BCUHA, and their resulting articulations, intelligibilities, and sound qualities were evaluated. The results showed that DSB-TC and TM had higher articulation and intelligibility scores than DSB-SC. Further, in terms of sound quality, the TM speech was closer than other types of BCU speech to AC speech. These results provide useful information for further development of the BCUHA.

2pPPb32. Communication aid utilizing bone-conducted sound via teeth by means of mouthpiece form actuator. Mikio Muramatsu (Dept. of Intermedia Art and Sci., Waseda Univ., Tokyo, Japan), Junko Kurosawa (Information Technol. Res. Organization, Waseda Univ., Tokyo, Japan), Yasuhiro Oikawa, and Yoshio Yamasaki (Dept. of Intermedia Art and Sci., Waseda Univ., 3-4-1, Okubo, Shinjuku-ku, Tokyo, Japan, yoikawa@waseda.jp)

Since bone-conducted sound is conveyed to cochlea directly, without passing through eardrum, it is audible even for hard-of-hearing people whose inner ears are still normal. In this study, we utilize bone-conducted sound via teeth so as to support sound communication. We implement a bone-conducted actuator on a tooth, while actuators of prevalent hearing aids are attached to mastoid, forehead, or jaw in general. Teeth convey sound excitation more easily, because they are bare bones, not covered with skin. Our hearing aid is made in the form of mouthpiece, and it can be readily put on and taken off from a tooth. Plus, we carry out experiments regarding sound localization and thresholds of bone-conducted sound via teeth, using this actuator. The results offer hearing characteristic of bone-conducted sound via teeth and show that examinees can perceive right and left using bone-conducted sound via teeth. In addition, we also attempt to record vibrations of teeth through a microphone, which is embedded on the mouthpiece form actuator. The aim of this study is to realize a hybrid actuator that enables both hearing and recording simultaneously and to suggest a new communication aid system not only for hard-of-hearing people but also for the robust.

Session 2pSA

Structural Acoustics and Vibration: Memorial Session in Honor of Miguel Junger

David Feit, Cochair
ASA, 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502

Joel M. Garrelick, Cochair
505 Tremont St., Apt 209, Boston, MA 02116

Chair's Introduction—1:35

Invited Papers

1:40

2pSA1. Recollections of my father: Miguel C. Junger. Sebastian Junger (P.O. Box 906, Truro, MA 02666, sjunger5@aol.com)

Thoughts and reflections on my father as such an inspirational and unforgettable person.

2:00

2pSA2. Miguel Junger: Legacy contributions to the field of structural acoustics. David Feit (ASA, 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502, feit.d@att.net)

The field of structural acoustics, i.e., the theory of acoustic radiation and scattering from elastic structures, developed primarily in the last half of the 20th century in response to Navy needs. Miguel Junger and colleagues at Cambridge Acoustical Associates under the sponsorship of the Office of Naval Research Sound and Structures made seminal contributions to the field. This presentation reviews some of this research, which ultimately was included in the book, "Sound, Structures and Their Interaction."

2:20

2pSA3. Three curious results of radiation loading calculations. Joel M. Garrelick (505 Tremont St., Unit 209, Boston, MA 02116, joelgarrelick@aol.com)

One of the many joys in reading Miguel C. Junger's contributions to the journal and elsewhere was the physical interpretations that accompanied theoretical results, which were at times unanticipated but always enlightening. In three instances however, he found the task of offering such an explanation particularly challenging and was beguiled by the fact. This is documented in his paper "Three apparently paradoxical results of sound radiation theory" [J. Acoust. Soc. Am. **106**(3, Pt. 1), 1589–1590 (1999)]. The specific results are as follows: (1) At frequencies above coincidence, the power radiated by a point driven infinite thin plate exposed to a low impedance fluid is equal to the power radiated by that plate when submerged in a high impedance fluid. (2) At low frequencies, the entrained mass that is associated with a translating circular piston fully submerged in an acoustic medium is equal to that acting on the piston when it is baffled and exposed to the medium on one side only. (3) The low frequency admittance of a fluid loaded infinite thin plate driven by a point force exhibits a spring-like reactance and yields a phase angle magnitude that is equal to that of the plate when driven by a line force. These three topics are revisited in this paper with an attempt to distinguish between paradox and coincidence.

2:40

2pSA4. Acoustic waves in violently collapsing bubbles. Thomas L. Geers (Mech. Eng., Univ. of Colorado, Campus Box 427, Boulder, CO 80302, geers@spot.colorado.edu)

Among Miguel Junger's many contributions to acoustical science and engineering were his papers and presentations on bubble acoustics. Among his many contributions to the well being of his colleagues at Cambridge Acoustical Associates was the mentoring of this presenter during the latter years of graduate study at MIT. Hence, this presentation in this session. In an evaluation of five reduced models for spherically symmetric bubble collapse and rebound [J. Appl. Phys. **112**, 054910 (2012)], it was found that some recent models, which incorporate wave propagation in both the external fluid and internal gas, did not perform as well as the long-established model by Keller and Kolodner, which incorporates wave propagation in the fluid but not in the gas [J. Appl. Phys. **27**, 1152–1161 (1956)]. Performance was assessed through comparisons against response histories produced by finite-difference solution of the Euler equations under adiabatic conditions. Further investigation revealed that neither acoustic-wave nor shock-wave propagation in the gas was apparent, but that a standing wave in the gas was. This prompted an enhancement of the Keller and Kolodner model that accounts for the standing wave. The formulation and evaluation of the enhanced model is the subject of this presentation.

3:00–3:20 Break

3:20

2pSA5. Poisson coupling in the in vacuo dynamics of an infinite cylindrical shell. Rudolph Martinez (Acoustics, Appl. Physical Sci., 49 Waltham St., Lexington, MA 02421, rmartinez@aphysci.com)

This paper applies a perturbation analysis to the "n = 0" in vacuo dynamics of an infinite cylindrical shell as presented in their exact form by Junger and Feit in *Sound, Structures, and Their Interaction*. The small parameter in the asymptotic theory is the square of the Poisson ratio ν. Two sets of results emerge: (1) Relating to the system's dispersion relation and therefore irrespective of the type of loading, the approximate roots of the governing cubic polynomial display explicitly the mutual contamination and influence of axial compression and latent skin flexure, the latter becoming actual past the ring frequency ωra/cp = 1. (2) In the parametric range of small values of ωa/cp, and for a radial ring force, our asymptotic expansions establish that the shell's normal-to-surface response is one of local reaction to zeroth order in ν2. It is not until the analysis is carried out to O(ν2) that wave propagation from axial compression and nearfield skin flexure begin to assert themselves away from the driven station.

3:40

2pSA6. Sound radiation by parallel coated plates separated by a fluid layer: Now and Then. Ann Stokes (Appl. Physical Sci., 49 Waltham St., Lexington, MA 02421, astokes@aphysci.com)

In 1987, I had the opportunity to work with Miguel Junger on a paper entitled "Sound Radiation by parallel coated plates separated by a fluid layer." A critical piece of the analysis involved analytically inverting a 5x5 matrix, which resulted in an extremely complex algebraic expression for the far-field of a point-excited 5-layer configuration. In order to understand the physical mechanisms, and validate the results, Dr. Junger developed asymptotic forms of the solution for familiar configurations and made use of spring-mass models to understand resonance enhancement effects. This talk introduces our current semi-analytical waveguide analysis of sound radiation by parallel, multi-layer plates. An advantage of the waveguide approach is that finite-length structures can be modeled, and axial discontinuities, including ribs and wavebearing bulkheads, can also be included very efficiently. Physical mechanisms are interpreted in terms of the propagating and evanescent waves of the structures. A focus of this talk is the extension of the perfectly matched layer (PML) model of the exterior fluid to waveguide models.

4:00

2pSA7. Cooperating with Miguel on improvements of the acoustical product - SOUNDBLOX. Klaus Kleinschmidt (Retired, 100 Newbury Court, Concord, MA 01742, ksquare@comcast.net)

Some 15 years after joining Miguel's consulting firm Cambridge Acoustical Associates around 1960 he asked me to help him with improving the sound absorbing quality of a patented slotted concrete block. The original concept of the block, sold under the trademark, SOUNDBLOX, was create a Helmholtz resonator by providing a slot in one face of a standard concrete block whose natural frequency would match the fundamental frequency of a common noise source, transformer hum. A broader absorption spectrum was desired to expand the rather limited applications of the original design to compete with certain acoustical materials widely used in schools, gymnasiums, auditoriums, and swimming pools. This presentation will describe the nature of our cooperation and the successes and failures of various concepts. The fact that the product, first introduced in the late 1950's, is still on the market speaks well of our collaboration.

Contributed Papers

4:20

2pSA8. Sound radiation from finite surfaces. Jonas Brunskog (Acoust. Technol., DTU Elec. Eng., Elektrovej, Bldg. 352, Kgs. Lyngby DK-2800, Denmark, jbr@elektro.dtu.dk)

A method to account for the effect of finite size in acoustic power radiation problem of planar surfaces using spatial windowing is developed. Cremer and Heckl presents a very useful formula for the power radiating from a structure using the spatially Fourier transformed velocity, which combined with spatially windowing of a plane waves can be used to take into account the finite size. In the present paper, this is developed by means of a radiation impedance for finite surfaces, which is used instead of the radiation impedance for infinite surfaces. In this way, the spatial windowing is included in the radiation formula directly, and no pre-windowing is needed. Examples are given for the radiation efficiency, and the results are compared with results found in the literature.

4:40

2pSA9. Modal contributions to sound radiated from a fluid loaded cylinder. Herwig Peters, Nicole Kessissoglou (School of Mech. and Manufacturing Eng., The Univ. of New South Wales, UNSW, Sydney NSW 2052 Australia, z3268667@student.unsw.edu.au), and Steffen Marburg (LRT4 - Inst. of Mech., Universitat der Bundeswehr Munchen, Neubiberg, Germany)

A modal decomposition technique to compute the individual modal contributions to the sound radiated from a cylindrical shell submerged in water is presented. The wet structural modes are calculated by means of a polynomial approximation and symmetric linearization of the underlying nonlinear eigenvalue problem. A Krylov subspace technique is used to reduce the model size of the structural domain, while the fluid domain remains unchanged. Results for the radiated sound power and sound pressure directivity are presented for groups of circumferential modes with common mode number. Under axial and transverse excitation, the cylinder breathing and bending modes are respectively the major modes contributing to the radiated sound at low frequencies. The contribution of the rigid body modes to the radiated sound is also observed.

5:00

2pSA10. Acoustic radiation mode shapes for control of plates and shells. William R. Johnson (Mech. Eng., Brigham Young Univ., 1085 N. 1750 W., Provo, UT 84604, will.johnson@byu.edu), Pegah Aslani (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Daniel R. Hendricks (Mech. Eng., Brigham Young Univ., Provo, UT)

During the advent of active structural acoustic control, attempts were made to target and control structural vibration mode shapes to reduce radiated sound power. In the late 1980s and early nineties work on acoustic radiation mode shapes developed an alternative way to target structural acoustic radiation. By attempting to control the radiation mode shapes, contributing structural modes could be more easily targeted. Radiation mode shapes have been examined previously for rectangular plates. The method has been extended to demonstrate radiation mode shapes of circular plates and cylindrical shells. Certain spatial derivatives of plate vibration have been found to be highly correlated with the most efficiently radiating radiation mode shapes at low frequencies. A weighted sum of these spatial derivatives is proposed as a new, generalized control metric.

Session 2pSCa

Speech Communication: Variability in Speech Intelligibility: Behavioral and Neural Perspectives

Rajka Smiljanic, Cochair
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Bharath Chandrasekaran, Cochair
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Sven Mattys, Cochair
Dept. of Psych., Univ. of York, York YO10 5DD, United Kingdom

Chair's Introduction—12:55

Invited Papers

1:00

2pSCa1. Processing speech of varying intelligibility. Rajka Smiljanic (Linguistics, Univ. of Texas-Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu) and Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas-Austin, Austin, TX)

In this talk, we examine how variation in intelligibility impacts speech processing with insights from behavioral and neuroimaging studies. We discuss a set of experiments that explore the extent to which listener-oriented speaking style changes, sentence context, and visual information contribute to enhanced word recognition in challenging listening conditions. We further examine whether these same enhancements impact speech processing beyond word recognition, namely recognition memory for sentences. The results show that both signal-related and contextual enhancements lead to improved speech recognition in noise and, crucially, to a substantially better sentence recall. We then discuss studies examining neural mechanisms involved in processing speech of varying intelligibility using fMRI. Previous fMRI studies have examined speech intelligibility by using artificially degraded speech stimuli. Few studies have examined natural variation in intelligibility. Here we present neuroimaging data from two studies that examine natural variations in speech intelligibility (native vs. non-native speech; audio versus audiovisual speech). Overall, combined insights from behavioral and neuroimaging studies provide important additions to our understanding of how different sources of variability in the speech signal affect speech processing and memory representations.

1:20

2pSCa2. The impact of variation in phoneme category structure on consonant intelligibility. Valerie Hazan, Rachel R. Romeo, and Michele Pettinato (Speech, Hearing and Phonet. Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, v.hazan@ucl.ac.uk)

Newman *et al*. [J. Acoustic. Soc. Am. **109**, 1181–1196 (2001)] suggested that phoneme identification accuracy and speed for a given talker was affected by the degree of variability in their production of phoneme categories. This study investigates how intra-talker variability in the production of two phoneme contrasts varies with age and gender, and how this variability affects speed of perceptual processing. Multiple iterations of tokens differing in initial consonants (/s/-/ʃ/, /p/-/b/) were collected via picture elicitation from 40 adults and 31 children aged 11 to 14; measures of within-category dispersion, between-category distance, overlap, and discriminability were obtained. While females produced more discriminable categories than males, children produced farther yet more dispersed—and thus similarly discriminable—categories than adults. Variability was contrast-specific rather than a general talker characteristic. Tokens with initial /s/-/ʃ/ from pairs of adult and child talkers varying in between-category distance or overlap were presented for identification. The presence of overlap had a greater effect on identification accuracy and speed than between-category distance, with strongest effects for adult speakers, but reaction time correlated most highly with within-category dispersion. These data suggest that talkers who are less consistent in their speech production may be perceived less clearly than more internally consistent talkers.

Contributed Paper

1:40

2pSCa3. On the tolerance of spectral blur in the perception of spoken words. Robert E. Remez, Chloe B. Cheimets, and Emily F. Thomas (Dept. of Psych. and Prog. in Neurosci. & Behavior, Barnard College, Columbia Univ., 3009 Broadway, New York, NY 10027, remez@columbia.edu)

How does a listener resolve linguistic properties conveyed by speech? Many descriptions of perception attribute a causal role to brief spectral details in narrow frequency ranges. Perceptual standards allow far more variety, revealed by the robustness of perception of many kinds of distorted speech. The present study considered the effects of spectral blur on the recognition of spoken words. Listeners heard successive presentations of noise-vocoded easy and hard words. The number of spectral channels composing the word increased with each presentation, reducing blur within a trial. Four conditions counterbalanced the number of presentations of each word in a trial, 3 or 5, and the severity of initial blur, either 1 or 5 channels. In all conditions, the final presentation had 9 bands, yielding a net blur reduction of 4 or 8 bands. These conditions were also tested with the instruction that the

words would become clearer during each trial. A control used two repetitions of each word at 9 spectral bands. Across the tests, exposure to spectral blur impaired the recognition of easy and hard words alike regardless of the listener's belief during presentation. However, intelligibility of hard words declined sharply when subjects were instructed to attend to the continuity and successive decrease in blur within a trial. The pattern of results exposes the role of attention, uncertainty, and spectral resolution in the phonetic contribution to word identification.

Invited Papers

2:00

2pSCa4. Intelligibility of interrupted speech in listeners of different age and hearing status. Valeriy Shafiro, Stanley Sheft, Robert Risley (Commun. Disord. & Sci., Rush Univ. Med. Ctr., 600 S. Paulina Str., AAC 1012, Chicago, IL 60612, valeriy_shafiro@rush.edu), and Brian Gygi (NIHR Nottingham Hearing Biomed. Res. Unit, Nottingham, United Kingdom)

In most real-world communicative environments, speech signals are fragmented and incomplete due to masking by other concurrent sounds. Experiments in the perception of gated and time-compressed speech provide a useful approach for systematically investigating factors involved in the perception of temporally fragmented speech. In the current work, the intelligibility of spoken sentences was measured following periodic signal interruption at various square-wave gating rates with time compression applied by either omitting or doubling silent intervals during gated-off times. Across interruption rates (0.5–16 Hz), speech perception of younger and older normal-hearing and older hearing-impaired adults was similar for slow interruption rates, but differed substantially for the faster rates. This rate-dependent variation was consistently found for different interruption methods and conditions. Importantly, speech perception at fast interruption rates correlated strongly with pure-tone thresholds and performance on speech-in-noise tests, while it did not correlate with results from tests of working memory or spectro-temporal pattern discrimination. These findings suggest that different perceptual processes are involved in the perception of interrupted speech at slow and at faster interruption rates, and have implications for developing diagnostic tests applicable to real-world listening environments. [Work supported by NIH.]

2:20–2:40 Break

2:40

2pSCa5. Impaired speech recognition under a cognitive load: Where is the locus? Sven Mattys (Dept. of Psych., Univ. of York, York YO10 5DD, United Kingdom, sven.mattys@york.ac.uk)

Improving the validity of speech-recognition models requires an understanding of the conditions in which speech is experienced in everyday life. Listening conditions leading to a degradation of the signal—noise, competing talkers, disordered speech—have received most of the attention in that literature. But what about adverse conditions that do not alter the integrity of the signal, such as listening to speech under a non-auditory cognitive load (CL)? Drawing upon a variety of behavioral methods, this presentation investigates the effects of a concurrent attentional or mnemonic task on the relative reliance on acoustic cues and lexical knowledge during speech-perception tasks. The results show that listeners under CL downplay the contribution of acoustic detail and increase their reliance on lexical-semantic knowledge. However, greater reliance on lexical-semantic knowledge under CL is a cascaded effect of impoverished phonetic processing, not a direct consequence of CL. Ways of integrating CL into the functional architecture of existing speech-recognition models are discussed.

Contributed Paper

3:00

2pSCa6. Development of the auditory evoked potential to amplitude rise time and rate of formant transition of speech sounds. Antoine Shahin and Allen Carpenter (Otolaryngology, The Ohio State Univ., 915 Olentangy River Rd., Columbus, OH 43212, tonyshahin@gmail.com)

We investigated the morphology of the N1-P2 auditory evoked potential (AEP) to changes in amplitude rise time (ART) and rate of formant transition (RFT) of consonant-vowel pairs (CVs) in 4–6-year olds and adults. In the AEP session, individuals listened passively to the CVs /ba/, /wa/, and a /ba/ with a superimposed slower-rising /wa/ envelope (/ba/wa). In the behavioral session, individuals listened to the same stimuli and judged whether they heard a /ba/ or /wa/. In 6-year olds and adults, the N1-P2 amplitude reflected a change in RFT (/ba/wa and /wa/) but not in ART (/ba/ and /ba/wa). In contrast, in the 4–5-year olds, the poorly developed N1-P2 did not show specificity to changes in RFT or ART. Behaviorally, 6-year olds and adults relied more strongly on RFT cues (classified /ba/wa as /ba/) during phonetic judgments, as opposed to 4–5-year olds, which utilized both cues equally (chance level). Our findings suggest that following age 4–5, the development of the N1-P2 AEP, representing maturation of synaptic connections in the superficial layer of the auditory cortex, reflects a shift toward weighting of spectral dynamics more than amplitude dynamics during /ba/ - /wa/ phonetic categorization.

Invited Paper

3:20

2pSCa7. Cortical responses to degraded speech are modulated by linguistic predictions. Jonathan E. Peelle (Dept. of Otolaryngol., Washington Univ. in St. Louis, 660 South Euclid, Box 8115, St. Louis, MO 63110, peellej@ent.wustl.edu)

Our perceptual experience is formed by combining incoming sensory information with prior knowledge and expectation. When speech is not fully intelligible, non-acoustic information may be particularly important. Predictions about this degraded acoustic signal can be provided intrinsically (if the speech is still partially intelligible) or extrinsically (for example, by presenting a written cue). I will discuss results from studies in which the neural response to speech was measured using magnetoencephalography (MEG), with speech clarity parametrically manipulated using noise vocoding. In one study, we found that during sentence processing the phase of ongoing

cortical oscillations is matched to that of the acoustic speech envelope in the range of the syllable rate (4–8 Hz). Critically, this phase-locking was enhanced in left temporal cortex when speech is intelligible. In a separate study of single word listening, accurate predictions provided by written text enhanced subjective clarity and changed the response in early auditory processing regions of temporal cortex. Both experiments thus highlight neural responses in brain regions associated with relatively low-level speech perception. Together these findings support the ability of linguistic information to provide predictions that shape auditory processing of spoken language, particularly when acoustic clarity is compromised.

3:40–4:20 Panel Discussion

TUESDAY AFTERNOON, 4 JUNE 2013 516, 2:20 P.M. TO 5:00 P.M.

Session 2pSCb

Speech Communication: Speech Intelligibility (Poster Session)

Yingjiu Nie, Chair
Speech Lang. Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr., SE, Minneapolis, MN 55455

Contributed Papers

All posters will be on display from 2:20 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:20 p.m. to 3:40 p.m. and contributors of even-numbered papers will be at their posters from 3:40 p.m. to 5:00 p.m.

2pSCb1. Effect of speech clarity on perception of interrupted meaningful and anomalous sentences. Rajka Smiljanic (Linguistics, Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu), Stan Shaft (Commun. Disord. and Sci., Rush Univ. Med. Ctr., Chicago, IL), Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas-Austin, Austin, TX), and Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Med. Ctr., Chicago, IL)

The influence of speech clarity on the perception of interrupted speech was examined for sentences distinguished by the presence of semantic-contextual cues. Semantically meaningful and anomalous sentences produced in either conversational or “clear” speech were periodically interrupted at gating rates ranging from slow (0.5 Hz) to fast (24 Hz) and presented to 32 native English listeners. At slow rates, speech perception may be based on integration of whole syllables and words, with “glimpsing” of (sub)phonemic segments playing a role at faster rates. Our results show that semantic context and speech clarity had a significant rate-dependent impact on the intelligibility of interrupted speech. At the lowest rates, intelligibility differences between conditions were minimal. Overall, interruption was most deleterious for anomalous conversational sentences. Such effects were seen even at the highest gating rate of 24 Hz for which interruption effects are generally minimal. The magnitude of the clear-speech benefit varied with gating rate for the two types of sentences, starting at 1 Hz for meaningful and 2 Hz for anomalous sentences. Acoustic-phonetic enhancements of clear speech thus “shifted” contextual benefit to lower gating rates. The implications of these results for our understanding of different time scales of speech processing will be discussed.

2pSCb2. Speech understanding in babble noise and syllabic-range temporal processing in aging. Pierre Divenyi (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., 660 Lomita Ct., Stanford, CA 94305, pdivenyi@ccrma.stanford.edu)

The ability to understand speech in multitalker noise was measured in a group of 46 elderly (71.6 +/− 6.03 years) having at worst moderate presbycusic hearing loss, using the SPIN test with the target and an 8-talker babble presented either monaurally (left or right) or, using generic HRTF spatialization, with the target at 45 degrees right or left flanked by two independent 4-talker babble on each side. The subjects’ temporal processing was measured as (1) the reverberation time yielding a 15% decrease in word comprehension and (2) as the minimum modulation index necessary to segregate two streams generated by modulating, at an average frequency of 4.375 Hz, two carriers consisting of harmonics 4, 5, and 6 having fundamental frequencies of 107 and 189 Hz, that produced two simultaneous three-pulse envelopes, one accelerating and one decelerating. With effects of hearing loss under control, the data exhibited correlations between intelligibility in babble and temporal processing that were significant for a speech-to-babble ratio of 4 dB but not for one of 8 dB. The results thus suggest that under unfavorable acoustic conditions the elderly achieve speech understanding by focusing on prosodic rhythm on the syllabic level. [Work supported by NIA and the VA Medical Research.]

2pSCb3. The effects of temporal envelope confusion on listeners’ phoneme and word recognition. Yingjiu Nie, Adam Svec, Peggy B. Nelson, and Benjamin Munson (Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, niex0008@umn.edu)

Broadened auditory filters in listeners with hearing loss may result in listeners’ increased reliance on temporal envelope cues for understanding speech. Previous data have shown that background noise may affect hearing-impaired (HI) listeners by negatively affecting the temporal envelope cues in speech. The current study investigates additional HI listeners’ understanding of vocoded spondees in the presence of fluctuating and stationary background noise. Stimuli were 8- and 32-channel noise vocoded double spondees, high-pass filtered at 1426 Hz. New data confirmed the previous finding that temporal envelope confusion in HI listeners resulted in speech understanding that is poorer in fluctuating noise (at a rate of 4 Hz) than in stationary noise. Preliminary analysis suggests HI listeners experience significant envelope confusion for both 8- and 32-channel vocoded stimuli. Additional analysis of phoneme errors suggests that envelope confusion affects HI listeners’ perception of both consonants and vowels. Further analysis of j-factors will indicate the relationship of phoneme to whole word understanding for vocoded speech in noise. Results confirm the importance of temporal envelope cues for phoneme and syllable recognition for listeners with hearing loss. [Work supported by NIH DC008306 to PB Nelson.]

2pSCb4. Quality of voices processed by hearing aids: Intra-talker differences. Ramesh Kumar Muralimanohar, Caleb Kronen, Kathryn Arehart, James Kates (Dept. of Speech, Lang., and Hearing Sci., UCB 409, Univ. of Colorado at Boulder, Boulder, CO 80309, muralima@colorado.edu), and Kathleen Pichora-Fuller (Dept. of Psych., Univ. of Toronto, Toronto, ON, Canada)

The effects of hearing aid signal processing depend on the voice characteristics of a talker. For example, we have found that the perceptual and acoustic consequences of hearing aid signal processing vary across talkers and that these effects can be explained, in part, by the acoustic differences between the different voices. However, we find that different utterances spoken by the same talker are also differentially affected by the hearing aid signal processing. In this study, we quantified the acoustic and perceptual consequences of hearing aid signal processing on several utterances spoken by the same talker. The voices were presented in quiet and in the presence of babble noise (10 dB SNR). The voices were processed with varying amounts of frequency compression, wide dynamic range compression and noise suppression (spectral subtraction). We considered intra-talker variability using an objective quality metric (Hearing Aid Sound Quality Index) and perceptual ratings of quality. We analyzed sentences from several different speech corpora, including the Hearing-in-Noise Test (HINT), IEEE, and TIMIT. The results showed interactions between the effects of signal processing and the acoustic characteristics of specific utterances spoken by an individual. [Work supported in part by GN ReSound and NSERC.]

2pSCb5. Perceptual confusability of French vowels. Kathleen C. Hall (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, kathleen.hall@ubc.ca) and Elizabeth Hume (Linguistics, Univ. of Canterbury, Christchurch, New Zealand)

The confusability of sounds is argued to both reflect phonological structure [e.g., Boomershine *et al*. (2008)] and be a source of phonological variability and change [e.g., Ohala (1981), Hume (1998)]. We present the results of a perception task in which 25 Parisian French-speaking participants identified the French vowels [i e ɛ y ø œ ə a u o ɔ ɔ̃ ɛ̃ ɑ̃], or ∅, in an aC_Ca context, using standard orthography in key words. The durations of the vowel in the vowel-containing stimuli were manipulated to be 0, 15, 25, 50, or 100% of their original durations. We can therefore determine which vowels are most confusable with each other (and thus likely to be the target for either mergers or dissimilatory processes) and which are most confusable with zero (and thus likely to be the target of processes such as deletion, assimilation, and metathesis). Results show high accuracy for [a i y u], even with very short durations; some degree of confusability within the nasal vowels; high confusability rates within the mid-front rounded vowels; and a tendency for zero to be confused with one of the mid-front rounded vowels. These results align with observed phonological patterns in French.

2pSCb6. Effects of phonetic experience on neural processing of English /r/ and /l/ by Korean and Japanese listeners. Lee Jung An, Brett A. Martin, and Glenis R. Long (Ph.D. Prog. in Speech-Lang.-Hearing Sci., The Grad. Ctr., CUNY, 365 Fifth Ave., 7th Fl., New York, NY 10016, lan@gc.cuny.edu)

The effects of phonetic experience on behavioral and neurophysiological processing of English /r/ and /l/ by Koreans and Japanese were compared to speakers of American English. Although English /r/ and /l/ are not phonemic in both Korean and Japanese languages, Koreans have a pseudo phonetic [ɾ]-[ll] model available for perception of English, /r/-/l/ sounds in medial position, while Japanese do not. Speech stimuli were a continuum of synthetic stimuli ranging from perceived /iri/ to perceived /ili/. To date, five subjects in each language group have been tested. As predicted, behavioral results show that English medial /r/ and /l/ were perceived in a categorical manner by Americans, in a categorical-like manner by Koreans and in a non-categorical manner by Japanese. Neural responses tapped by the ACC did not differ significantly between language groups for P1-N1-P2, suggesting little effect of phonetic experience on the encoding of these sounds. In contrast, the T-complex (Tb latency and Ta morphology) differed significantly between groups. The T-complex morphology had double-peaks in the Japanese group. These findings suggest that the T-complex may index the effects of phonetic experience on speech perception.

2pSCb7. Perception of Thai distinctive vowel length in noise. Chutamanee Onsuwan (Linguistics, Thammasat Univ., 2 Prachan Rd., Pranakhon, Bangkok, Bangkok 10200, Thailand, consuwan@tu.ac.th), Charturong Tantibundhit, Nantaporn Saimai, Tanawan Saimai (Elec. and Comput. Eng., Thammasat Univ., Khlongluang, Pathumthani, Thailand), Patcharika Chootrakool, and Sumonmas Thatphithakkul (National Electron. and Comput. Technol. Ctr. (NECTEC), Khlongluang, Pathumthani, Thailand)

A forced choice identification perception experiment using 150 monosyllabic rhyming-word stimulus pairs (with identical consonants and tone) in four conditions of white Gaussian noise was conducted to explore vowel confusions in Thai, a language with nine monophthongs and length (short-long) contrast for all vowels (e.g., /i/-/iː/ and /o/-/oː/). Each stimulus containing speech and noise portions is equal in length. Perceptual results of 18 vowels from 36 Thai listeners at a noise level (SNR) of -24 dB, where the percent intelligibility is the most interpretable, showed that stimuli with short vowels are more accurately perceived than those with long vowels (93.46 vs. 85.64%) with /oː/ and /eː/ as the most confusable. Interestingly, asymmetrical confusions are observed with very few short vowels being misperceived as long vowels, but a larger number of long vowels misperceived as short. Consistent with previous studies of perception of English vowels in white noise [e.g., Benki (2003)], the findings confirm perceptual robustness of vowel height (correlating with F1) over vowel front/backness (correlating with F2). Lastly, an analysis for listeners' misidentified responses shows that the listeners generally favor short over long vowels.

2pSCb8. Effects of semantic predictability and dialect variation on vowel production in clear and plain lab speech. Rory Turnbull and Cynthia G. Clopper (Linguistics, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43215, turnbull@ling.osu.edu)

Speech addressed to a non-native or hearing impaired listener features longer, more peripheral vowels. In addition, more extreme dialect-specific forms are produced in semantically predictable contexts, and less extreme forms (more standard forms) in unpredictable contexts. This study investigated the interactions between predictability and speaking style on Southern American English monophthongization of the vowel /aj/. The Midland dialect of American English served as the comparison. Participants read a set of sentences with monosyllabic target words in sentence-final position. Target words varied in semantic predictability based on the preceding sentential context. Each set of sentences was produced twice by each participant—first as if talking to a friend ("plain" speech) and again as if talking to a non-native or hearing impaired listener ("clear" speech). The duration, dispersion, and trajectory length of the vowel in each target word were measured. Preliminary results suggest that, as expected, Southern /aj/ has a shorter trajectory length than Midland /aj/, and in both dialects, /aj/ has a shorter trajectory length in clear speech than plain speech. However, these processes do not interact with each other or with semantic predictability, suggesting that style and predictability effects are independent of the realization of some dialect variants.

2pSCb9. Continuous recognition memory for spoken words in noise. Susanne Brouwer (Linguistics, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, smbrouwer@hotmail.com)

Previous research has shown that talker variability affects recognition memory for spoken words [Palmeri *et al*., (1993)]. This study examines whether additive noise is similarly retained in memory for spoken words. In a continuous recognition memory task, participants listened to a list of spoken words mixed with noise consisting of a pure tone or of high-pass filtered white noise. The noise and speech were in non-overlapping frequency bands. In experiment 1, listeners indicated whether each spoken word in the list was "old" (heard before in the list) or "new." Results showed that listeners were as accurate and as fast at recognizing a word as old if it was repeated with the same or different noise. In experiment 2, listeners also indicated whether words judged as "old" were repeated with the same or with a different type of noise. Results showed that listeners benefitted from hearing words presented with the same versus different noise. These data suggest that spoken words and temporally overlapping but spectrally non-overlapping noise are retained or reconstructed together for explicit, but not for implicit recognition memory. This indicates that the extent to which noise variability is retained seems to depend on the depth of processing.

2pSCb10. When spectral smearing can increase speech intelligibility. James A. Bashford, Richard M. Warren, and Peter W. Lenz (Psychology, Univ. of Wisconsin-Milwaukee, P.O. Box 413, Milwaukee, WI 53201, bashford@uwm.edu)

Sentences were reduced to an array of 16 effectively rectangular bands (RBs) having center frequencies ranging from 0.25 to 8 kHz spaced at $^1/_3$-octave intervals. Four arrays were employed, each having uniform subcritical bandwidths which ranged from 40 to 5 Hz. The 40 Hz width array had intelligibility near ceiling, and the 5 Hz array about 1%. The finding of interest was that when the subcritical speech RBs were used to modulate RBs

of noise having the same center frequency as the speech, but having bandwidths increased to a critical (ERBn) bandwidth at each center frequency, these spectrally smeared arrays were considerably more intelligible in all but the 40 Hz (ceiling) condition. For example, when the 10 Hz bandwidth speech array having an intelligibility of 8% modulated the ERBn noise array, intelligibility increased to 48%. This six-fold increase occurred despite the elimination of spectral fine structure and the addition of stochastic fluctuation to speech envelope cues. (As anticipated, conventional vocoding with matching bandwidths of speech and noise reduced the 10-Hz-speech array intelligibility from 8% to 1%.) These effects of smearing confirm findings by Bashford, Warren, and Lenz (2010) that optimal temporal processing requires stimulation of a critical bandwidth. [Work supported by NIH.]

2pSCb11. Can physical metrics identify noise reduction settings that optimize intelligibility? Gaston Hilkhuysen and Mark Huckvale (Speech, Hearing and Phonet. Sci., Univ. College London, 31, Chemin Joseph Aiguier, Marseille 13402, France, ghilkhuysen@gmail.com)

Noise reduction algorithms often include adjustable parameters. Their settings can have important consequences for the intelligibility of an enhanced noisy speech signal, but it is not clear how to choose the best settings. In previous work, we have found that even experienced listeners prefer non-optimal settings, which degrade intelligibility. Measuring the effect of settings directly using an intelligibility listening task is often infeasible because of time, cost, or the large number of parameter combinations. In this paper we investigate whether physical intelligibility metrics can provide an efficient mean to optimize settings for noise reduction. A number of intelligibility metrics can be found in the recent research literature. An evaluation of these metrics has shown disappointing performance in their abilities to predict the absolute intelligibility of enhanced noisy speech. However, to find optimal settings, it may be sufficient to predict the relative change in intelligibility caused by a parameter change. Five metrics were used to predict the optimal settings of two parameters of a noise reduction system. The change in each metric with parameter settings are compared to listeners' performance on an intelligibility test with the enhanced signals. We show which metrics are best suited.

2pSCb12. Perception of English vowels and use of visual cues by learners of English and English native speakers. Yasna I. Pereira (Speech, Hearing and Phonet. Sci., Univ. College London, 2 Wakefield St., London WC1N 1PF, United Kingdom, yasnap@hotmail.com)

English vowels may be difficult to discriminate for many learners of English (L2 learners). Research in L2 speech perception has shown that the use of visual cues improves speech perception, at least for visually salient contrasts. This study investigated the use of visual cues in the perception of English vowels by L2 Advanced learners (Spanish native speakers) and English native speakers (ENS). Thirty-seven L2 learners and 20 ENS were given a vowel test that presented real CVC words in audio (A), audiovisual (AV), and video-alone (V) mode. The A and AV conditions were presented in noise (-10 dB SNR) to ENS and in quiet to L2 learners. For ENS, identification rates were significantly higher in AV than in A condition, suggesting there were visual cues to vowel identity. For L2 learners, A scores were significantly lower than for ENS, and AV scores did not differ significantly from results in A mode. This suggests low sensitivity to visual cues to vowel identification, though L2 learners achieved better than chance scores when forced to attend to visual information in the V mode. These results support previous findings of relatively poor sensitivity to visual cues to phoneme identity in L2 learners.

2pSCb13. The development of clear speech strategies in 9–14 year olds. Michèle Pettinato and Valerie Hazan (Speech, Hearing and Phonet. Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, m.pettinato@ucl.ac.uk)

This study investigated the development of global clear speech strategies of child talkers. Two groups of 20 talkers aged 9–10 (children) and 13–14 (teens) were recorded in pairs while they carried out spot the difference picture tasks (diapix), either hearing each other normally (NB condition) or with one talker hearing the other via a three-channel noise vocoder (VOC condition). Acoustic-phonetic analyses focused on the talker having to overcome the communication barrier. Data were compared to those for twenty of the adults in Hazan and Baker [J. Acoust. Soc. Am. **130**, 2139–2152 (2011)]. The three age groups did not differ in task transaction time for NB, but children took significantly longer to complete the task in VOC than teens or adults who took equally long. Children spoke at a slower speech rate overall than teens, while teens and adults did not differ; all groups significantly reduced their speech rate in VOC relative to NB. Adults hyperarticulated vowels in VOC, but children and teens showed only minor adaptations. These results suggest that although 9–10 year olds use some strategies to clarify their speech in difficult conditions, other strategies continue to develop into late adolescence.

2pSCb14. Consonant confusability and its relation to phonological dissimilarity. Sameer ud Dowla Khan (Linguistics, Reed College, 3203 SE Woodstock Boulevard, Portland, OR 97202, sameeruddowlakhan@gmail.com)

Consonant similarity can be measured indirectly through a language's phoneme inventory, lexicon (e.g. cooccurrence restrictions), or phonology (e.g., processes that take similarity or dissimilarity into account). It can also be measured more directly as confusability in a perception task. Thus far, consonant similarity in Bengali has only been measured indirectly, through the inventory, lexicon, and phonology. Previous studies [Khan (2006)] claim that Bengali speakers judge the similarity of consonants in echo reduplication, where the initial consonant of the base is systematically replaced with a phonologically dissimilar consonant in the reduplicant, e.g., kashi "cough" > kashi-tashi "cough, etc." but thonga "bag" > *thonga-tonga > thonga-fonga "bags, etc."). This measurement of similarity assumes a set of features assigned language-specific weights; for example, [voice] is weighted more heavily that [spread glottis], to explain why speakers treat the pair [t, th] as more similar than the pair [t, d]. But does the measurement of similarity inherent in the echo reduplicative construction correspond directly to the relative perceptibility of different consonant contrasts? The current study examines data collected in a perception experiment, comparing the relative confusability of Bengali consonants produced in noise with the claims of phonological notions of similarity associated with echo reduplication.

2pSCb15. Ambiguity related to French liaisons: The role of bottom-up and top-down processes. 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 is a phonological process involving the surfacing of an underlying floating consonant as the onset of a vowel-initial word when the word is preceded by a liaison-causing word. We used vowel- and consonant-initial ambiguous targets following four liaison-related contexts /z/, /n/, /t/, /r/ (e.g., ces onches - ces zonches). Targets included nouns and pseudonouns. Quebec-French-speaking adults performed two tasks (production, discrimination). One bottom-up (acoustic cues) and three top-down (noun token frequency, word onset probability, contextual liaison knowledge) factors were investigated in the production task. Participants had to produce the last word upon hearing each phrase. Their productions thus reflected their interpretation of the onset of the liaison-ambiguous words. Perception of acoustic cues was also tested in the discrimination task: participants judged if two phrases were different or same. No perception of any acoustical distinction was revealed in the tasks: participants often produced the intended target incorrectly, and differently intended phrasal pairs were judged same (e.g., ces onches - ces zonches). Effects of top-down information were found in the production task. Among the top-down factors, liaison knowledge related to liaison-causing words had a dominant impact on participants' interpretation of the ambiguous phrases, showing the importance of contextual knowledge in lexical recognition.

2pSCb16. The identification of high pass filtered vowels. Jeremy Donai and Dwayne Paschall (Speech-Lang.-Hearing Sci., Texas Tech. Univ. Health Sci. Ctr., 3601 4th St., Lubbock, TX 79430, jeremy.donai@ttuhsc.edu)

Vowels are typically described according to their spectral prominences (i.e., formants). Previous studies have shown that the first three formants provide important information for identification (Hillebrand *et al*. (1995), Miller (1989), Molis (2005), Peterson and Barney (1952)]. The present study measured identification accuracy for six naturally produced vowels spoken by a male and female talker with these spectral prominences removed. The six hVd tokens for each talker were high-pass filtered to remove the first three formants from the vowels and then identified by 24 normal hearing listeners. Results suggest that listeners identified a majority of the tokens above chance levels. The average identification of the male vowels was 29% (range, 17%–47%), with two vowels identified with nearly 50% accuracy. Average identification for the six female vowels was 53% (range, 37%–72%), with 5 of 6 vowels being identified with over 40% accuracy.

2pSCb17. The effects of surgical masks on speech perception in noise.
Kelsi J. Wittum, Lawrence L. Feth, and Evelyn M. Hoglund (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, wittum.2@osu.edu)

Surgical masks and blood shields worn by anesthesiologists and surgeons in hospital operating rooms may negatively impact speech communication and put patients at needless risk. Young adult subjects listened to sentences from the Speech Perception in Noise Test (SPIN) recorded by a male and female talker. All eight SPIN lists were recorded under three different speaking conditions: (1) speaking normally without any obstruction, (2) wearing a typical surgical mask, and (3) wearing a surgical mask with an attached blood shield. Multi-talker babble was mixed with the SPIN sentences at several signal-to-noise ratios to simulate conversation in noisy environments. Speaker gender and recording conditions were counterbalanced across listeners to control for learning and fatigue effects. SPIN test scores for each of the three types of recordings and both talker genders were compared in order to determine the degradation that blood-shields and surgical masks may have speech communication in the operating room. [Research supported by research grants from the Division of Social and Behavioral Sciences and a scholarship from the College of Arts and Sciences at The Ohio State University.]

2pSCb18. The role of high-frequency envelope fluctuations for speech masking release.
Søren Jørgensen and Torsten Dau (Elec. Eng., Ctr. for Appl. Hearing Res., DTU, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, sjor@elektro.dtu.dk)

The speech-based envelope power spectrum model [sEPSM; Jørgensen and Dau (2011), Jørgensen *et al*. (2013)] was shown to successfully predict speech intelligibility in conditions with stationary and fluctuating interferers, reverberation, and spectral subtraction. The key element in the model was the multi-resolution estimation of the signal-to-noise ratio in the envelope domain (SNR_{env}) at the output of a modulation filterbank. The simulations suggested that mainly modulation filters centered in the range from 1 to 8 Hz contribute to speech intelligibility in the case of stationary maskers whereas modulation filters tuned to frequencies above 16 Hz might be important in the case of fluctuating maskers. In the present study, the role of high-frequency envelope fluctuations for speech masking release was further investigated in conditions of speech-on-speech masking. Simulations were compared to various measured data from normal-hearing and hearing-impaired listeners [Festen and Plomp (1990), Christiansen *et al*. (2013)]. The results support the hypothesis that high-frequency envelope fluctuations ($>$30 Hz) are essential for speech intelligibility in conditions with speech interferers. While the sEPSM reflects effects of energetic and modulation masking in speech intelligibility, the remaining unexplored effect in some conditions may be attributed to, and defined as, "information masking."

2pSCb19. Regional linguistic variations in Canadian French: Do they affect performance on speech perception in noise?
Josée Lagacé, Stéphanie Breau-Godwin, and Christian Giguère (Health Sciences Faculty, University of Ottawa, 451 Smyth Rd, Ottawa, ON K1H 8M5, Canada, josee.lagace@uottawa.ca)

Many audiologists working with a Canadian French population use word recognition tests to measure speech perception abilities in noise. The TMB test ("Test de Mots dans le Bruit") includes four lists of 35 words presented in babble noise. The test is intended to measure the pre-cognitive perceptual stage of auditory processing and does not require understanding of the phonetic differences between speech sounds at a cognitive level. Previous studies examining performance on auditory tests similar to the TMB showed differences between populations speaking the same language but with different accentuations, such as the English spoken in the United States versus the United Kingdom. Variations in performance were attributed to accentuation differences between the speaker and the listener. To the authors' knowledge, no study appears to have investigated the effect of regional linguistic variations of Canadian French on word recognition in noise. Normative data for the TMB are being collected in three regions of Canada: Moncton, Montréal, and Ottawa. Participants are all native speakers of Canadian French, but there are important linguistic variations across the three regions. Knowledge of the effect of regional linguistic variations on the TMB performance will help refine interpretation of test results in the audiology clinics.

2pSCb20. The role of across-frequency envelope processing for speech intelligibility.
Alexandre Chabot-Leclerc, Søren Jørgensen, and Torsten Dau (Ctr. for Appl. Hearing Res., Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds plads, Bldg. 352, Kongens Lyngby 2800, Denmark, alech@elektro.dtu.dk)

Speech intelligibility models consist of a preprocessing part that transforms the stimuli into some internal (auditory) representation, and a decision metric that quantifies effects of transmission channel, speech interferers, and auditory processing on the speech intelligibility. Here, two recent speech intelligibility models, the spectro-temporal modulation index [STMI; Elhilali *et al*. (2003)] and the speech-based envelope power spectrum model [sEPSM; Jørgensen and Dau (2011)] were evaluated in conditions of noisy speech subjected to reverberation, and to nonlinear distortions through either a phase jitter process or noise reduction via spectral subtraction. The contributions of the individual preprocessing stages in the models and the role of the decision metrics were analyzed in the different experimental conditions. It is demonstrated that an explicit across-frequency envelope processing stage, as assumed in the STMI, together with the metric based on the envelope power signal-to-noise ratio, as assumed in the sEPSM, are required to account for all three conditions. However, a simple weighting of the across-frequency variance of the modulation power at the output of the (purely temporal) modulation filterbank is assumed to be sufficient to describe the data, i.e., a joint two-dimensional modulation filterbank might not be required.

2pSCb21. Using landmark detection to measure effective clear speech.
Suzanne E. Boyce, Sarah Hamilton (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Mail Location 379, Cincinnati, OH 45267, Suzanne.Boyce@uc.edu), Joel MacAuslan (S.T.A.R. Corp., Bedford, MA), Jean Krause (Dept. of Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), Rajka Smiljanic (Dept. of Linguist., Univ. of Texas at Austin, Austin, TX), and Ann Bradlow (Dept. of Linguist., Northwestern Univ., Evanston, IL)

While the relationship of speaking style to intelligibility under challenging conditions has been established, it is a common observation that some speakers seem to be more intelligible than others for most listeners. In previous work, we have reported that automatic measures based on the technique of Landmark Detection appear to track differences between Clear and Conversational speaking style. One question that remains is whether Landmark measures can be used to predict which speakers are most likely to produce highly intelligible speech. In this study, we took advantage of a set of previously acquired databases covering a total of 31 American English speakers who produced Clear and Conversational Speech to examine correlations between our Landmark-based measures and the Clear Speech productions of highly intelligible speech. Across these databases, we had data on intelligibility for 13 speakers. Results showed that speakers with high overall intelligibility in Clear Speech showed significantly different patterns on Landmark-based automatic measures, compared to speakers with more moderate performance on intelligibility measures. Applications of these results to problems in speech technology, linguistic education, and clinical practice will be discussed.

2pSCb22. Comprehending speech at artificially enhanced rates.
Lucia da Silva, Adriano V. Barbosa, and Eric Vatikiotis-Bateson (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, BC, Canada, helena.rozario@gmail.com)

This study is part of a larger study comparing the production, perception, and long-term comprehension of natural fast speech and artificially sped-up speech. In the 1950's, Fairbanks and colleagues made the interesting claim that artificially compressing speech to half its duration (twice the

rate) and listening to it twice is comprehended better than listening to the original natural speed utterance once. Since this finding has been disputed [T. Sticht, J. Expr. Ed. **37**, 60–62 (1969)], we focus here on providing baseline results for perception of naturally produced speech at conversational and fast rate and of speech sped up to twice its original speed. Specifically, we compare how well the different stimuli are perceived when presented once, twice in rapid succession, and twice with delays of hours and days between presentations. We evaluate perception for both audio-only and audiovisual stimuli.

2pSCb23. Influence of cochlear implantation on sentence intelligibility and duration. Olga Peskova (Commun. Sci. and Disord., Callier Adv. Hearing Res. Ctr., The Univ. of Texas at Dallas, Office JO3.104, 800 W.Cambell Rd., Richardson, TX 75080, oxp100020@utdallas.edu), Nirmal Kumar Srinivasan (Dept. of Elec. Eng., Univ. of Texas at Dallas, Richardson, TX), Sujin Shin, Madhu Sundarrajan, and Emily Tobey (Commun. Sci. and Disord., Callier Adv. Hearing Res. Ctr., The Univ. of Texas at Dallas, Richardson, TX)

Cochlear implants (CI) allow children with hearing loss (HL) to achieve speech perception and production outcomes that make their spoken speech understandable to normal hearing adult listeners. This capability is characterize by wide variability of scores. In order to understand the factors that contribute to the overall variability, we investigated the effects of duration of cochlear implantation on speech intelligibility and sentence duration over time. Participants were 105 children implanted between the ages of 2 and 4 and tested at 2 time points there they were 8 and 16 years old. Participants repeated McGarr sentences, which vary in length from 3 to 5 to 7 syllables. Recording were analyze using acoustic software to designate the beginning and end of each sentence in listeners who only heard one sentence from one child. Speech intelligibility scores were related statistically to the duration of each sentence. Durations of sentences that approximated that of normal hearing listeners were those with high intelligibility judgment. In addition, it appears that the children with the longest experience of CI use continue to improve their intelligibility. [Sponsored by NIH.]

2pSCb24. Influence of duration on the perception of consonants /x/ and /j/ in Chinese. Li Feng (Unit of Perceptual Psych., Kyushu Univ., Kyushu University, 4-9-1Shiobaru Minami-ku, Fukuoka, Japan, Fukuoka, Fukuoka 815-8540, Japan, rihou1986@yahoo.co.jp), Nakajima Yoshitaka, and Ueda Kazuo (Dept. Human Sci., Kyushu Univ., Fukuoka, Japan)

In the current study, we investigated whether Mandarin Chinese native speakers' perception of /x/ and /j/ was affected by the duration of consonant parts. Two perceptual experiments, in which the method of constant stimuli was employed, were conducted. Six normal-hearing adults (four females) took part in the experiment. Their average age was 24 yr. All subjects were native speakers of Mandarin Chinese. In the first experiment, Chinese syllables that begin with /x/ or /j/ were extracted from a speech database, and the consonant parts were manipulated in terms of duration. As the duration of /x/ was decreased or the duration of /j/ was increased, to a certain extent, the consonant which had been originally /x/ was perceived as /j/, and vice versa. Synthesized noises instead of recorded consonants were utilized in the second experiment, similar effects of the consonant duration appeared.

2pSCb25. Quality of older voices processed by hearing aids: Acoustic factors explaining inter-talker differences. Huiwen Goy, Margaret K. Pichora-Fuller (Psychology, Univ. of Toronto, 3359 Mississauga Rd. North, Mississauga, ON L5L 1C6, Canada, huiwen.goy@utoronto.ca), Pascal van Lieshout (Dept. of Speech-Lang. Pathol., Univ. of Toronto, Toronto, ON, Canada), and Kathryn H. Arehart (Dept. of Speech, Lang., Hearing Sci., Univ. of Colorado Boulder, Boulder, CO)

Hearing aid signal processing algorithms are often evaluated with professional recordings of voices. However, hearing aid users often listen to older talkers who may have poorer voices than younger talkers. The purpose of this study was to quantify the extent to which the acoustic and perceptual consequences of hearing aid digital signal processing algorithms differ for talkers that vary in their vocal characteristics. There were six older talkers (3 males and 3 females) selected from a larger database of 79 talkers; their voices had good, moderate, or poor quality based on perceptual data from younger and older listeners. The voices were presented in quiet and in the presence of babble noise (10 dB SNR). The voices were processed with varying amounts of frequency compression, wide dynamic range compression, and noise suppression (spectral subtraction). There were interactions between signal processing and voice characteristics as measured using the Hearing Aid Speech Quality Index (HASQI) and listener ratings of perceived sound quality of the processed speech. In this paper, we examine the possible acoustic sources that may explain these interactions, including inter-talker differences in formant space and pitch variation. [Research supported by NIH, GN Resound, NSERC, and the Canada Research Chairs program.]

2pSCb26. The effect of background noise on the ability to perceive and remember unrelated words in nonnative listeners. Meital Avivi-Reich, Caterina Y. Leung, and Bruce A. Schneider (Psychology, Univ. of Toronto Mississauga, 1909-35 Charles st.W, Toronto, ON m4y 1r6, Canada, me_avv@yahoo.com)

Nonnative listeners find it more difficult to meet the challenges presented by additional background noise than do native listeners [Ezzatian *et al*., Speech Commun. **52**, 919–929 (2010)], but it is not known whether it is more difficult for them to remember what was said in noisy situations than it is for native listeners. Previous studies have acknowledged that the effect of background noise on the ability to perceive and remember unrelated words is greater in older adults than younger adults. The present study investigates auditory memory performance in nonnative younger adults, using a paired-associate paradigm in three conditions: quiet, continuous babble and babble during word presentation only. Noise levels were adjusted to equate for individual differences in the ability to identify single words in noise. The initial results suggest that nonnative listeners perform similarly to native young adults in the quiet and continuous conditions but worse in the babble during the word-presentation-only condition. These results suggest that stream segregation may be slower in nonnative listeners when the masker and the target words start at the same time.

2pSCb27. Advantage of talker differences and spatial separation for speech-on-speech listening in younger and older adults with good audiograms. Jana Besser (ENT/Audiology, VU Univ. Med. Ctr., De Boelelaan 1118, Amsterdam 1081HV, Netherlands, besser.jana@gmail.com), Kathleen Pichora-Fuller (Dept. of Psych., Univ. of Toronto, Mississauga, ON, Canada), and Joost M. Festen (ENT/Audiology, VU Univ. Med. Ctr., Amsterdam, Netherlands)

Older adults have more difficulty than younger adults understanding speech when there is competing speech, even if they have good audiograms. Age-related differences in listening may be due to declines in auditory temporal processing and/or cognition. We administered the LiSN-S [Cameron *et al*. (2011)] to measure speech reception thresholds (SRTs) in younger and older adults with good audiograms. There were four test conditions, in which the target and competing speech were presented with the same or different voices at the same or different locations. Compared to younger listeners, older listeners obtained worse SRTs in all test conditions and they realized less advantage from talker differences and spatial separation between the target and competing speech. For both groups, the results obtained in the four test conditions were strongly associated with each other. We also assessed cognitive abilities and auditory temporal processing in the older adults. LiSN-S results in this group were strongly associated with measures of cognition, measures of temporal processing (tapping the use of fine structure and gap cues), as well as pure-tone averages (PTA) for 9 and 10 kHz, but not PTAs for frequencies in the standard audiometric range.

Session 2pSP

Signal Processing in Acoustics: Acoustic Signal Processing for Various Applications

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Contributed Papers

1:00

2pSP1. Ideal signals and processing for continuous active sonar. Harry A. DeFerrari (Appl. Marine Phys., Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, hdeferrari@rsmas.miami.edu)

The ideal signal for continuous active sonar would be linear in both time and Doppler and have no time or Doppler leakage. Here two signals/signal processing methods are presented that have half the requisite condition—zero time leakage. But we show a way to use the time property to eliminate Doppler interference (clutter) from zero Doppler reverberation (bottomed) sources and from direct arrivals. Here we develop an m-sequence processed with a matched filter approach and a special class of inverted binary sequences processed with a matched-inverse filter. In both cases, there is a perfect pulse correlation property; that is, a large signal when they line up in time and hard zero when they do not. This property is used to develop a Complete Ortho-Normal (CON) data sets in both waveform and pulse response space. The CON pulse response allows elimination of the easily identified zero Doppler reverberation arrivals. Then, reprocessing for Doppler, the zero Doppler signals are cleanly removed as is all of their Doppler leakage. In effect one has the ideal clutter free signal with leakage from direct arrivals and bottom reverberation removed and a target signal displayed in a time-Doppler plane against a noise only background.

1:20

2pSP2. Toward blind reverberation time estimation for non-speech signals. João F. Santos (INRS-EMT, Institute National de la Recherche Scientifique, 800, Rue de La Gauchetière Ouest, Ste. 6900, Montreal, QC H5A-1K6, Canada, jfsantos@emt.inrs.ca), Nils Peters (Qualicomm Technol. Inc., Berkeley, CA), and Tiago H. Falk (INRS-EMT, Institute National de la Recherche Scientifique, Montreal, QC, Canada)

Reverberation time (RT) is an important parameter for room acoustics characterization, intelligibility and quality assessment of reverberant speech, and for dereverberation. Commonly, RT is estimated from the room impulse response (RIR). In practice, however, RIRs are often unavailable or continuously changing. As such, blind estimation of RT based only on the recorded reverberant signals is of great interest. To date, blind RT estimation has focused on reverberant speech signals. Here, we propose to blindly estimate RT from non-speech signals, such as solo instrument recordings and music ensembles. To estimate the RT of non-speech signals, we propose a blind estimator based on an auditory-inspired modulation spectrum signal representation, which measures the modulation frequency of temporal envelopes computed from a 23-channel gammatone filterbank. We show that the higher modulation frequency bands are more sensitive to reverberation than the modulation bands below 20 Hz. When tested on a database of non-speech sounds under 23 different reverberation conditions with early decay time (EDT) ranging from 0.26 to 7.6 s, a blind estimator based on the ratio of high-to-low modulation frequencies outperformed two state-of-the-art methods and achieved correlations with EDT as high as 0.80 for solo instruments and 0.75 for ensembles.

1:40

2pSP3. Feedback active noise control in a crew rest compartment. Delf Sachau and Oliver Pabst (Mechatronics, Helmut-Schmidt-Univ., Holstenhofweg 85, Hamburg, Hamburg D-22043, Germany, sachau@hsu-hh.de)

Active systems for noise cancelation are typically used to reduce low frequency noise (below 500 Hz). In this study, a method to reduce the delay in the control loop of feedback noise control systems is discussed. There are certain applications in which we may choose to control a specific frequency range; hence, a selection of the control-band must be performed. This selection is commonly done with analog filters introducing delay in the signal path. However, in feedback systems, the reference signal is calculated from the error signal which makes their performance more sensitive to these delays. In this contribution, the filtered-error feedback filtered-reference LMS algorithm (FE-FBFxLMS) is analyzed. Consequently, control-band selection is performed in an auxiliary loop removing the additional delay required in the signal path. This method of band selection also allows a reduction of the delay due to the anti-aliasing filters by using faster low-order filters in terms of group delay. With this delay reduction, higher noise attenuation is expected at the cost of a slower convergence rate. This method is applied to control broadband noise in a single channel feedback system in simulation and experiment.

2:00

2pSP4. A polar-coordinate-discretized wave equation finite-difference time-domain simulation for controlling the emission characteristics of sound source. Kota Nakano, Masato Nakayama, Takanobu Nishiura, Yoichi Yamashita (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Noji-higashi, Kusatsu, Shiga 525-8577, Japan, cm010064@ed.ritsumei.ac.jp), and Toshiyuki Kimura (Universal Commun. Res. Inst., National Inst. of Information and Commun. Technol., Soraku-gun, Japan)

In sound field simulation with conventional WE-FDTD, the simulated field and wave equation are discretized with symbol time domain and spatial domain with Cartesian coordinate, and sound pressure distribution is calculated. Spatial-discretization with Cartesian coordinate easily achieves calculation and sound synthesis. However, the discretized Cartesian coordinate cannot model sound emission characteristic exactly, because the emission characteristics is under spherical diffusion and discretize Cartesian coordinate cannot define spherical. Emission characteristic of sound source is indispensable for sound field reproduction. Then, we propose a new approach for WE-FDTD method to simulate emission characteristic of sound source object. Polar coordinate discretization is employed in the new approach instead of Cartesian coordinate. The wave equation with spatial Cartesian coordinate is projected onto polar coordinates, and discretized. The discretized polar coordinate is better way to define spherical wave surface than Cartesian one, and it is expected that the emission characteristics are simulated more exactly. The objective evaluations were conducted for the proposed approach. According to the result, the sound field simulation with the new approach can flexibly define exact emission characteristics. However, the discretized interval of polar coordinate is coarse in distance. Verifying the interval limit is important issue in future work.

2:20

2pSP5. Wind noise reduction using empirical mode decomposition. Kohei Yatabe and Yasuhiro Oikawa (Intermedia Art and Sci., Waseda Univ., 59-407 3-4-1 Okubo, Shinjuku-ku, Tokyo 169-8555, Japan, k.yatabe@asagi.waseda.jp)

One common problem of recorded sound outdoors is interfusion of wind noise, which has highly non-stationary characteristics. Although there are a lot of noise reduction methods which produce good results for general kinds of noises, most methods perform worse for wind noise due to its non-stationary nature. Therefore, wind noise reduction need special technique to overcome this non-stationarity. Empirical mode decomposition (EMD) is a relatively new method to decompose a signal into several nonlinear and non-stationary bases which are modeled as amplitude and frequency modulated sinusoids that represent wind noise well. Thus, EMD has a possibility to reduce wind noise from recorded sounds in entirely different way from ordinary methods. In this paper, a preliminary discussion of applying EMD to wind noise reduction is presented. Since EMD decomposes a signal into monocomponent bases, it is easier to treat them as analytic signals via Hilbert transform. Our method utilize this characteristics in order to reduce wind noise. The experiment is performed on female voice superimposed with wind noise and shows it possibility and effectiveness.

2:40

2pSP6. Efficient speech encryption using chaotic cat map for code-excited linear prediction based coders in packet networks. Fatiha Merazka (Telecommunications, Univ. of Sci. & Technol. Houari Boumediene, P.O. Box 32 El Alia, Algiers 16111, Algeria, fmerazka@usthb.dz)

The increasing importance of multimedia applications is placing a great insist on content protection and customer privacy. Communications can be intercepted, especially over wireless links. Since encryption can effectively prevent eavesdropping, its use is widely advocated. The codec G. 729 based CS-ACELP algorithm is standardized as voice codec by ITU-T for multimedia and Voice over Internet Protocol (VoIP) applications. In this paper, we introduce a speech encryption method based chaotic cat map algorithm. Cat map extended to two-dimensional NxN matrix. It takes concepts from linear algebra and uses them to change the positions of the values of the matrix. The result after applying the Cat Map will be shuffled signals that contain the same values of the original signals. We applied our encryption scheme to the standard ITU-T G.729 standard speech coder to evaluate its performance. Simulation results show that G.729 based cat map encryption is very efficient since the encrypted speech is similar to a white noise. The perceptual evaluation of speech quality (PESQ) and enhanced modified bark spectral distortion (EMBSD) tests for speech extracted from TIMIT database confirm the efficiency of our proposed scheme.

3:00–3:20 Break

3:20

2pSP7. An investigation into the relationship between sound quality and demodulation ratio for parametric loudspeakers. Daisuke Ikefuji, Masato Nakayama, Takanobu Nishiura, and Yoichi Yamashita (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, cm000074@ed.ritsumei.ac.jp)

Recently, parametric loudspeakers with a powerful ultrasound have been utilized for reproduction of sound to particular area. Amplitude of the ultrasound is modulated with a target audible sound. The emitted amplitude modulated wave is demodulated into the target audible sound in the air. Demodulation ratio affects sound quality of the reproduced sound with parametric loudspeakers. Moreover, demodulation ratio depends on the distance between the parametric loudspeaker and the listener. Therefore, the listener should utilize parametric loudspeakers at the suitable distance, which is required for sufficient demodulation of the amplitude modulated wave. Thus, we have proposed the criterion for measuring the demodulation ratio. However, it had not been conducted to investigate a relationship between subjective sound quality and demodulation ratio. In this paper, we investigate the relationship for estimation of the suitable distance. We carried out subjective evaluation experiment for confirming sound quality at each distance. As a result, we confirmed the consistency about fluctuation tendency of the subjective sound quality and demodulation ratio at each distance.

3:40

2pSP8. An adaptive noise power spectral density estimation of noisy speech using generalized gamma probability density function. Xin Dang (Information Sci. and Technol., Grad. School of Sci. and Technol., Grad. School of Eng., Hamamatsu, Shizuoka 432-8012, Japan, f5045013@ipc.shizuoka.ac.jp) and Takayoshi Nakai (Information Sci. and Technol., Grad. School of Sci. and Technol., Hamamatsu, Shizuoka, Japan)

An estimation of the power spectral density (PSD) of noise is a crucial part to retrieve speech in a noisy environment. A novel estimation method for non-white noise of noisy speech on the basis of a generalized Gamma distribution is proposed. Because of highly non-stationary nature of speech, its probability density function (PDF) is difficult to derive using any modeling technique, while a segmental noise is more stationary and can be fitted more accurately by a generalized Gamma PDF, which is a natural extension of the Gaussian modeling of a non-white components distribution. In the experiment, different types of non-white noises are added to the clean speech signal at different SNRs to study the estimation of noise using different types of PDF. It is found that non-white noise spectrums fit more accurately on the generalized Gamma PDF with adaptive parameters instead of a Gaussian distribution function. The reported generalized Gamma PDF model shows the best performance to estimate the noise spectral amplitudes as compared with Minimum Statistics (MS), Speech absence Probability (SAP), and MMSE based PSD estimation methods. The performance of the proposed noise estimation is good when it is integrated with the speech enhancement technique as demonstrated by both the subjective and objective measures.

4:00

2pSP9. Markov random field in speech enhancement: Application for tonal languages. Tanawan Saimai, Charturong Tantibundhit (Elec. and Comput. Eng., Thammasat Univ., 99 Moo 18 Phaholyothin Rd., Khlongluang, Pathumthani 12120, Thailand, 5310030068@student.tu.ac.th), Chutamanee Onsuwan (Linguistics, Thammasat Univ., Khlongluang, Pathumthani, Thailand), and Chai Wutiwiwatchai (National Electron. and Comput. Technol. Ctr. (NECTEC), Khlongluang, Pathumthani, Thailand)

This paper proposed speech enhancement algorithm based on Markov random field (MRF) model for Thai, a tonal language. Firstly, a noisy speech signal is transformed using the short time Fourier transform (STFT). In so doing, noise is removed and speech is preserved, especially harmonics information as f0 patterns are relevant perceptual cues for lexical tones. The voice activity detector is used to classify each STFT time frame into voiced and unvoiced. Harmonics information is retrieved from each voiced time frame, where four neighborhoods of the analyzed STFT coefficients include its adjacent time frames (left, right) and nearest harmonics (top, bottom). For the unvoiced, four adjacent coefficients (left, right, top, and bottom) are used. A two-state MRF model is used to classify STFT coefficients into speech and noise. Those with speech state are retained, while the rest is set to zero. The enhanced speech is estimated by the inverse STFT. Results from quality evaluation test on four sets of Thai rhyming words corrupted by white noise at SNR levels of 0, 5, and 10 dB showed that the proposed algorithm significantly improved SNR of noisy speeches compared with spectral subtraction (1.3 dB on average) and Wiener filtering (1.9 dB on average).

4:20

2pSP10. Dereverberation of a closed test section of a wind tunnel with a mult imicrophones cesptral method. Daniel Blacodon and Jean Bulté (DSNA/ACOU, ONERA, 29, avenue de la Division Leclerc, BP N° 72, Châtillon 92322, France, daniel.blacodon@onera.fr)

Today, the manufacturers of aircraft want to perform at the same time aeroacoustic and classical aerodynamic tests in the wind tunnels with closed test sections in order to reduce test duration and costs involved in new aircraft developments. However, this kind of wind tunnels is not optimal for acoustics testing because the measurements are often masked by background noise (i.e., noise generated by the wind tunnel flow mechanism) and reverberations on the walls without acoustic liner within the test section. This is a serious problem because these unwanted phenomena limit the accuracy of identification, and quantification of acoustic sources. The reduction of background noise can be obtained with subspace techniques or noise subtraction based on a noise reference measurement. Concerning the reduction of the contamination of the echos, it can be achieved with an inverse filtering technique or with a single microphone cepstral method. A new approach is proposed in this study to remove the spurious effects of the echos. It is based on multi microphones cesptral technique. This new technique has been successfully applied on numerical simulations, and experimental data. The theoretical developments of this method and the obtained results will be presented in the full paper.

TUESDAY AFTERNOON, 4 JUNE 2013 511AD, 1:00 P.M. TO 5:40 P.M.

Session 2pUWa

Underwater Acoustics and Acoustical Oceanography: Ocean Ambient Noise

Rex K. Andrew, Chair
Appl. Phys. Lab., 1013 NE40th St., Seattle, WA 98105

Contributed Papers

1:00

2pUWa1. Seabed characterization using ambient noise and compact arrays on an autonomous underwater vehicle. Peter L. Nielsen (Res. Dept., STO-CMRE, V.S. Bartolomeo 400, La Spezia 19126, Italy, nielsen@cmre.nato.int), Martin Siderius (Elec. and Comput. Eng. Dept., Portland State Univ., Portland, OR), Jim Miller (Research Dept., STO-CMRE, La Spezia, Italy), Steven Crocker (Sensors & SONAR Systems Dept., NUWC, Newport, RI), and Jennifer Giard (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Estimating the seabed geoacoustic properties at various fidelity levels has been a research topic for several decades. The majority of the applied seabed characterization techniques often require significant involvement of surface vessels, complex experimental setup, and human interaction. Technical advances in underwater autonomy and the development of energy efficient electronics provide new opportunities to optimize underwater environmental surveys in particular of the seabed. In 2012, the CMRE conducted the GLASS'12 experiment in the Mediterranean Sea with the objective to investigate the feasibility of utilizing a hybrid autonomous underwater vehicle equipped with a compact nose array for long-duration seabed characterization over large areas. The vehicle has the capability of operating in traditional propulsion and glider mode, and the nose-mounted array consists of a 5-element vertical and 4-element tetrahedral array. The sound sources used as information carrier were ambient noise, e.g., sea surface generated noise and loud distant sources of opportunity. The experimental setup together with the newly developed autonomous equipment will be presented and examples of inferred reflection loss and sub-bottom profiling from the ambient noise are compared to ground truth measurements. [Work supported by the STO-CMRE, ONR-G Grant No. N62909-12-1-7040, the ONR N-STAR/ILIR program.]

1:20

2pUWa2. Monterey Bay ambient noise profiles using underwater gliders. Tarun K. Chandrayadula, Chris W. Miller, and John E. Joseph (Oceanography, Naval Postgrad. School, 650 Sloat Ave., # 3, Monterey, CA 93940, tkchandr@nps.edu)

In 2012, during two separate week-long deployments, underwater gliders outfitted with external hydrophones profiled the upper 100 m of Monterey Bay. The environment contains various noises made by marine mammals, ships, winds, and earthquakes. Unlike hydrophone receivers moored to a fixed location, moving gliders measure noise variability across a wide terrain. However, underwater mobile systems have limitations such as instrument and flow noise, that are undesired. In order to estimate the system noise level, the hydrophones on the gliders had different gain settings on each deployment. The first deployment used a 0 dB gain during which the ambient noise recordings were dominated by the glider. The second used two hydrophones, one with a 0 dB gain and the other with 20 dB. Apart from system sounds, the higher-gain hydrophone also recorded far-away sources such as whales and ships. The noise recordings are used to estimate the spectrograms across depth and record time. The spectrograms are integrated with the glider engineering data to estimate histograms of noise power as a function of depth and glider velocity. The statistics from the two different deployments are compared to discuss the value of gliders with external hydrophones in ambient noise studies.

1:40

2pUWa3. Measuring the spatial characteristics of the ambient noise field from an autonomous underwater vehicle. Stephanie Fried and Henrik Schmidt (Massachusetts Inst. of Technol., 77 Massachusetts Ave., Rm. 5-204, Cambridge, MA 02139, eowyn@mit.edu)

For autonomous underwater vehicles (AUVs), the primary method of sensing the local environment is through acoustics. The local noise field contains a wealth of information the AUV uses—from target tracking to communication to general understanding of the environment. A measure of the spatial composition of the ambient noise field can provide details about the physical environment as well as information for the AUV to incorporate into its control decisions. The challenge is in accurately measuring the directionality of the noise field from a single line array and continuously updating this measure to reflect changes in the environment and additional information as the AUV moves. Here we present a method for continuously assessing the spatial characteristics of an ocean ambient noise field measured by an AUV with a towed hydrophone array.

2:00

2pUWa4. Synthetic-array beamforming for bottom-loss estimation using marine ambient noise. Lanfranco Muzi and Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, muzi@pdx.edu)

Previous studies have shown that, using a vertical line array, the bottom loss can be estimated in an ambient-noise field from the output power of beams steered toward the sea surface and beams reflected off the seabed. With short arrays, the low angular resolution of the bottom-loss estimate is one of the main limitations of the approach. *Synthetic-array processing* is proposed as a technique that can improve the angular resolution of the bottom-loss estimate to a level comparable to that of an array with twice as many physical sensors (at equal inter-sensor spacing). The proposed technique follows naturally from a new derivation in frequency-wavenumber domain of the bottom-loss estimation procedure. The conditions under which the approach can be successfully applied are analyzed, with particular regard to the need for the array cross spectral density matrix to be "close" to Toeplitz. A metric is proposed for assessing this particular circumstance in practice. The technique is illustrated using synthetic data from simulations and then applied to data from several experimental campaigns. Results show that a 16-element synthetic array can achieve an angular resolution comparable to that of a 32-element array.

2:20

2pUWa5. Flow noise measurements at strong tidal current area in Uldolmok Waterway. Myungkwon Ko and Jee Woong Choi (Dept. of Marine Sci. and Convergence Technol., Hanyang Univ., 55 Hanyangdaehak-ro, Sangnok-gu, Ansan, Gyeonggi-do 426-791, South Korea, buymk@hanyang.ac.kr)

Flow noise is a kind of hydrodynamic noise, which is created by turbulent flow in the boundary layer around the hydrophone. Although the turbulent pressure is not a true acoustic noise in that its influence decreases rapidly with distance, it acts as a self-noise source. In general, since the spectrum level of flow noise has been reported to increase rapidly with the increment of flow speed, it is possible to monitor the current velocity from the flow noise measurements. Uldolmok waterway in Korea is one of the locations where currents are very strong, with maximum speed of about 5 m/s. The measurements of flow noise were conducted in Uldolmok waterway using two different shapes of hydrophone. In this talk, the flow noise spectra for various flow speeds will be presented for the frequency range of 20–100 Hz, and the comparison of spectra between two different-shape hydrophones will be given.

2:40

2pUWa6. On the origins of ambient biological sounds in shallow water tropical ecosystems. Simon E. Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., 3725 Miramar St. Apt. C, La Jolla, CA 92037, sfreeman@ucsd.edu), Forest L. Rohwer, Allison Gregg, Laura Coleman (Rohwer Lab., San Diego State Univ., San Diego, CA), and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

Although discovered more than 60 years ago, the origins of much ambient underwater biological noise remain unclear. Snapping shrimp sounds dominate some environments but elsewhere the shallow-water biological sound field is often heterogeneous. Here we show that dominant components of underwater ambient noise recorded on coral reefs around five islands in the central Pacific may be caused by the interaction of hard-shelled benthic macro-organisms with the substrate. Recordings show a consistent, nightly 4.7 to 6.9 dB increase in estimated pressure spectral density level in the 11 to 17 kHz band with a spectral peak centered between 14 to 15 kHz. Underwater time-lapse photography reveals a marked night-time increase in benthic invertebrate activity at most locations, temporally consistent with the increase in pressure spectral density level. Intensity-filtered recordings of an example species, the hermit crab *Clibanarius diugeti*, in quiet aquarium conditions reveal that transient sounds produced by interaction between the crustacean's carapace, shell, and coral substrate are spectrally consistent with the central Pacific recordings. Passive acoustic monitoring of such ambient noise may be useful as a complementary ecological survey technique to SCUBA-based visual observations, which are typically poor in estimating the abundance and diversity of cryptobenthic organisms.

3:00

2pUWa7. Prediction of underwater noise and far field propagation due to pile driving for offshore wind farms. Stephan Lippert, Tristan Lippert, Kristof Heitmann, and Otto von Estorff (Inst. of Modelling and Computation, Hamburg Univ. of Technol., Denickestr. 17, Hamburg 21073, Germany, s.lippert@tu-harburg.de)

Wind energy plays a key role toward a greener and more sustainable energy generation. Due to limited onshore areas and possible negative effects on human living space, offshore wind parks become increasingly popular. During construction by pile driving, however, high levels of underwater sound emission are observed. To avoid negative effects on marine mammals and other sea life, different approaches, like, e.g., bubble curtains or cofferdams, are currently investigated to cut down the sound pressure levels. In order to predict the expected underwater noise, both with and without sound damping measures, numerical simulation models are needed to avoid complex and costly offshore tests. Within this contribution, possible modeling strategies for the prediction of underwater noise due to pile driving are discussed. Different approaches are shown for the direct adjacencies of the pile and for the far field sound propagation. The effectivity of potential noise mitigation measures is investigated using a detailed finite element model of the surroundings of the pile. The far field propagation in the kilohertz range at distances of several kilometer from the pile, on the other hand, is computed by a model based on wavenumber integration. Finally, the model validation with corresponding offshore tests is addressed.

3:20

2pUWa8. Wind-dependence of low-frequency ambient noise in the deep-ocean sound channel. Stephen Nichols (Grad. Prog. in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, smn5198@psu.edu) and David L. Bradley (Appl. Res. Lab., The Penn State Univ., State College, PA)

In the low-frequency range (1–125 Hz), the deep-ocean ambient noise field is produced by seismic, marine life, ship traffic, and wind-dependent hydrodynamic noise mechanisms. This study focuses on the contribution of wind-related source mechanisms to the overall ambient noise field, as well as previous attempts to understand the physics of these mechanisms. The Comprehensive Nuclear-Test Ban Treaty Organization (CTBTO) hydroacoustic monitoring system has produced nearly continuous recordings of the low-frequency deep-ocean ambient noise field at sites in the Pacific, Atlantic, and Indian Oceans, each spanning several years in length. Additionally, wind speed data have been recorded at the host island of each station by the National Oceanic and Atmospheric Administration (NOAA). Correlation techniques are used with these two datasets to determine the relationship between wind speed and the sound level in different frequency bands, and to determine the prominence of wind-related noise in the combined ambient noise spectrum. Results from the three sites are compared to each other to assess the uniformity of wind-generated noise over the world's ocean basins.

3:40

2pUWa9. Model for underwater noise radiated by submerged wind turbine towers. Todd Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (ARL:UT, P.O. Box 8029, Austin, TX 78713, hamilton@mail.utexas.edu)

Sustained tonal noise radiated by towers supporting offshore wind turbines contains energy in frequency bands that may disturb marine mammals, or interfere with passive sonar and seismic sensors and underwater communication equipment. Understanding the generation and propagation of underwater noise due to the operation of wind farms is important for determining strategies for mitigating the environmental impact of these noise sources. An analytic model based on a Green's function approach was previously developed for the sound radiated in the water column by a pulsating cylindrical structure embedded in horizontally stratified layers of viscoelastic sediment [Hay *et al*., J. Acoust. Soc. Am. **130**, 2558 (2011)]. This model has since been adapted to include relaxation and viscous losses in seawater and empirical loss factors for the sedimentary layers. In order to validate the model simulations were compared with reported measurements collected near an operating wind turbine that include radial acceleration of the tower, taken to be the source condition, and sound pressure levels in the water column. For long-range propagation over range-dependent environments, the analytic model has been coupled to a parabolic equation code. Simulations are presented for several bathymetries, sediment types, and tower array configurations. [Work supported by Department of Energy DE-EE0005380.]

4:00

2pUWa10. A quasi-analytic model of the underwater sound signal from impact driving of an offshore semi-infinite pile. Marshall V. Hall (Marshall Hall, 9 Moya Crescent, Kingsgrove, NSW 2208, Australia, marshallhall@optushome.com.au)

A quasi-analytic model is derived for the underwater sound signal radiated when an offshore pile is struck on its face by a hammer. The pile is modeled as a semi-infinite cylindrical shell of an elastic solid. The impact generates a pulse of vibration that travels down the pile at the longitudinal

sound-speed. At a given distance below the pile face, the axial displacement after the peak has arrived decreases exponentially with time. There are two coupled equations of motion for the axial and radial displacements. A closed form expression is derived for the radiated sound pressure (which is proportional to the radial acceleration) in terms of the Poisson ratio and Young's Modulus of the pile material, the hammer velocity, contact area between hammer and pile, pile radius, hammer mass, the pile's longitudinal sound-speed, and the sound-speed and density of the external medium. This model is applied to a published scenario for which the radiated sound pressure had been computed using a Finite Element Model, but is found to produce a different result. Some assumptions used in the model are identified that may explain the difference.

4:20

2pUWa11. Implementing physical constraints for noise only normal mode shape estimation. Ian M. Rooney, John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts at Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, irooney@umassd.edu), and Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., Fairfax, VA)

Many underwater acoustic tasks employ a vertical line array to sample a continuous wave pressure field. In the absence of strong local sources, the noise sampled by an array consists of both a spatially correlated and a spatially uncorrelated component. The correlated component is generally due to distant sources and can be described in terms of the modes of the waveguide. If the number of propagating modes is less than the array size, the diffuse noise lies within a lower dimensional subspace than the array data. The mode shapes defining this subspace can be estimated from noise measurements. Propagation physics constrain the mode shapes defining this subspace to be real, but the basis vectors obtained from a singular value decomposition of noise snapshots are generally complex. A phase rotation for each basis vector is required to rectify this. This work proposes a weighted average of the phase angles for each element that minimizes the variance of the rotation angle estimate. Simulations compare the proposed algorithm against prior approaches such as rotating each basis vector by the phase of the largest magnitude sample, rotating by the average of the phase, or taking the magnitude and ignoring the phase. [Work supported by ONR.]

4:40

2pUWa12. Analysis of the vertical structure of deep ocean noise using measurements from the SPICEX and PhilSea experiments. Kathleen E. Wage, Mehdi Farrokhrooz (Elec. and Comput. Eng. Dept., George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu), Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA)

There are open research questions about the vertical structure of low-frequency ambient noise in deep water. For example Gaul *et al.*'s [*IEEE JOE* (2007)] analysis of the Church Opal data set showed that noise decreases substantially (on the order of 20 dB) below the critical depth, whereas other researchers have reported more modest reductions [Morris, *J. Acoust. Soc. Am.* (1978)]. Two deep water experiments provided a unique opportunity to measure ambient noise using large vertical arrays. In 2004–2005, SPICEX used two arrays to sample a North Pacific environment. One array was centered on the sound channel axis, and the other array had hydrophones above and below the critical depth. In 2010–2011, the PhilSea experiment deployed a single array with 150 hydrophones spanning the full water column. Both experiments made repeated short measurements (each 2–3 min long) of the field at the arrays. This talk compares the ambient noise observed during SPICEX and PhilSea with results reported in the literature. Since these data sets contain receptions over the period of a year, we focus on the seasonal dependence of the noise field. In addition to investigating noise level as a function of depth, we consider wind dependence and vertical directionality.

5:00

2pUWa13. Modeling of underwater piling noise mitigation using an array of soft spheres in the ocean. Keunhwa Lee (Ocean Eng., Seoul National Univ., Kwanak-ro 1, Seoul 151742, South Korea, nasalkh2@snu.ac.kr), Kyungmin Baik (CFFA Div. of Physical Metrology, Korea Res. Inst. of Standards and Sci., Daejeon, South Korea), and Woojae Seong (Ocean Eng., Seoul National Univ., Seoul, South Korea)

The ocean noise generated by marine piling affects severely fish, other marine life, and fishery activities. Accordingly, a few kind of noise mitigation system are presented. Among them, the noise mitigation system using soft scatterers such as air bubbles or rubber spheres is reported to show higher noise reduction than the classical cofferdam system composed of mass-absorbing materials. In this proceeding, a numerical scheme is developed to model and design the noise mitigation system using an array of soft spheres. This scheme is originally based on self-consistent equation of Zhen Ye for multiple scattering [Z. Ye and A. Alvarez, Phys. Rev. Lett. **80**, 3503 (1998)]. We generalize the original self-consistent equation for the oceanic waveguide using the waveguide green function. This generalized self-consistent equation is useful to model the noise propagation through an array of soft spheres in the ocean and assess the ability of the noise mitigation system. The effect of the oceanic waveguide on the noise reduction is studied and the validity of effective medium approach for a bubbly layer is also analyzed numerically.

5:20

2pUWa14. Spatial filtering in ambient noise crosscorrelation. Olivier Carriere, Peter Gerstoft, and William S. Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr, La Jolla, CA 92093-0238, ocarriere@ucsd.edu)

Ambient noise crosscorrelation is an attractive approach for estimating the properties of a propagation medium (presence of scatterers, wave propagation speed, etc.) without having to deploy active acoustic source. According to the theory, Green's function between pair of receivers can be extracted from the crosscorrelation of the received signals, given that the noise source is diffuse. In practice, this ideal condition is rarely met in the ocean and directional sources, often related to human activity in the vicinity of the receivers, may bias the travel time estimates. Using array processing, matrix spatial filters can be built independently from the environment properties (as opposed to model-based techniques) to filter specific directions of arrival. Such filter might be useful to remove an interferer or to restrict the effective source distribution to a reduced cone area. Here we study the use of matrix spatial filters for removing unwanted contributions in the crosscorrelations. Based on theory and simulations, we discuss the effect of array size, dimension and geometry, processed frequency band, and environmental conditions in the filtered crosscorrelations. An example of application to real data from the Shallow Water '06 experiment further illustrates the potential of the method.

Session 2pUWb

Underwater Acoustics and Acoustical Oceanography: Arctic Acoustics and Applications

Stan E. Dosso, Cochair
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Juan I. Arvelo, Cochair
Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723

Chair's Introduction—12:55

Invited Papers

1:00

2pUWb1. Under-ice shallow-water sound propagation and communication in the Baltic Sea. Erland Sangfelt, Sven M. Ivansson, and Ilkka Karasalo (Underwater Res., Swedish Defence Res. Agency, FOI kista, Stockholm SE-16490, Sweden, erland.sangfelt@foi.se)

Sound propagation under ice in the Baltic is of interest for military as well as civilian purposes. Important questions are, for example, how sonar systems are affected by ice in shallow waters and how marine mammal life under ice is affected by ship traffic. Changing climate and increasing ship traffic are factors of great concern also in the Arctic region. Modeling results for sound propagation under ice in the Baltic Sea are presented for low as well as high frequencies. The sound propagation is influenced by bottom as well as ice-cap interaction in these shallow waters. The low- and high-frequency modeling is performed with wavenumber and ray models, respectively. Both types of models are amended to include a solid ice layer on top of the water column. In addition, the modeling efforts are extended to study the performance of an underwater communication system developed at FOI. The modeling results are evaluated using data from sea trials in the Gulf of Bothnia under ice-covered and ice-free conditions.

1:20

2pUWb2. Propagation of seismic exploration noise in the Marginal Ice Zone. Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no) and Hanne Sagen (Nansen Environ. and Remote Sensing Ctr. (NERSC), Bergen, Norway)

This paper presents data from ambient noise recordings in the Marginal Ice Zone (MIZ) in the Fram Strait (deep water) and Barents Sea (shallow water) with focus on noise due to seismic exploration activity. Data were recorded on fields of sonobuoys deployed from P-3C aircraft in June 2011 and May 2012, each covering a 100 km x 100 km area. Recordings from three bouys in each area, from open water to within the solid ice cover, are analyzed for low-frequency (10 Hz–1 kHz) noise levels, with broadband noise due to distant seismic exploration identified. Strong attenuation of this noise component with distance into the MIZ is observed in the shallow-water data, whereas noise levels decrease less with distance in the deep-water data despite under-ice conditions. Propagation loss is modeled using a raytrace model including under-ice reflection loss due to an elastic ice cover, with environment input from the TOPAZ coupled ocean-sea ice model and from satellite images. Properties of the noise field, modeling results, and model-data discrepancies will be discussed. [Data collected under the ACOBAR and WIFAR projects at NERSC.]

1:40

2pUWb3. Modelled and measured sound levels from a seismic survey in the Canadian Beaufort Sea. Marie-Noël R. Matthews and Alexander O. MacGillivray (JASCO Appl. Sci., 202 - 32 Troop Ave., Dartmouth, NS B3B 1Z1, Canada, Marie-Noel.Matthews@jasco.com)

JASCO Applied Sciences performed acoustic modeling and measurements to calculate marine mammal exclusion zones for Chevron Canada Limited's 2012 Sirluaq 3-D seismic program in the Canadian Beaufort Sea. The Sirluaq survey was located in deep water (>650 m), on and beyond the continental slope, and presented unique challenges from both modeling and measurement standpoints. The modeling was performed with JASCO's Marine Operations Noise Model (MONM), which uses a parabolic-equation-based algorithm to accurately predict $N\times 2$-D sound propagation in ocean environments. Sound levels were measured with five calibrated Autonomous Multichannel Acoustic Recorder (AMAR) systems at distances of 50–50 000 m from the airgun array, in water depths ranging from 50 to 1500 m. The sensors were laid out to capture sound levels in both the broadside (perpendicular to survey line) and endfire (along the survey line) directions. The high-resolution digital recordings of seismic sounds were analyzed to determine peak and rms sound pressure levels (SPL), and sound exposure levels (SEL) as functions of range from the airgun array. The model estimates were generally conservative; however, the model predictions at the specific depth of the receivers accurately predicted the existence of a shadow zone and the overall transmission loss trend.

2:00

2pUWb4. Arctic ambient noise measurements in support of the Northern Watch Project. Garry J. Heard, Nicos Pelavas, Sean Pecknold (Atlantic, Defence R&D Canada, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, garry.heard@gmail.com), Carmen E. Lucas (Dartmouth, Nova Scotia, Canada), and Bruce Martin (Jasco Res., Dartmouth, NS, Canada)

During August 2012, acoustic recording systems were deployed in Barrow Strait as part of the Defence Research and Development Canada (DRDC) Northern Watch Technology Demonstration Project. Two Starfish Sensor Cubes each with a 1-m cube of seven hydrophones operating in the frequency range of 5–750 Hz, and two single-hydrophone, Autonomous Multichannel Acoustic Recorders (AMAR) providing a 30-kHz signal bandwidth were deployed. The Starfish were deployed for two one-week intervals. One AMAR was deployed for two weeks partially overlapping the Starfish deployment. The second AMAR was deployed for a period of one year with recovery planned for August 2013. The observed underwater noise picture is one of high variability ranging from an extremely quiet to a noisy environment. Noise sources included: A 500-m long iceberg grounded within 500 m of one of the Starfish; a large ice island (4–5 sq-km) that passed within 4 km of the sensors; a small number of motoring vessels; significant wind events that caused rapid and strong variations in the noise field; and a small number of marine mammal detections. After our departure, a large number of Beluga whales were observed visually. The remaining AMAR may detect these late summer visitors.

Contributed Papers

2:20

2pUWb5. Acoustic manifestations of frozen bubbles. Alexey Maksimov (Phys. of the Ocean, Pacific Oceanological Inst. Far Eastern Branch of the Russian Acad. of Sci., 43, Baltic St., Vladivostok 690041, Russian Federation, maksimov@poi.dvo.ru)

The hydrocarbon seeps emitting buoyant bubble plumes from seafloor vents have been actively investigated in different regions of the World Ocean. In winter, rising bubbles, which have reached the sea surface, freeze in ice. These clouds of the frozen bubbles are observed in Arctic seas and represent a common element of an ice cover of lakes. Deposits of gas hydrates were discovered in many marginal seas of the World Ocean. The nature of the relationship between the acoustic and physical properties of gas hydrates is still under debate for both saturated sediments and sediment containing free-gas bubbles. Acoustic manifestations of bubbles frozen in ice and gas hydrate are the subjects of the current study. On the basis of the general solution of the elastic wave scattering problem for the sphere, the scattering cross section for a bubble frozen in an ice has been found. The derived expressions coincide in the limiting cases with known results: a bubble in a rubber-like media, where the longitudinal wave speed is significantly faster than the transverse wave speed and an empty cavity in the elastic media. The structure of low-frequency resonances of bubbles clouds of the simplest geometry has been investigated.

2:40

2pUWb6. Source bearing and range estimation using an ice-mounted triaxial geophone. Stan E. Dosso (School of Earth & Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

This paper presents results from Arctic field trials to estimate the bearing and range of an acoustic source in the water column using seismic particle motion measured at a tri-axial geophone on the sea ice surface. Measurements were carried out on smooth, rough, and ridged annual ice, and on a multi-year ice floe. At each site, a hammer seismic study was carried out to selectively excite various ice seismic waves and investigate their propagation properties. Impulsive acoustic sources were deployed in the water at a variety of bearings and ranges from 0.2 to 50 km. Source bearings are estimated by applying polarization filters to suppress shear waves with transverse particle motion and computing the incident power as a function of radial look angle; the inherent 180-degree ambiguity is resolved by requiring prograde particle motion in the vertical-radial plane. Results indicate good bearing estimation (<10-degree average errors) at all ranges with little dependence on ice type. Source range is estimated from the time difference between the water-borne arrival and the critically-refracted longitudinal plate (Lp) wave. Results are limited due to strong attenuation of the Lp wave, with good range estimation to <1 km for smooth annual ice and <0.5 km for other sites.

3:00

2pUWb7. Measurements and modeling of transmission loss variability in Barrow Strait. Sean Pecknold, Nicos Pelavas, and Garry Heard (DRDC Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, sean.pecknold@drdc-rddc.gc.ca)

During the summer of 2012, a field trial was held in Barrow Strait, south of Devon Island in the Canadian Arctic. The trial included a set of acoustic transmission loss experiments recorded on Starfish Sensor Cubes, which include a 1-m cube of seven hydrophones operating in the frequency range of 5–750 Hz. The transmission loss runs consisted of 10-min and 20-min duration transmissions of 400 and 500 Hz tones made at a discrete set of distances up to 60 km from the recorders. Supporting environmental measurements included sets of CTD (conductivity-temperature-depth) profiles and bathymetric measurements. The effects of the measured environmental properties and variability are investigated via propagation modeling, and compared to the experimental data acquired during these experiments.

3:20

2pUWb8. A parabolic equation for under ice propagation. Adam M. Metzler (Environ. Sci. Lab., Appl. Res. Lab.: The Univ. of Texas at Austin, ARL-Environ. Sci. Group, P.O. Box 8029, Austin, TX 78713, ametzler@arlut.utexas.edu), Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, Texas)

Parabolic equation methods are useful to accurately and efficiently model propagation in the ocean for range-dependent environments. Most methods treat environments where the ocean surface is a flat perfect reflector. For some problems, the ocean surface characterized as a random rough scattering water-air boundary produces significant effects on the overall propagation. A rough surface parabolic equation has been developed [A. P. Rosenberg, J. Acoust. Soc. Am. **105**, 144 (1999)] that extends the split-step Padé approach in the RAM implementation, and others have extended split-step Fourier methods. For an upper ice surface, elasticity should be incorporated into the rough scattering boundary. In this paper, a parabolic equation is presented that captures effects of seismo-acoustic interactions and scattering from a rough surface modeled as rigid as opposed to pressure release. Particular attention is given to upward-refracting sound speed profiles, typically found in Arctic environments, which encourage interactions with an ice surface. The scattering technique can be extended to environments with fluid/ice interfaces. [Work supported by ARL:IR&D.]

3:40

2pUWb9. Effective ice model for under-ice propagation using the fluid-fluid parabolic equation. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com), Richard L. Campbell (OASIS Inc., Seattle, Washington), and Lee Freitag (Woods Hole Oceanogr. Inst., Woods Hole, Washington)

An approach is presented to permit efficient computation of the waterborne acoustic field in the presence of sea ice. The range-dependent wide-angle parabolic equation (PE) is used to model the acoustic field with a hard, lower-density layer (ice) placed above the ocean and seafloor. The ice layer is characterized by its thickness, compressional speed, density, and attenuation. Acoustic loss due to sea ice is primarily driven by conversion to shear waves, and in this model the effect will be approximated by attenuation within the ice layer. Rough interface scattering at the air-ice and ice-water interfaces will be handled by generating range-dependent realizations from a data-derived ice thickness model. An inversion for ice parameters is conducted using the work of Jin *et al*. [J. Acoust. Soc. Am. **96**(5)] and by matching the transmission loss of Diachok [J. Acoust. Soc. Am. **59**(5)]. Model-data comparisons between this three-layer PE and measurements the taken in the Fram Strait in 2010 and in the Canada Basin in 2011 will be presented.

TUESDAY EVENING, 4 JUNE 2013 — 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Tuesday are as follows:

Architectural Acoustics	513abc
Engineering Acoustics	512ae
Musical Acoustics	512dh
Physical Acoustics	519b
Psychological and Physiological Acoustics	514abc
Structural Acoustics and Vibration	512cg