11:20

1aUW8. Evidence for a common scale O(0.1) m that controls seabed scattering and reverberation in shallow water regions. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16801, holland-cw@psu.edu)

Analysis of the spectral content of long-range reverberation yields two observations. First, there is a remarkably similar scale, O(0.1)m, between three diverse continental shelf regions. This is surprising given general understanding of the complexity and diversity of geologic processes. Second, there is strong evidence that the scale is associated with heterogeneities within the sediment. Thus, sediment volume scattering, not interface scattering, controls long-range reverberation from a few hundred Hertz to several kilohertz. This is also unexpected given that at long-ranges the vertical grazing angles are less than the critical angle, and hence, the penetration of the acoustic field into the sub-bottom is expected to be modest. The consistency of the scale, O(0.1)m, suggests an underlying feature or mechanism that is consistent across many ostensibly diverse geological settings. Neither the feature nor mechanism is known at this time. Several hypotheses will be presented. [Work supported by ONR Ocean Acoustics.]

11:40

1aUW9. Rayleigh scattering of sound by spherically symmetric bodies. 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)

An obstacle's shape is often approximated by a sphere in analyses of sound scattering by air bubbles, objects on or near the seafloor, marine organisms, clouds of suspended particles, etc. Here, an asymptotic technique is developed to study low-frequency sound scattering from spherically symmetric inhomogeneous obstacles. The obstacle can be fluid, solid, or a fluid-filled solid shell. Physical properties of the obstacle are arbitrary piece-wise continuous functions of the distance to its center. The radius of the obstacle is assumed to be small compared to the wavelengths of sound in the surrounding fluid as well as of compressional and shear waves inside the obstacle. General properties of the sound scattering by spherically symmetric bodies are established. Resonant Rayleigh scattering is studied in detail. For plane and spherical incident waves, it is discussed which physical and geometrical parameters of the obstacle can be retrieved from the scattered acoustic field.

MONDAY AFTERNOON, 3 JUNE 2013 513ABC, 12:55 P.M. TO 4:20 P.M.

Session 1pAAa

Architectural Acoustics and Signal Processing in Acoustics: Advanced Analysis of Room Acoustics: Looking Beyond ISO 3382 II

Boaz Rafaely, Cochair
Dept. of Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva 84105, Israel

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

Michael Vorländer, Cochair
ITA, RWTH Aachen Univ., Neustr. 50, Aachen 52066, Germany

Chair's Introduction—12:55

Invited Papers

1:00

1pAAa1. Theoretic considerations on how the directivity of a sound source influences the measured impulse response. Ingo B. Witew, Tobias Knüttel, and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany, ingo.witew@akustik.rwth-aachen.de)

In previous investigations, it has been shown that the directivity of a measurement sound source has a significant influence on the measured room impulse response (RIR). Using a specialized method of analysis, the sources influence can be identified even in the very late part of the RIR even in very reverberant environments. These results seem to be surprising at first and contradict intuitive expectations. In this contribution, the findings are briefly discussed, and the congruence with general room acoustic theory is revised and discussed.

1:20

1pAAa2. Enhanced spatial analysis of room acoustics using acoustic multiple-input multiple-output systems. Hai Morgenstern and Boaz Rafaely (Ben-Gurion Univ. of the Negev, Ben-Gurion Univ. of the Negev, Beer-Sheva 84105, Israel, hai.morgenstern@gmail.com)

Standard acoustic measurements in enclosures typically employ single-input single-output (SISO) acoustic systems. The parameters obtained from these measurements describe features of energy decay and do not characterize spatial attributes of the enclosure. Directional analysis of enclosures became popular with the introduction of microphone and loudspeaker arrays. In particular, spherical arrays have been shown to be highly beneficial for spatial analysis. Spherical microphone arrays facilitate the estimation of the arrival direction of the direct and reflected sound, while the use of both loudspeaker and microphones arrays can support the estimation of both radiation and arrival directions, with the application of conventional beamforming methods. However, when several reflections are attributed to the same time bin in a discrete impulse response, reflection paths may not be uniquely determined by existing beamforming techniques. We present a new method to uniquely determine source and receiver directions for multiple reflections when time separation is unfeasible. The paper presents the formulation of the proposed method, also showing a simulation study to demonstrate the performance of the proposed method.

1:40

1pAAa3. Spatio-temporal energy measurements in renowned concert halls with a loudspeaker orchestra. Sakari Tervo, Jukka Pätynen, and Tapio Lokki (Media Technology, Aalto Univ., P.O. Box 15500, Aalto FI-00076, Finland, Tapio.Lokki@aalto.fi)

Room acoustical parameters are commonly measured as spatial averages from a few source positions to several listening positions. In addition, ISO3382-1:2009 suggests that subjective listener aspect could be predicted with a few parameters at mid frequencies. It is obvious that such recommendation is inadequate to describe all perceptual differences between concert halls or seating positions. Explaining multidimensional subjective impressions is not possibly with a few averaged numbers. In our opinion, to compare accurately between concert halls, the measurement positions should be exactly at the same distance from the sources in each hall. And sources should be numerous to represent the typical real source on a wide area—a symphony orchestra. In this paper, we compare renowned European concert halls with spatio-temporal visualization of sound energy. Measurements at identical distances in halls enable an accurate comparison of sound energy distributions. The analysis of measured spatial impulse responses is performed with spatial decomposition method (SDM). The spatial sound energy distribution is presented at different frequency bands as a function of time to visualize the cumulative energy before and after 100 ms.

2:00

1pAAa4. Three-dimensional spatial analysis of concert and recital halls with a spherical microphone array. Samuel Clapp (Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, clapps@rpi.edu), Anne Guthrie (Arup Acoust., New York, NY), Jonas Braasch, and Ning Xiang (Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

The most well-known acoustical parameters—including reverberation time, early decay time, clarity, and lateral fraction—are measured using data obtained from omnidirectional or figure-of-eight microphones, as specified in ISO 3382. Employing a multi-channel receiver in place of these conventional receivers can yield new spatial information about the acoustical qualities of rooms, as well as the potential for new parameters that could have greater predictive power in terms of listeners' subjective preferences. In this research, a spherical microphone array was used to measure the room impulse responses of a number of different concert and recital halls. The data were analyzed using spherical harmonic beamforming techniques, along with other direction of arrival estimation algorithms, to understand how the soundfield evolves spatially over time at different points in the room. The results were compared to geometrical acoustic simulations and used to differentiate between listener positions which exhibited similar values for the standard parameters. In addition, new parameters were examined, including soundfield homogeneity and other spatial ratios.

2:20

1pAAa5. Listening space measurements resolved by direction and spectro-temporally. Adam O'Donovan, Dmitry N. Zotkin (Comput. Sci. and UMIACS, Univ. of Maryland, College Park, MD), and Ramani Duraiswami (VisiSonics Corp., A.V. WIlliams Bldg., #115, College Park, MD 20742, ramani@umiacs.umd.edu)

Measurement of a listening space might be done to characterize it in a gross way, or to identify some deficiencies in the space, which can then be corrected. Alternately, these measurements might be performed to create inputs to an auralization software, which might seek to recreate a virtual listening experience. We propose that it might be possible to perform audio-visual measurements of a listening space that allow the entire response of the listening space to be understood and visualized. The goal is to understand the response completely in terms of its components: along direction, frequency, early and late characteristics, and finally at the level of the structural elements of the scattering surfaces. Measurements with spherical arrays of cameras and microphones provide measurements that allow the response to be decomposed in the desired fashion. Post-processing software that allows the measurements to be analyzed instantly following the measurement will also be described and demonstrated.

2:40–3:00 Break

3:00

1pAAa6. Large-scale multiple input/multiple output system identification in room acoustics. Martin Schneider and Walter Kellermann (Multimedia and Signal Processing, Univ. Erlangen Nürnberg, Cauerstr.7, Erlangen, Germany, schneider@LNT.de)

In audio reproduction scenarios, room acoustics may be described as a MIMO system response from multiple loudspeaker to multiple microphones in the listening space. This system response may, e.g., be used for an equalization of the listening room and must be identified from observing the available loudspeaker and microphone signals in real-world systems. For few transducers this task is mostly solved, but massive multichannel reproduction with dozens to hundreds of loudspeakers left many research questions open. This contribution points out the fundamental challenges, previous solutions and recent advances. As a key issue, the so-called nonuniqueness problem for MIMO system identification by adaptive filtering will be discussed along with decorrelation schemes for the loudspeaker signals to alleviate this problem. Successful adaptation algorithms suitable for these scenarios imply considerable computational demands and require additional measures to ensure robustness. Recently emerging system models in spatial transform domains allow for approximative models and seem to be promising for robust real-time implementations.

3:20

1pAAa7. Robustness analysis of room equalization for soundfield reproduction within a region. Dumidu S. Talagala, Wen Zhang, and Thushara D. Abhayapala (Res. School of Eng., College of Eng. and Comput. Sci., Australian National Univ., Canberra, ACT 0200, Australia, thushara.abhayapala@anu.edu.au)

Recent works on soundfield reproduction have presented several methods of recreating a desired soundfield within a region. Estimation or prior knowledge of the inverse reverberant channels now becomes an essential element of equalizing the room effects. However, it has been shown that designing point-to-point equalizers by sampling the reverberant soundfield is only practical within a few tenths of

a wavelength of the sampled locations. This work investigates the robustness of the equalization process applied to a region, with respect to changing of actual microphone positions from their expected locations. We use a modal description of the equalized soundfield to obtain theoretical results for region equalization error due to positioning errors. Simulation results suggest that equalizing the reverberant soundfield recorded at multiple positions around the edge of the reproduction region is more immune to the positioning errors.

Contributed Papers

3:40

1pAAa8. A numerical and experimental validation of the room acoustics diffusion theory inside long rooms. Chiara Visentin, Nicola Prodi (Dipartimento di Ingegneria, Università di Ferrara, via Saragat 1, Ferrara 44122, Italy, chiara.visentin@unife.it), Vincent Valeau (Institut PPRIME, CNRS-Université de Poitiers-ENSMA, Poitiers, France), and Judicäel Picaut (IFSTTAR, LUNAM Université, Bouguenais, France)

The paper focuses on the validation of the recently proposed room-acoustics diffusion theory by means of numerical simulations and experimental measurements. The analysis aims to verify the equation underlying the theory (Fick's law of diffusion) which relates the energy density gradient and the sound intensity inside a room through a constant diffusion coefficient. In this work, the acoustic quantities are numerically/experimentally derived under stationary conditions, and their ratio is employed to estimate the effective value of the diffusion coefficient inside long rooms. The numerical study was carried out with particle-tracing simulations. The measurements were performed with a Microflown® three-dimensional sound intensity probe inside a 1:16 scale model of a long room, varying the absorption and the scattering properties at the boundaries. A comparison between numerical and experimental results is carried out with a least-square algorithm, showing a fair agreement between the diffusion coefficients estimated with the two methods. The results lead to the conclusion that the reverberant sound field inside long rooms can be described by a non-homogeneous diffusion process: the local diffusion coefficient is not a constant inside the room but increases with the distance from the source and depends on the acoustical properties of the room boundaries.

4:00

1pAAa9. Energy evolution in enclosure geometries as exhibited by a finite difference time domain method. Zackery Belanger (53 3rd St, Troy, New York 12180, zb@archgeometer.com)

Measurements and simulations conducted for the purpose of extracting or constructing impulse responses are inextricably dependent on a limited number of receiver locations. Wave-based simulations offer the opportunity to discard this dependency and assess an entire evolving sound field. In this work, a finite difference time domain method is implemented to simulate an evolving sound field in a range of enclosure geometries with reflective boundaries. The distribution of energy is monitored statistically as mixing and ergodic states are approached, and evidence is presented for the predictability of this evolution based on room geometry alone. A system for importing geometries from the Rhinoceros 3D CAD software, using the Grasshopper parametric environment, is also presented. A form known as the Barnett Billiard Table in mathematics is included in this study, which exhibits exceptional behavior even without the presence of diffusive treatment.

MONDAY AFTERNOON, 3 JUNE 2013 — 513DEF, 12:55 P.M. TO 3:20 P.M.

Session 1pAAb

Architectural Acoustics and Musical Acoustics: Vibration in Music Performance

Clemeth Abercrombie, Cochair
Artec Consultants Inc., 114 W 26th St., 10th Fl., New York, NY 10001

M. Ercan Altinsoy, Cochair
Chair of Commun. Acoust., Dresden Univ. of Technol., Helmholtzstr. 18, Dresden 01062, Germany

Chair's Introduction—12:55

Invited Papers

1:00

1pAAb1. Perceptual evaluation of violin vibrations and audio-tactile interaction. M. Ercan Altinsoy, Sebastian Merchel, and Sebastian Tilsch (Chair of Commun. Acoust., Dresden Univ. of Technol., Helmholtzstr. 18, Dresden 01062, Germany, ercan.altinsoy@tu-dresden.de)

When playing a violin, the musician communicates with his instrument not only through his ears but also his fingers, chin, shoulder, and eyes. He uses different sensory inputs, which are provided by different sensory channels, such as auditory, tactile, kinesthetic, and visual, to play his musical instrument. The perceived vibrations are useful for the player to feel and to control the instrument. The interaction between sound and vibration plays also a role on the overall instrument perception. In this study, violin vibrations and their interaction with violin sounds were evaluated. Therefore, the vibration amplitudes of the neck and the violin sounds were recorded simultaneously during normal playing. The vibration recordings were analyzed, and then additional stimuli were generated by filtering or modifying frequency components. In the first experimental session, the vibration stimuli, which were presented to the subjects via a mini electrodynamic shaker, were evaluated. In the second experimental session, an investigation with multimodal (auditory-haptic) stimuli was conducted. The results show the importance of vibrations on the overall perception of the instrument and provide information on useful vibration features for the player-instrument interaction.

1:20

1pAAb2. Telehaptic interfaces for interpersonal communication within a music ensemble. Jonas Braasch, Pauline Oliveros, and Doug Van Nort (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Visual communication is an important aspect of music performance, for example, to pick up temporal cues and find the right entries. Visual cues can also be instrumental to negotiate the solo order in improvized music or enable social exchange, for example, by signaling someone that her solo was well received. The problem with visual communication is that one has to catch someone else's attention, and visual cues outside someone's visual field cannot be detected, even more so if the addressee is busy reading a music score or closing his eyes in a Free Music session. Acoustic communication does not encounter these challenges, but of course someone does not want to disturb the music with other acoustic signals. The haptic modality has the advantage that it does not necessarily interfere with the acoustic signal and does not require attention. However, it allows interpersonal communication if both parties are within close proximity. Using telematic interfaces solves the problem of proximity by allowing participants to communicate over any physical distance. In the project presented here, haptic interfaces were explored in connection with an intelligent music system, CAIRA, to examine both the effect of human/machine and inter-human communication. [Work supported by the National Science Foundation, No. 1002851.]

1:40

1pAAb3. Stage floor vibrations and bass sound in concert halls. Anders Askenfelt (Dept. of Speech, Music and Hearing, KTH Royal Inst. of Technol., Lindstedtsvägen 24, Stockholm 10044, Sweden, andersa@csc.kth.se) and Knut Guettler (Norwegian State Acad. of Music, Jar, Norway)

The double bass and cello sections in the orchestra transmit vibrations to the stage floor through the end pins. Whether or not these vibrations may contribute to the perceived sound in the hall has been investigated since the 1930s. In this study, the conditions for an efficient transfer of instrument vibrations to the floor, as well as the radiation from the floor to the audience area, are investigated. The study includes measurements of the impedance matching between bass and stage floor, the vibration velocity transfer to the floor via the endpin, and radiation from point-driven bending waves in the stage floor well below the coincidence frequency. The impedance conditions and radiation properties for the stage floors of five concert halls were investigated. In the most promising hall, a full-scale experiment was run with an artificially excited double bass supported via the end pin on the stage floor, and on a concrete support below, respectively. The contribution from the stage floor radiation to the sound level in the audience area was 5 dB or more between 30 and 60 Hz. This range covers the fundamental frequencies over one octave starting from the lowest note (B0) of a five-string bass.

2:00

1pAAb4. Recent experiences with vibration of stage and audience floors in concert halls. Thomas Wulfrank (Kahle Acoust., Ave. Moliere 188, Brussels 1050, Belgium, thomas.wulfrank@kahle.be), Igor Lyon-Caen (Alstom Transport TGS, Saint-Ouen, France), Yann Jurkiewicz, Johan Brulez, and Eckhard Kahle (Kahle Acoust., Brussels, Belgium)

Vibration of (wood) surfaces plays a significant role in concert hall acoustics, as confirmed by musicians and music lovers. Many acoustic engineers, on the other hand, tend to have strong reservations against vibrating surfaces, and usually try to minimize surface vibration in order to maximize RT and airborne strength (G) at bass frequencies. This has led to a generally accepted preference for massive and stiff surface constructions in new halls. Problems have been known to occur when this general guideline was also applied to the design of wooden floors, in particular stage floors. Despite some good scientific research in this field, a big gap still remains between the vibro-acoustic behavior of wooden floors and subjective preferences of musicians and audiences. This paper further explores the role of vibrations in concert hall design, and the need for balancing surface reflectivity versus vibration transmission. Recent experiences, including the new Konserthus in Stavanger and the renovation of the Bolshoi Hall of the Moscow Conservatory, will be described as well as vibration measurements carried out on a number of existing stage floors. Some implications for the design of wooden floor constructions will be discussed.

Contributed Paper

2:20

1pAAb5. Auditory-tactile music perception. Sebastian Merchel and M. Ercan Altinsoy (Chair of Commun. Acoust., TU Dresden, Helmholzstr. 18, Dresden 01069, Germany, sebastian.merchel@tu-dresden.de)

The coupled perception of sound and vibration is a well-known phenomenon during live pop or organ concerts. However, even during a symphonic concert in a classical hall, sound can excite perceivable vibrations at the body surface. However, the concert visitor might not be aware of those vibrations, because the tactile percept is integrated with the other senses into one multi-modal percept. This article discusses the influence of whole-body vibrations on the listener experience during the reproduction of concerts recordings. Four sequences were selected from classical and modern music, which include low frequency content (e.g., organ, kettledrum, contrabass). A stimulus length of 1.5 min was chosen in order to provide enough time for habituation. The audio signal was reproduced using a surround setup. Additional seat vibrations have been generated from the audio signal. Test participants were asked to rate the overall quality of the concert experience. The results show that vibrations have a significant influence on our perception of music. This finding is interesting in the context of audio reproduction, but also for the construction of concert venues.

2:40–3:20 Panel Discussion

Session 1pAB

Animal Bioacoustics: Sound Generation and Perception in Animals

Nancy Allen, Chair
Defence R&D Canada - Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada

Contributed Papers

1:00

1pAB1. Acoustic communication in crocodiles: How do juvenile calls code information? Nicolas Mathevon, Amélie Vergne (ENES Lab CNRS UMR8195, Univ. of Lyon/Saint-Etienne, 23 rue Michelon, Saint-Etienne 42023 cedex 2, France, mathevon@univ-st-etienne.fr), and Thierry Aubin (Equipe Communications Animales CNPS CNRS URM 8195, Univ. Paris XI-Orsay, Orsay, France)

In spite of its importance for the understanding of the evolution of sound communication, information concerning the vocal world of crocodilians is limited. Experimental works have brought evidence of the biological roles of juvenile sound signals, with "hatching calls" eliciting care by the mother and synchronizing clutch hatching, "contact calls" gathering groups of juveniles, and "distress calls" inducing maternal protection. Recently, we investigated the question of species-specific information coding within juvenile calls. The analysis of signal acoustic structure shows inter-specific differences between calls. However, using playback experiments, we bring the evidence that these differences are not relevant to animals, either juveniles or adults. By using calls modified in the temporal and the frequency domains, we isolate the acoustic cues necessary to elicit a behavioral response from receivers, underlying the importance of the frequency modulation slope. Considering previous results underlying the absence of information about individual identity in juvenile calls, we make the hypothesis that these signals basically support a "crocodilian" identity.

1:20

1pAB2. Design and field test of a low-cost-portable linear array for marine mammal localization. Omar A. Bustamante (ESIME, Austral #77 Col. Atlanta, Cuautitlan Izcalli 54740, Mexico, omarb.p@hotmail.com), Eduardo Romero Vivas (CIBNOR, La Paz, BCS, Mexico), and Sergio Beristain (ESIME, IPN,IMA, Mexico, D. F., Mexico)

Marine mammals are reliable bioindicators of aquatic systems health. Within this group, cetaceans are well known by their high dependence on sound for many of their vital activities such as socialization and mating, prey catching, and navigation. Due to its high dependence on sound, bioacoustic methods become very important for the study of these species. Acoustic monitoring in the field is usually performed using omnidirectional hydrophones to assess the presence of mammals, but for some behavioral studies, it is also important to locate the animals, something which is not possible with that arrangement. Although there are very well known techniques to detect the direction of arrival of the sound, the equipment required is highly specialized and expensive. In this paper, the design and field test of digital and analog versions of a portable linear array of hydrophones capable of locating animal sounds by beamforming, using low cost and easily available equipment is presented. The array was tested in La Paz bay, Mexico, by experts of the Marine Mammals Research Program of the University of Baja California Sur, which were able to locate dolphins (*Tursiops truncatus*) only by their sound, despite strong sources of noise in the area.

1:40

1pAB3. Auditory reaction time measurements and equal-latency curves in the California sea lion (*Zalophus californianus*) and bottlenose dolphin (*Tursiops truncatus*). Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmpfoundation.org) and James J. Finneran (U.S. Navy Marine Mammal Program, SSC Pacific, San Diego, CA)

Subjective equal-loudness contours are used to create weighting functions for human noise-mitigation criteria. Comparable direct measurements of subjective loudness with animal subjects are, however, difficult to conduct. Using methods similar to those used in previous mammalian studies, this study estimated subjective loudness through the measurement of response time (RT) in an auditory signal-detection task. Measurements were conducted in a sound-attenuating hut with a California sea lion and under water in a quiet pool with a bottlenose dolphin. Tonal stimuli were presented at supra- and near-threshold sound pressure levels (SPLs) using a method of constants. Median RT increased with decreasing SPL for both species across all tested frequencies. A Piéron function, which models RT as a function of SPL, was fitted to the RT-SPL curves in a nonlinear fashion. Equal-latency curves were based on the Piéron functions at each frequency. Preliminary results for the sea lion suggest that the equal-latency curves are similar to the audiogram at longer median RTs (~300 ms), with increasing deviation from the audiogram at the fastest median RTs (~200 ms). Continued testing with additional subjects will provide further data for designing marine mammal auditory weighting functions. [Funded by U.S. Navy Living Marine Resources Program.]

2:00

1pAB4. Methods for determining free-swimming positioning and echolocation beam patterns. Danielle Greenhow (College of Marine Sci., Univ. of South Florida, 140 7th Ave. S, St. Petersburg, FL 33701, dgreenho@mail.usf.edu), Heidi Harley (Div. of Social Sci., New College, Sarasota, FL), Wendi Fellner (The Seas, Epcot, Animal Programs, Walt Disney World Resort, Lake Buena Vista, FL), Adrienne Cardwell (Mote Marine Lab., Sarasota, FL), and David Mann (College of Marine Sci., Univ. of South Florida, Sarasota, FL)

Echolocation beam patterns were recorded using a 25-element autonomous, self-contained hydrophone array during free-swimming echoic match tasks. Two GoPro HERO2 cameras mounted at 45-degree angles to the plane of the 4.5×4.5 ft PVC array recorded the position of the dolphin during approach and investigation of the sample object presented in front of the array. Using a pre-session calibration file and MaxTRAQ3D software, analysis of video determined exact positioning of the head region during echolocation, including distance from the array and angle to the plane of the array. An LED and ITC-1042 transducer, operated by an Arduino Uno processor board and a Hafler amplifier, were mounted on the array. Simultaneously emitted visual pulses (LED) and acoustic pulse trains were used for synchronization of the acoustic recorders and video cameras. The distinct pulse train was used during post-recording analysis to synchronize the hydrophones using a custom-designed MATLAB routine. The hydrophones are sampled at 400 kHz and analyzed in real-time. The maximum and minimum values and time stamps within a 0.32 ms window are stored continuously to reduce the amount of data that needs to be stored. Using the 3-D positioning of the animal, dynamic beam patterns can be reconstructed.

2:20

1pAB5. Threshold of hearing for swimming Bluefin tuna (*Thunnus orientalis*). Arthur N. Popper (Dept. of Biol., Univ. of Maryland, College Park, MD, apopper@umd.edu), Jonathan Dale (Tuna Res. and Conservation Ctr., Hopkins Marine Station, Stanford Univ., Pacific Grove, CA), Michael D. Gray (G.W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), William Keith (Naval Undersea Warfare Ctr. Div. Newport, Newport, RI), Barbara A. Block (Tuna Res. and Conservation Ctr., Hopkins Marine Station, Stanford Univ., Pacific Grove, CA), and Peter H. Rogers (G.W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Hearing thresholds for three pairs of 1 m long Pacific bluefin tuna (*Thunnus orientalis*) were measured utilizing operant conditioning procedure with a food reward and a staircase psychophysical technique. Fish, swimming at 1–4 m/s, quickly learned to approach the feed when they heard a sound. Measurements were made at the Tuna Research and Conservation Center (Stanford University) in a 9.14 m diameter, 1.65 m deep indoor cylindrical tank. The acoustic stimulus was produced by radially oriented piezoelectric line sources centered at the bottom of the tank, which produced a circumferentially uniform sound field. The acoustics of the tank was thoroughly characterized for both acoustic pressure and particle motion using hydrophones and two neutrally buoyant accelerometers with response axes oriented in the radial and vertical directions. Thresholds, expressed in terms of pressure and particle acceleration, were obtained at six sinusoidal frequencies between 325 Hz and 800 Hz, a range that was limited by source and tank acoustics. The lowest mean threshold for the three fish pairs, expressed in terms of acoustic pressure, was 83 dB re 1 μPa at 500 Hz. [Work supported in part by ONR/CNR Challenge Grant: "Mitigation of flow noise effects by fish."]

2:40

1pAB6. Prey pursuit strategy of Japanese horseshoe bats, *Rhinolophus ferrumequinum Nippon*, during target selection task. Yuki Kinoshita, Daiki Ogata (Faculty of Life and Med. Sci., Doshisha Univ., 1-3 Miyakotani Tatara, Kyotanabe 610-0321, Japan, dmm1014@mail4.doshisha.ac.jp), Ikkyu Aihara (Brain Sci. Inst., RIKEN, Wako, Japan), Yoshiaki Watanabe, Hiroshi Riquimaroux, Tetsuo Ohta, and Shizuko Hiryu (Faculty of Life and Med. Sci., Doshisha Univ., Kyotanabe, Japan)

We investigated prey pursuit behavior of Japanese horseshoe bats, while they were tasked to make a choice between two tethered fluttering moths during flight. Echolocation pulses were recorded by a telemetry microphone mounted on the bat, combined with a 17-ch horizontal microphone array to measure pulse directions. Flight paths of the bat and moths were monitored by using two high-speed video cameras. Acoustical measurements of CF echoes from fluttering moths (67 kHz : CF2 frequency) was conducted using an ultrasonic loudspeaker, turning the head direction of the moth to the loudspeaker from 0° to 180° in the horizontal plane. Amount of acoustical glints caused by moth fluttering varied with the sound direction, showing the maximum between 70° and 100°. In the flight experiment, moths chosen by the bat fluttered within or moved across these angles to the bat's pulse direction, which would cause dynamic changes in frequency and amplitude of acoustical glints during flight. This result suggests that dynamic changes in acoustical glints appear to attract the bats for prey selection. Furthermore, mathematical modeling implied that the bats possibly took the optimum flight path for capturing a target, which the bat selected based on the acoustical cues in the echoes.

3:00

1pAB7. The alignment problem for bat biosonar beampatterns. Philip Caspers (Dept. of Mech. Eng., Virginia Tech, 1110 Washington St., Blacksburg, VA 24061, pcaspers@vt.edu), Rongjiang Pan (School of Comput. Sci. & Technol., Shandong Univ., Jinan, China), Alexander Leonessa, and Rolf Müller (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA)

Unlike the beampatterns of technical acoustical systems, the biosonar beampatterns of bats are highly variable in the shapes of their main- and sidelobes over frequency. Some of this variability could represent adaptations to different sensing tasks. In order to understand such possible adaptations, a quantitative method for the analysis of variability (e.g., principal component analysis) is needed. Since the orientation of biosonar beampatterns is highly variable *in-vivo*, e.g., due to ear/head movements, and not preserved in isolated noseleaf/ear samples, orientation is left out of the initial analysis. Instead, beampatterns should be aligned to characterize their orientation-independent features. For this purpose, a framework to characterize the beampattern alignment problem and perform the alignment has been drawn up. For each frequency, beampatterns are compared using a distance metric (e.g., a p-norm). By investigating the value of this distance metric over the space of all possible beampattern rotations, it is possible to gain insights into the alignment problem, e.g., with regard to the existence of multiple minima in the metric. This space can also be used to test alignment strategies across multiple frequencies, e.g., through a weighted sum of the respective distances.

3:20

1pAB8. Effects of the source location on numerical biosonar beampattern predictions for bat noseleaves. Anupam Kumar Gupta and Rolf Müller (Dept. of Mech. Eng., Virginia Tech, 1110 Washington St., Blacksburg, VA 24061, anupamkg@vt.edu)

Approximately 300 bat species are known to emit their ultrasonic biosonar pulses through the nostrils. In these animals, ultrasound is produced by the larynx, propagates along the vocal tract, exits through the nostrils, and is finally diffracted by intricately shaped baffle structures known as "noseleaves." Noseleaf geometry determines diffraction and hence the spatial distribution of the emitted ultrasound. As a consequence, numerical predictions of the noseleaves' acoustic function can be made based on the digital models of noseleaf shape. To limit model size and computational effort associated with numerical beampattern predictions, the vocal tract is often only partially included in these models or left out completely. In order to investigate the effect of source placement within a complete or partial vocal tract attached to a noseleaf shape on the numerical beampattern prediction, the noseleaf of the Great Roundleaf Bat (*Hipposideros armiger*) was studied as a model. Numerical beampattern estimates were obtained for a single monopole source positioned near the vocal folds or closer to the nostrils. Two monopoles sources placed in each nostril were also investigated. It was found that source positioning could impact the beampattern whenever they broke the symmetry in the near-fields of the two nostrils.

3:40

1pAB9. Investigation of acoustic gaze strategy by *Pipistrellus abramus* and *Rhinolophus ferrumequinum Nippon* during obstacle avoidance flight. Yasufumi Yamada, Arie Oka, Shizuko Hiryu, Tetsuo Ohta, Hiroshi Riquimaroux, and Yoshiaki Watanabe (Faculty of life and Med. Sci., Doshisha Univ., 1-3, 1-3, Miyako-tani, Tatara, Kyotanabe,Kyoto 610-0394, Japan, dml0142@mail4.doshisha.ac.jp)

We investigated the acoustic gaze (angle between pulse and flight directions) and the beam width of echolocation pulses emitted by bats during obstacle avoidance flight in a chamber (7×3×2 m). Echolocation pulses were recorded by a telemetry microphone mounted on the bat's back and a horizontal 20-ch microphone array set up in the chamber. Flight path measurements were conducted using two high-speed video cameras. While the bat showed a circular flight avoiding the surrounding walls, the acoustic gaze showed significant linear correlation to the angular velocity of the flight. This means that the bat adjusted the pulse direction to precede its own flight direction. The correlation coefficient increased with complexity of obstacle conditions. We compared changes in acoustic gaze between FM bats (mean beam width: ±50°) and CF-FM bats (mean beam width: ±22°). We found that FM bats smoothly shifted the acoustic gaze within 10° during the flight whereas the CF-FM bats frequently shifted the acoustic gaze within 25°. These results indicate that the shifting acoustic gaze by CF-FM bats compensates their own narrow beam width. Both bat species may keep approximately ±50° of their echolocation sights in order to sense the space for moving forward safely.

4:00

1pAB10. Buckling as a source of sound, with application to the modeling of cicada sound generation. Allan D. Pierce (Woods Hole Oceanogr. Inst., P.O. Box 339, East Sandwich, MA 02537, adp@bu.edu), Derke Hughes (Naval Undersea Warfare Ctr., Newport, RI), Kossi Edoh (North Carolina A&T Univ., Greensboro, NC), Richard A. Katz, and Robert M. Koch (Naval Undersea Warfare Ctr., Newport, RI)

A basic model of a ribbed finite plate is first considered, with the plate connected to a parallel surface by a nonlinear spring. When individual ribs are placed under compression, the linearized version of the model predicts eventual exponential growth of the transverse displacement when the compressional load exceeds the buckling load. The nonlinear spring, however, stops this growth and a subsequent oscillation ensues. The anatomy of the cicada is, of course, much more complicated, and the basic model is extended to give a mathematical formulation of the model proposed by Bennet-Clark and Daws (J. Exper. Biol. 1999) for the *Cyclochila australasiae* (a relatively large species of cicada), with explicit elasto-mechanical parameters, a principal objective being the computational prediction of the observed far-field waveforms. Possible explanations are advanced for the anatomical cause of the nonlinear springs. Further extensions of the theory are applied to the development of mathematical models for species of cicadas (considerably smaller) commonly found in the United States.

4:20

1pAB11. Nature of nonlinear mechanisms in the generation and propagation of sound in the cicada mating call. Derke Hughes (NUWC-DIVNPT, 1176 Howell St., Newport, RI 02841, derke.hughes@verizon.net), Kossi Edoh (North Carolina A&T State Univ., Greensboro, NC), and Allan Pierce (Woods Hole Oceanogr. Inst., Thalmouth, MA)

Experiments and analyses (Hughes *et al*., J. Acoust. Soc. Am. 2009) on the relationship of body surface displacements with external acoustic pressure measurements while cicadas are generating mating calls resulted in the conclusion that the relationship is substantially nonlinear. The present analysis assesses whether the propagation through the air is nonlinear propagation. Suspicion that such might be the case is suggested by the fact that the sound levels at distances of the order of several meters from a sounding cicada are as high as 90 dB at distances of the order of several meters. Computational results are reported in the present paper that indicate that nonlinear effects are not important for propagation of typical cicada frequencies of 10 kHz over a radial distance interval beginning 2 cm and ending at 10 m. A suggested explanation is based on the observation that the displacement of the body surface varies with position over the surface. The time history of displacements at different points is not as repetitive during successive tone burst generations as is the total volume displacement of the surface. This is consistent with the observation that buckling of the long ribs in the tymbals is the ultimate cause of the surface displacement vibrations.

4:40

1pAB12. Giraffe Helmholtz resonance. Elizabeth vonMuggenthaler and Meredith Bashaw (Fauna Commun. Res. Inst., 818 Lake Orange Rd., Hillsborough, NC 27278, L@animalvoice.com)

Many animal species, including elephants and okapi, use sounds above and below the range of human hearing to communicate. A longitudinal study presented here suggests giraffe produce infrasonic vocalizations using Helmholtz resonance. Recordings were made of giraffe (*Giraffe camelopardalis*) in controlled indoor conditions and naturalistic outdoor conditions. The portable recording and analysis system consisted of a trigger oscilloscope; DAT recorders; Nagra IV-SJ; and computers. Each signal was low-pass/high-pass filtered, and FFT and STFT were performed using PolynesiaTM real-time scrolling analysis. In controlled recordings and in naturalistic situations, two types of signals were identified: audible bursts [(11 Hz (75 dB +/− 3) to 10,500 Hz (80 dB +/− 3)] dominant frequencies between 150 and 200 Hz and covert vocalizations [(14 Hz (60 dB +/− 3) to 250–275 Hz (30 dB +/− 3)] dominant frequencies between 20 and 40 Hz. Both audible and covert signals coincided with neck throw or head toss behaviors. The shape of the giraffe's respiratory apparatus during this behavior and the frequencies produced implicate Helmholtz resonance as a production mechanism. In naturalistic recordings, two of the five infrasonic vocalizations identified were produced during close range social interactions, suggesting that giraffe use these vocalizations to communicate with con-specifics. The social functions, air and seismic transmission mechanisms of these vocalizations should be further assessed.

MONDAY AFTERNOON, 3 JUNE 2013 518C, 1:35 P.M. TO 3:40 P.M.

Session 1pBAa

Biomedical Acoustics: Acoustic Microscopy: Biomedical Applications

John S. Allen, Chair
Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822

Chair's Introduction—1:35

Invited Papers

1:40

1pBAa1. An all-optical photoacoustic microscopy system for remote, noncontact characterization of biological tissues. Ashwinkumar Sampathkumar, Parag V. Chitnis (Biomedical Eng., Riverside Res., 156 William St., #9, New York, NY 10038, asampathkumar@riversideresearch.org), and Ronald H. Silverman (Dept. of Ophthalmol., Columbia Univ. Medical Ctr., New York, NY)

Conventional photoacoustic microscopy (PAM) employs light pulses to produce a photoacoustic (PA) effect and detects the resulting acoustic waves using an ultrasound transducer acoustically coupled to the target tissue. The resolution of conventional PAM is limited by the sensitivity and bandwidth of the ultrasound transducer. We have investigated an all-optical, "pump-probe" method employing interferometric detection of the acoustic signals that overcomes limitations of conventional PAM. This method does not require contact with the specimen and provides superior resolution. A 532-nm "pump" laser with a pulse duration of 5 ns excited the PA effect in tissue. Resulting acoustic waves produced surface displacements that were sensed interferometrically with a GHz bandwidth using a 532-nm CW "probe" laser using a Michelson interferometer. The pump and probe beams were coaxially focused using a 50× objective giving a diffraction-limited spot size of 0.5 μm. The phase-encoded probe beam was demodulated using a homodyne interferometer. The detected time-domain signal was time reversed using k-space wave-propagation methods to produce a spatial distribution of photoacoustic sources in the target tissue. Performance was assessed using 3D images of fixed, *ex vivo*, retina specimens. Apparatus design and imaging results for the all-optical PA system and possible applications will be discussed.

2:00

1pBAa2. Multimodal ultrasound microscopy for biomedical imaging. Yoshifumi Saijo (Grad. School of Biomedical Eng., Tohoku Univ., 4-1 Seiryomachi, Aoba-ku, Sendai 980-8575, Japan, saijo@idac.tohoku.ac.jp)

Acoustic microscopy provides not only high resolution imaging but also basic data for interpreting clinical ultrasound images and information on biomechanics of the tissues. Multimodal ultrasound microscope is developed for quantitative measurement of sound speed of the tissue. The frequency dependent characteristics of the amplitude and phase of a single pulse deduce the tissue thickness and sound speed. Specific acoustic impedance and elastic bulk modulus are derived by the sound speed and density of the tissue. Ultrasound impedance microscope visualizes microscopic image of the tissue surface by just touching the probe to the tissue. The reflection from the interface between the tissue and plastic plate is obtained to visualize two-dimensional distribution of specific acoustic impedance of the tissue. The multimodal ultrasound microscope realized conventional C-mode, surface impedance mode, B-mode, 3D mode, and combination of photoacoustic imaging. The series of the ultrasound measurements of gastric cancer, renal cancer, prostatic cancer, myocardial infarction, atherosclerosis, cartilage-bone complex, and brain have provided important information for clinical ultrasound imaging and pathophysiology from the point of view of biomechanics. Development of higher frequency transducer or arrayed transducer with newest technologies would realize higher resolution imaging and easier handling.

2:20

1pBAa3. Speed of sound of fatty and fibrosis liver measured by 80-MHz and 250-MHz scanning acoustic microscopy. Tadashi Yamaguchi (Res. Ctr. for Frontier Med. Eng., Chiba Univ., 1-33 Yayoicho, Inage, Chiba 2638522, Japan, yamaguchi@faculty.chiba-u.jp), Kenta Inoue (Grad. School of Eng., Chiba Univ., Chiba, Japan), Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), Kazuto Kobayashi (Honda Electron. Co., Ltd., Toyohashi, Aichi, Japan), and Yoshifumi Saijo (Grad. School of Biomedical Eng., Tohoku Univ., Sendai, Miyagi, Japan)

Early detection of hepatitis is critical for proper patient management and improving disease prognosis. Ultrasound imaging is ideally suited for early-stage assessments, but conventional ultrasound images based on backscatter do not display quantitative tissue information because conventional ultrasound lacks essential modeling of the complex interaction between ultrasound and liver tissue in normal and diseased states. Therefore, speed-of-sound (SOS) measurements were obtained from three types of rat livers (normal, fatty, and fibrosis). Livers were harvested, fixed, and embedded in paraffin; a single 10-μm thin section was obtained using a microtome and placed on a microscope slide. A scanning acoustic microscope incorporating transducers operating at 80- and 250-MHz center frequencies was used to scan the 10-μm section. An adjacent 4-μm thin section was stained with H&E (normal and fatty livers) or Azan (fibrosis livers). The SOS measured with both transducers displayed the same trend: SOS in fatty liver was lower than in normal liver and SOS in fibrosis liver was higher than in normal liver. SOS differences were greater at 250 MHz because of the improved spatial resolution, which allowed choosing region-of-interests containing only fat or fibrosis tissue. These initial results also were used to correlate the pathologic state with the SOS.

2:40

1pBAa4. Sound speed estimation in single cells using the ultrasound backscatter power spectrum. Eric M. Strohm and Michael C. Kolios (Physics, Ryerson Univ., 350 Victoria St, Toronto, ON M5B2K3, Canada, estrohm@ryerson.ca)

The acoustic properties of single cells such as their size, sound speed and attenuation are known to change depending on the type, state, or disease progression of a cell. Typically, ultrasonic pulse echo methods on adherent cells are used. We propose using the ultrasound backscatter power spectrum on cells in suspension to extract the acoustic parameters. When the ultrasound wavelength is on the same order as the dimensions of the cell, periodically varying minima and maxima occur throughout the power spectrum that depend on the sound speed and density of the object and the surrounding fluid, respectively. The ultrasound parameters can be determined by comparing the measured spectrum to a theoretical scattering model. We measured the backscattered ultrasound signals from single MCF7 breast cancer cells in suspension using a 200 MHz transducer. The cell diameter was determined through simultaneous optical imaging. The sound speed was calculated by adjusting the parameters in the scattering model until a good fit of the spectral features between the model and measured agreed. The sound speed from single cells found to vary between 1540 to 1580 m/s when the density was fixed at 1050 kg/m^3.

3:00

1pBAa5. Ultrasonic biomicroscopy and micro-Raman spectroscopy for the mechanochemical characterization of atheromatous lesions. Pavlos Anastasiadis (Molecular Biosciences and Bioengineering, Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, pavlos@hawaii.edu), Shiv Sharma (Hawaii Inst. of Geophys. & Planetology, Univ. of Hawaii, Honolulu, HI), Michelle L. Matter (John A. Burns School of Med., Univ. of Hawaii, Honolulu, HI), and John S. Allen (Mech. Eng., Univ. of Hawaii, Honolulu, HI)

Atherosclerosis is defined as a focal, inflammatory, and fibro-proliferative response to endothelial injury. The development of atheromatous lesions in the coronary tree is predominantly a quiescent asymptomatic process without any clinical manifestations. The unpredictable and acute nature of cardiovascular complications such as vulnerable plaque rupture makes diagnosis and treatment of this disease an outstanding medical challenge. We investigate non-invasive techniques that facilitate mechanical measurements at the microscopic level, which can then be directly correlated to biomarker localization within lesion sites. To characterize these sites *in vitro*, we used time-resolved scanning acoustic microscopy (TRSAM). This technique allows for non-invasive interrogation of tissue samples with optical resolution at the micrometer scale. Furthermore, we combined TRSAM with micro-Raman (micro-RS) spectroscopy to investigate plaque morphology with regard to specific biomarkers. We characterized mechanoelastic and biochemical regions containing high cholesterol, phosphate, and carbonate apatite that are characteristic of atherosclerotic lesions. The mechanoelastic evaluation of these regions was determined using TRSAM. Calcified lesions, for example, exhibit ultrasonic velocities of 1810 m/s $\pm$ 25 m/s and are more rigid and stiffer than normal blood vessel tissues.

Contributed Paper

3:20

1pBAa6. An analysis of the acoustic properties of the cell cycle and apoptosis in MCF-7 cells. Maurice M. Pasternak, Eric M. Strohm, and Michael C. Kolios (Physics, Ryerson Univ., 10 Torresdale Ave. B-1, Toronto, ON M2R3V8, Canada, emilku@hotmail.com)

Through the use of high frequency acoustic microscopy, the acoustic properties of cells through various stages of interphase (G1/G2), mitosis (metaphase, M-phase), and apoptosis were ascertained. The cell thickness, sound velocity, acoustic impedance, density, bulk modulus, and attenuation were determined through a quantitative analysis of the pulse echoes from the cell membrane and substrate using a 375 MHz transducer. Hoechst 33342, Annexin-V, propidium iodide, and FITC-448 Anti-cyclin B1 and D1 mouse antibodies were used to identify cell cycle stage. ANOVA and Tukey *post hoc* statistical tests were used to quantify differences between cell stages. A total of 174 cells, 58 within each category, were measured. A statistically significant increase in thickness (9.4–11.4 μm), and decrease in attenuation (1.20–1.05 dB/cm/MHz) was observed between G1 and G2 cells, respectively. A statistically significant increase in thickness, and decrease in acoustic impedance, density, bulk modulus, and attenuation was observed between M-phase and G1 or G2. During apoptosis, minor differences were observed between interphase and early apoptosis; however, significant differences in nearly all properties were observed as the cells progressed to late stage apoptosis. The differences found indicate considerable structural and/or organizational alterations occurring as the cell progresses through these phases.

MONDAY AFTERNOON, 3 JUNE 2013 — 519A, 1:00 P.M. TO 5:20 P.M.

Session 1pBAb

Biomedical Acoustics: Ultrasound Contrast Agents and Passive Cavitation Mapping of High Intensity Focused Ultrasound Lesion Formation

Eleanor Stride, Chair
Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom

Contributed Papers

1:00

1pBAb1. Acoustic characterization and modeling of poly-lactic acid-encapsulated contrast microbubbles. Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu) and Shirshendu Paul (Mech. Eng., Univ. of Delaware, Newark, DE)

Biodegradable polymers like polylactic acid hold potential for better stability and control over encapsulation properties of ultrasound contrast microbubbles. We report here several interesting acoustic properties of air-filled PLA shelled microbubbles through both *in vitro* experiments and mathematical modeling. Attenuation measurements with PLA microbubbles (average diameter 1.9 micrometer), indicated a resonance frequency of 2.5–3 MHz, which, in contrary to other encapsulated microbubbles, is lower than the resonance frequency of a free bubble of similar size. Pressure dependent scattering measurements at two different excitation frequencies (2.25 and 3 MHz) show strongly non-linear behavior with distinct second and subharmonic responses. Subharmonic responses are registered above a relatively low generation threshold of 100–150 kPa. To investigate the underlying mechanisms, we utilized several preexisting interfacial models describing encapsulated bubble dynamics. The attenuation data were utilized to determine the interfacial rheological properties of the encapsulation for each of these models. The model predictions are then compared with scattered nonlinear—sub- and second harmonic—responses. Our studies indicate that the extremely low surface elasticity (around 0.01 N/m) and reduced surface tension (0.01–0.03 N/m) along with the polydispersity of the bubble suspension play a critical role in determining the acoustic properties of PLA microbubbles.

1:20

1pBAb2. Acoustic and optical characterization of targeted ultrasound contrast agents. Camilo Perez, Jarred Swalwell (Ctr. for Industrial and Med. Ultrasound (CIMU), Univ. of Washington Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105-6698, camipiri@uw.edu), Juan Tu (Key Lab. of Modern Acoust., Nanjing Univ., Nanjing, China), Hong Chen, Andrew Brayman, and Thomas J. Matula (Ctr. for Industrial and Med. Ultrasound (CIMU), Univ. of Washington Appl. Phys. Lab., Seattle, WA)

We previously developed a flow cytometer system that incorporates the action of ultrasound to characterize shell properties of ultrasound contrast agents (UCA's). The most recent manifestation involves a flow cytometer modified with a custom square quartz flow cell in place of the standard nozzle and fluid jet. Acoustic coupling to the carrier sheath fluid and UCA samples occurs through a PZT bonded to one side of the flow cell. The PZT-driven UCA oscillations were processed and fitted to the Marmottant UCA model. Shell properties for UCAs (including Definity, Optison, SonoVue, and even homemade bubbles) were determined. A major limitation of the previous work involved a lack of knowledge of the actual acoustic pressure incident on the UCA. The focus of this talk will be on optimization of the pressure inside the flow cell using finite element methods, and the comparison with additional measurements of unpublished data from targeted UCA's. [Work funded in part by the Life Sciences Discovery Fund #3292512.]

1:40

1pBAb3. Investigating the effect of fabrication method on the stability and acoustic response of microbubble agents. Graciela Mohamedi (Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Rd. Campus, Headington, Oxford OX3 7DQ, United Kingdom, graciela.mohamedi@eng.ox.ac.uk), Neveen A. Hosny (Dept. of Chem., Imperial College London, London, United Kingdom), Paul Rademeyer (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Yoonjee Park (Dept. of Biomedical Eng., Boston Univ., Boston, MA), Joshua Owen (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Tuan Pham, Joyce Y. Wong (Dept. of Biomedical Eng., Boston Univ., Boston, MA), Marina Kuimova (Dept. of Chem., Imperial College London, London, United Kingdom), and Eleanor Stride (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Microbubbles stabilized by a surfactant or polymer coating are already in clinical use as ultrasound imaging contrast agents. They have also been widely investigated as vehicles for drug delivery and gene therapy that can be tracked and triggered using ultrasound. Extensive studies have been made of the effects of the coating material and gas core on microbubble characteristics, but the influence of the fabrication method has received less attention. The aim of this study was to compare the behavior of microbubbles prepared using different techniques. Phospholipid-coated microbubbles were produced using sonication, electrospraying, or in a specially designed microfluidic device. The microbubbles were observed using optical, electron, and fluorescence lifetime imaging microscopy (FLIM) to interrogate their surface microstructure and stability over time. Their acoustic response was then determined in a flow chamber by detecting the pressure scattered from individual microbubbles as they passed through the focal region of a transducer (center frequencies 1, 2.25, and 3.5 MHz; peak negative pressures 50–300 kPa). The method of bubble generation was found to significantly affect the bubble surface characteristics, stability, and acoustic response. The results demonstrate that the processing method affects not only the bubble size distribution but other characteristics important for biomedical applications.

2:00

1pBAb4. Radiation for bubble contrast agents in inhomogeneous media. Chrisna Nguon, Max Denis, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, 63 Hemlock St., Dracut, MA 01826, chrisna_Nguon@student.uml.edu)

The acoustic field generated by a distribution of micro-bubbles serving as contrast agents for a three-dimensional scattering volume is evaluated. A dual-frequency incident field generated by a confocal transducer insonifies the target and creates a scattered field that includes the difference frequency component that is of interest for improving the resolution in imaging of biological tissue. The scattered pressure is computed for a range of compressibility contrast parameters and wavenumbers using Born series and Padé approximants to ensure convergence as the medium contrast is increased. The effect of this pressure field on the resonant radiation of bubbles is examined, and bubble parameters that influence the amplification of the field measured exterior to the scattering volume are identified. A baseline comparison of the scattered pressure field with and without the presence of bubble contrast agents is presented.

2:20

1pBAb5. Temporal evolution of subharmonic emissions from a lipid-encapsulated contrast agent. Himanshu Shekhar (Elec. and Comput. Eng., Univ. of Rochester, 212 Conant Rd. Apt. C, Rochester, NY 14623, himanshushekhar@rochester.edu), Joshua J. Rychak (Targeson Inc., San Diego, CA), and Marvin M. Doyley (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

A large body of work has investigated the influence of the excitation pulse, agent size distribution, and the ambient pressure on the subharmonic response of microbubble contrast agents (MCA). The purpose of this study was to investigate the temporal evolution of the subharmonic emissions, i.e., whether the subharmonic response is influenced by the time elapsed since agent constitution. We measured subharmonic emissions from a commercial lipid-encapsulated contrast agent (Targestar-P®, Targeson Inc., San Diego) over the time span of 60 min. The excitation parameters were as follows: 10-MHz frequency, 30–290 kPa pressures, 60 cycles, and 1-kHz pulse repetition frequency. The subharmonic emissions were observed to increase by 11 dB over 60 min relative to those measured immediately after reconstitution. The most striking increase (>8 dB) was observed in the first 15 min. Although we did not observe a change in the agent size distribution, the pressure threshold for subharmonic emissions reduced by nearly two-fold within the time span of our measurements. This work demonstrates that time evolution of subharmonic emissions could bias quantitative estimates obtained from techniques such as subharmonic imaging and subharmonic-aided pressure estimation. Additionally, these findings suggest the possibility for improving subharmonic emission by careful agent design.

2:40–3:00 Break

3:00

1pBAb6. Simulations of transcranial passive acoustic mapping with hemispherical sparse arrays using computed tomography-based aberration corrections. Ryan Jones (Med. Biophysics, Univ. of Toronto, 2075 Bayview Ave., Focused Ultrasound Lab (C713), Toronto, ON M4N 3M5, Canada, ryanjones017@gmail.com), Meaghan O'Reilly (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada), and Kullervo Hynynen (Med. Biophysics, Univ. of Toronto, Toronto, ON, Canada)

Passive acoustic mapping (PAM) is receiving increasing interest as a method for monitoring focused ultrasound (FUS) therapy. PAM would be beneficial during transcranial cavitation-enhanced FUS treatments, particularly non-thermal, cavitation-mediated applications such as FUS-induced blood–brain barrier disruption or sonothrombolysis, for which no real-time monitoring technique currently exists. However, the use of PAM in the brain is complicated by the presence of the skull bone. If not properly accounted for, skull-induced aberrations of propagating cavitation emissions will lead to image distortion and artifacts upon reconstruction. Through the use of numerical simulations, this study investigated the feasibility of transcranial PAM via hemispherical sparse hydrophone arrays. A multi-layered ray acoustic transcranial ultrasound propagation model based on computed tomography-derived skull morphology was developed. By incorporating skull-specific aberration corrections into a conventional passive beamforming algorithm [Norton and Won, IEEE Trans. Geosci. Remote Sens. **38**, 1337–1343 (2000)], simulated acoustic source fields were spatially mapped through digitized human skulls. The effects of array sparsity and receiver element configuration on the formation of passive acoustic maps were examined. Multiple source locations were simulated to determine the imageable volume within the skull cavity. Finally, the reconstruction algorithm's sensitivity to noise was explored.

3:20

1pBAb7. Transcranial spatial and temporal assessment of microbubble dynamics for brain therapies. Costas Arvanitis and Nathan McDannold (Radiology, Brigham and Women's Hospital, Harvard Med. School, 221 Longwood Ave., Rm. 514a, Boston, MA MA 02115, cda@bwh.harvard.edu)

Harnessing ultrasound/microbubble interactions in the brain may make possible a number of therapeutic ultrasound applications, such as targeted drug delivery, sonothrombolysis, and cavitation-enhanced ablation. However, methods to guide these emerging therapies are presently lacking. Here, we integrated a linear US imaging transducer with a clinical transcranial MRI-guided focused ultrasound (MRgFUS) system and evaluated passive cavitation imaging to monitor microbubble-enhanced sonications. A nonhuman primate skull filled with brain-mimicking phantom was used for the experiments. First, we sonicated the phantom over a range of powers (20–60 W) to induce cavitation-enhanced heating. Using transcranial passive cavitation mapping and MR thermometry, we assessed the ability of the integrated system to simultaneously visualize temperature changes and microbubble activity. In another experiment, we traversed the phantom with a 2 mm channel through which microbubbles could flow and applied burst sonications (5 W) to generate stable and inertial cavitation. In the first experiment, cavitation activity and heating were colocalized. In the second, the location of the cavitation activity was coincident with the targeted location in the channel within the expected resolution of the passive imaging. We conclude that combined MR/ultrasound imaging can provide comprehensive guidance to simultaneously localize and quantify both acoustic cavitation activity and heating.

3:40

1pBAb8. Spatial specificity and sensitivity of passive cavitation imaging for monitoring high-intensity focused ultrasound thermal ablation in ex vivo bovine liver. Kevin J. Haworth (Internal Med., Univ. of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209, kevin.haworth@uc.edu), Vasant A. Salgaonkar (Biomedical Eng. Program, Univ. of Cincinnati, San Francisco, California), Nicholas M. Corregan (Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH), Christy K. Holland (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH)

Passive cavitation images (PCIs) generated from scattered acoustic waves are a potential technique for monitoring lesion formation during high-intensity focused ultrasound (HIFU) thermal ablation. HIFU lesion prediction by PCIs was assessed in *ex vivo* bovine liver samples ($N = 14$) during 30-s sonications with 1.1-MHz continuous-wave ultrasound (1989 W/cm^2 estimated spatial-peak intensity). Treated samples were sectioned, optically scanned, and the HIFU lesions segmented based on tissue discoloration. During each insonation, a 192-element, 7-MHz linear array (L7/Iris 2, Ardent Sound) passively recorded emissions from a plane containing the HIFU propagation axis oriented parallel to the image azimuth direction. PCIs were formed from beamformed A-lines filtered into fundamental, harmonic, ultraharmonic, and inharmonic frequency bands. Lesion prediction was tested using binary classification of local tissue ablation based on thresholded PCIs, with spatial specificity and sensitivity of lesion prediction quantified by the area under receiver operating characteristic curves (AUROC). Tadpole-shaped lesions were best predicted by harmonic emissions ($AUROC = 0.76$), prefocal lesions were best predicted by harmonic or ultraharmonic emissions ($AUROC = 0.86$), and cigar-type focal lesions were best predicted by fundamental and harmonic emissions ($AUROC = 0.65$). These results demonstrate spatial specificity and sensitivity when predicting HIFU lesions with PCIs. [Work supported in part by NIH grants F32HL104916 and R21EB008483.]

4:00

1pBAb9. Real-time three-dimensional passive cavitation detection for clinical high intensity focused ultrasound systems. Jamie Collin, Christian Coviello, Erasmia Lyka (Inst. of Biomedical Eng., Univ. of Oxford, ORCRB, Headington, Oxford OX3 7DQ, United Kingdom, jamie.collin@eng.ox.ac.uk), Tom Leslie (Dept. of Urology, Oxford Univ. Hospitals, Oxford, United Kingdom), and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Bubble activity during high intensity focused ultrasound (HIFU) surgery has been linked with desirable effects, such as an enhanced heat deposition caused by inertial cavitation, and undesirable effects, such as lesion migration caused by boiling bubbles. There is presently no reliable way of achieving spatiotemporal monitoring of cavitation activity during clinical HIFU treatments. In the present work, a near-acoustically transparent two-dimensional 32-element PVDF array was designed and mounted on the therapy transducer of a clinical HIFU device (Model JC200, Chongqing Haifu) to enable detection of acoustic emissions arising from cavitation during therapy. The signal detected by each of the elements was digitized and processed in real time on a graphical processing unit (GPU), and beamformed using our previously described passive acoustic mapping (PAM) algorithm to produce real-time three-dimensional (3D) maps of cavitation activity with a frame rate in excess of 5 Hz. The system was initially validated in agar-based tissue-mimicking materials, demonstrating that the displayed volume of cavitation activity agreed with predictions based on in situ pressure calibrations. The system was further validated during clinical HIFU treatments of kidney tumor, liver tumor, and uterine fibroid ablation, and was found to enable accurate localization of the HIFU focus at sub-lesioning intensities.

4:20

1pBAb10. Passive acoustic mapping using optimal beamforming for real-time monitoring of ultrasound therapy. Christian Coviello (Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., ORCRB, Oxford OX3 7DQ, United Kingdom, christian.coviello@eng.ox.ac.uk), Richard Kozick (Dept. of Elec. Eng., Bucknell Univ., Lewisburg, PA), James J. Choi (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Miklós Gyöngy (Faculty of Information Technol., Pázmány Péter Catholic Univ., Budapest, Hungary), Jamie Collin, Carl R. Jensen, Penny Probert Smith, and Constantin C. Coussios (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

In ultrasound therapy, passive acoustic mapping (PAM) has been shown to be an effective method for imaging the acoustic emissions generated during treatment providing the potential for real-time therapy monitoring. In both high intensity ultrasound (HIFU) ablative surgery and targeted drug delivery, imaging artifacts at higher amplitude exposure conditions have been observed, which make the localization and dosimetry of therapeutically relevant cavitation activity a challenge. Due to these artifacts, correlating drug release or lesion volumes to the PAMs is hindered for many exposures. It is proposed that incorporating optimal beamforming techniques into the PAM framework can reduce and remove these artifacts, allowing determination of the extent of cavitation activity during ultrasound therapy. Additionally, optimal beaforming is found to yield improved resolution, good interference suppression, and robustness against steering vector errors. A description of the origin of the artifacts as well as reduction of them by implementing optimal beamforming within PAM will be demonstrated in the context of targeted drug delivery.

4:40

1pBAb11. Passive acoustic mapping of magnetic microbubbles in an *in vitro* flow model. Calum Crake, Marie de Saint Victor, Christian Coviello, Joshua Owen, Constantin-C Coussios, and Eleanor Stride (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, calum.crake@eng.ox.ac.uk)

Magnetic microbubbles can be successfully retained near a vascular target and simultaneously imaged using conventional B-mode ultrasound. When further modified to carry a drug, they could enable significant enhancements in targeted drug delivery for applications such as sonothrombolysis, where stable cavitation has been shown to play a key role. However, the effect of the increased proximity of the microbubbles under the effect of the magnetic field on their acoustic response remains unknown. Passive acoustic mapping is a method that enables real-time spatiotemporal monitoring of cavitation dynamics in an arbitrary plane or volume within the field of view of the ultrasound probe, and classification of the type of cavitation activity on the basis of the spatial distribution of frequency-domain emissions. In the present work, PAM is used to investigate the effect of bubble proximity and flow rate on the type, sustainability, intensity, and spatial distribution of cavitation activity observed for both magnetic and non-magnetic microbubbles excited by 0.5 MHz therapeutic ultrasound in an *in vitro* flow model. It is hoped that this study will not only yield a new method for real-time monitoring of drug delivery using magnetically trapped microbubbles, but will also help elucidate complex bubble–bubble interactions in therapeutic ultrasound fields.

5:00

1pBAb12. A two-component speckle model for detection of microbubble signals in linear contrast-enhanced ultrasonography. Matthew R. Lowerison (Dept. of Medical Biophys. and Robarts Res. Inst., The Univ. of Western Ontario, 95 Salem Place, London, ON N6K 1T8, Canada, mloweri@imaging.robarts.ca), M Nicole Hague, Ann F. Chambers (London Regional Cancer Program, London, ON, Canada), and James C. Lacefield (Dept. of Medical Biophys. and Robarts Res. Inst., The Univ. of Western Ontario, London, ON, Canada)

Contrast-enhanced ultrasound (CEUS) serves oncology by imaging tumor blood supply to enable quantification of longitudinal vascular changes and monitoring of treatment responses. Unfortunately, the linear subtraction methods commonly used for preclinical imaging are susceptible to registration errors and motion artifacts that lead to reduced contrast-to-tissue ratios. In this presentation, an alternative approach is proposed to improve discrimination between the contrast and tissue signals by comparing the

1p MON. PM

first-order speckle statistics of images acquired before and after injection of microbubbles. The microbubble signal component is modeled as a temporally varying random process superimposed on a Rayleigh-distributed speckle signal representing backscatter from tissue. Images were acquired at 18 MHz from a murine orthotopic (mammary fat pad) xenograft breast cancer model following a bolus injection of microbubbles. Images were processed using gold-standard pulse inversion nonlinear CEUS, conventional linear subtraction, and the proposed statistical method. In comparison to conventional linear CEUS, the statistical method produced a wash-in curve that showed closer agreement to the gold-standard nonlinear CEUS data. The statistical method eliminates baseline image subtraction from linear CEUS processing, which should streamline the imaging workflow, improve the robustness of image quantification, and enable real-time perfusion imaging with linear CEUS.

MONDAY AFTERNOON, 3 JUNE 2013 512AE, 1:00 P.M. TO 4:40 P.M.

Session 1pEAa

Engineering Acoustics: Active and Passive Control of Fan Noise

Alain Berry, Cochair
Dept. Mech. Eng., Univ. of Sherbrooke, Sherbrooke, QC J1K 2R1, Canada

Anthony Gerard, Cochair
Soft dB, 1240 Ave. Beaumont, Bureau 206, Mont Royal, QC H3P3E5, Canada

Invited Paper

1:00

1pEAa1. Active control of axial and centrifugal fan noise. Scott D. Sommerfeldt and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N181 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

Both axial and centrifugal fans are used to cool information technology (IT) equipment. These fans generate noise that can be annoying to their users, particularly the tonal noise that can be radiated. Work has focused on developing a method to attenuate the tonal noise associated with both of these types of fans. A compact system is used, whereby control sources are placed in close proximity to the fan. A genetic algorithm has been implemented to determine optimal source configurations. The attenuation associated with some configurations is found to be much more sensitive to error than others. For a given configuration, by using a relatively simple point source model it becomes possible to identify optimal near-field error sensor locations, which results in a compact noise control solution that provides significant global attenuation of the radiated tonal noise. This paper will review progress that has been made to apply this method to both axial and centrifugal fans. Experimental results confirm that it is feasible to achieve significant global control using this method.

Contributed Papers

1:20

1pEAa2. Industrial fan noise control using flow obstructions. Remy Oddo, Anthony Gérard (GAUS, 2500 Boul. de l'Universite, Sherbrooke, QC J1K2R1, Canada, Remy.Oddo@USherbrooke.ca), Michel Pearson (Soft dB, Québec, QC, Canada), Adrien Amyotte, Patrice Masson (GAUS, Sherbrooke, QC, Canada), Franck Sgard (IRSST, Montréal, QC, Canada), and Alain Berry (GAUS, Sherbrooke, QC, Canada)

Fans are used in a lot of industrial processes and are sometimes a source of important noise for workers. The first aim of this study was to identify some problematic fans in Québec industries, for which the noise exposure exceeds the CSST (Comission de la santé et sécurité au travail) limit of 90 dBA for 8 h. We have focused on fans having a high tonal content, for which the Simple Silence technology can be applied, i.e., tonal fan noise control using obstructions in the flow. Two analytical models of the tonal noise radiated by these fans have been proposed: a free field model based on the Lowson model and an in-duct model based on the Goldstein model. The first free-field model has been applied to the control of the noise from a series of eight evaporator fans, having a strong blade passage frequency (BPF) tone at 90 Hz. These fans have been controlled *in-situ*, in a cold storage room, using trapezoidal obstructions in the downstream flow field of the fans. The second in-duct model has been applied to control the tonal noise from an in-duct air-extractor fan used in the underground gold mining galleries. Several obstructions have been tested in the upstream flow field.

1:40

1pEAa3. Analyzing the impact of the inlet temperature on the acoustic noise production form a supersonic jet using large eddy simulations. Bernhard Semlitsch, Mihai Mihaescu, Laszlo Fuchs (Mechanics - Linne Flow Ctr., KTH - Royal Inst. of Technol., Osquars Backe 18, Stockholm 10044, Sweden, bernhard@mech.kth.se), and Ephraim Gutmark (School of Aerosp. Syst., UC - Univ. of Cincinnati, Cincinnati, OH)

Non-ideal expanded supersonic jets emerging from a convergent-divergent nozzle produce three different types of noise, i.e., the shock-associated broadband noise, the screech noise, and the turbulent mixing noise. Interesting to note that the screech tone outside of the nozzle was exposed mostly in downscaled laboratory experiments, while under realistic conditions the exhaust jet of a gas turbine engine does not show this phenomena. Apart from a geometric scaling difference, usually a lower temperature is employed in experimental studies. It is believed that the screech tone occurs due to self-excitation of the shear-layer in a feedback-loop. Acoustic waves generated by vortical structures interacting with a shock are propagating upstream within the subsonic region of the shear-layer or outside of it. They eventually hit the nozzle's lip and excite instabilities at a certain frequency. The compressible Navier-Stokes equations are simulated numerically by using Large Eddy Simulation approach. The effect of supersonic jet operation temperature onto the associated noise is investigated. The jet-exit Mach-number is 1.56, while the total temperature ratios considered at the

inlet plane of the nozzle are 1.27, 2.05, and 3.65. The differences in the near-field acoustics will be presented in each of the cases and the flow-acoustic interaction will be analyzed and quantified.

2:00

1pEAa4. Low frequency sound absorption of resonators with flexible tubes. Frank Simon (DMAE, Onera, 2 Ave. Edouard Belin, Toulouse 31055, France, frank.simon@onera.fr)

Classically, passive acoustic liners, used in aeronautic engine nacelles to reduce radiated fan noise, have a quarter-wavelength behavior, thanks to perforated sheets backed to honeycombs (SDOF, DDOF). So, their acoustic absorption ability is naturally limited to medium and high frequencies because of constraints in thickness. To drastically improve their capabilities to the lowest frequencies, the combination with active control systems or the using of foam architecture have shown an interest, but the industrial application is tricky (i.e., problems of fouling, robustness). A possible approach is to carry out a perforated panel resonator with flexible tube bundles to shift the resonance frequency to a lower frequency by a prolongation of air column length (Yadong Lu *et al.*, Internoise 2007). This paper describes theoretically this concept that allows a significant change in the acoustic impedance due to the large thickness of the resistive and reactive material and the coupling with the surrounding cavity. Applied to aeronautical configurations, the resonance frequency decreases considerably compared to a conventional resonator (factor of about 1/5) but with a reduction of the maximum absorption when the tubes fill the cavity. Experiments in impedance tube validate the theoretical approach.

Invited Paper

2:20

1pEAa5. Volumetric resistance blower. Mark MacDonald and Douglas Heymann (Intel Corp., HF2-40, 5200 Elam Young Pkwy, Hillsboro, OR 97124, mark.macdonald@intel.com)

This paper reports on a new low-noise blower rotor technology developed by Intel Corporation (patents pending). The new approach replaces the traditional centrifugal blower rotor with a block of continuous porous media. The porous media can be as simple as a low-cost, block of open-cell foam and has no blades or macroscale structure. As the porous media rotor rotates, viscous and inertial forces from the volumetric resistance of the porous media cause the air within the rotor to rotate with it, creating centrifugal forces that overwhelm the flow resistance in the radial direction and create a flow pattern similar to that achieved in a traditional blower. However, because of the lack of distinct blades, the porous-media generates nearly zero aerodynamic tonal noise and significantly reduced broadband noise. This allows the rotor to be operated at significantly higher RPM and reduced clearances relative to the traditional rotor design for further improved performance. This paper will discuss numerical modeling and experimental development of the new blower type. An iso-flow comparison of porous-media and traditional rotors with the same motor and housing demonstrate a 5 dBA reduction in broadband noise and a factor of two reduction in tonality while maintaining comparable overall efficiency. Impact of porosity and different rotor support structures are also discussed.

Contributed Papers

2:40

1pEAa6. Effect of standoff distance on the reconstruction of in-duct velocity field and regeneration of pressure field. Yong-Ho Heo and Jeong-Guon Ih (Mech. Eng., KAIST, Mech. Blg. Rm. 5121, KAIST, Daejeon, Chungcheongnam-do 305-701, South Korea, yonghoheo@kaist.ac.kr)

Identification of in-duct acoustic source characteristics is essential in the design of fluid machinery system for reducing and predicting the flow-generated noise. To this end, the inverse estimation method can be employed by using the measured sound field and matrix formulation for wave propagation within a duct. In this paper, the effect of the distance between source and measurement plane is investigated. At each standoff distance, pressures are measured at three planes with two different spacings to widen the estimation frequency range, and measurements are conducted with three different standoff distances. Modal decomposition is applied to estimate modal amplitudes, and the result is used to reconstruct the velocity field at the source plane and to obtain the regenerated pressure field at the measurement planes. It is shown that the modal amplitude identified by measured pressure field at the short standoff distance, i.e., at nearfield, can yield an accurate reconstructed velocity field of the source and regenerate the pressure field with smaller error, which is similar to the other inverse techniques such as equivalent source method and nearfield acoustical holography. A field reduction example by suppressing some parts of source velocity field is shown for demonstrating the effectiveness of the method.

3:00

1pEAa7. Numerical investigation of acoustically excited flow through an orifice using lattice Boltzmann method. Chenzhen Ji and Dan Zhao (Aerosp. Eng., Nanyang Technol. Univ., 50 Nanyang Ave., Singapore 639798, Singapore, cji1@e.ntu.edu.sg)

Two-dimensional time-domain numerical investigation of sound-induced flow through an orifice with a diameter 6 mm is conducted by using lattice Boltzmann method. Emphasis is placed on characterizing its acoustic damping behaviors. The main damping mechanism is identified as incident waves interact with the shear layers formed at the orifices rims and the acoustic oscillations destabilize the shear layers to form vortex rings. And acoustic energy is converted into vortical energy. To quantify the orifice damping effect, power absorption coefficient is used. It is related to Rayleigh conductivity and describes the fraction of incident acoustical energy being absorbed. Numerical simulations are conducted in time domain by forcing a fluctuating flow with multiple tones through the orifice. This is different from frequency-domain simulations, of which the damping is characterized one frequency at a time. Comparing our results with those from Howe's theoretical model, good agreement is observed. In addition, orifice thickness effect on its damping is discussed.

3:20

1pEAa8. Scattering of sound waves at an area expansion in a cylindrical flow duct. Susann Boij (Marcus Wallenberg Lab. for Sound and Vib. Res., Dept. of Aeronautical and Vehicle Eng., KTH Royal Inst. of Technol., 2004 Yolo Ave., Berkeley, California 94707, sboij@kth.se), Özge Yanac Cinar, Gökhan Cinar (Gebze Inst. of Technol., Kocaeli, Turkey), and Börje Nilsson (School of Comput. Sci., Phys. and Mathematics, Linneaus Univ., Växjö, Sweden)

Sound propagation in pipes and ducts with flow, like ventilation ducts and exhaust pipes, is influenced by flow separation and vortex production at sharp edges along the ducts, such as at bends and area expansions. Shear layers form at the separation points, and such layers are unstable to low frequency acoustic disturbances. An analytical model, aiming at physical insight into this interaction, is presented. Results in the plane wave region for the so called scattering matrix for a sudden area expansion with flow in cylindrical pipes are compared with experimental values. Both the magnitude and the phase, in the form of an end correction, is presented. The model is also compared to a 2 dimensional model, in order to evaluate the anticipated increased accuracy of the 3 dimensional modeling. The scattering coefficients are strongly dependent on the flow speed, which is up to a Mach

number of 0.5. It is observed that for low frequencies, the interaction is dominated by the dynamics of an unstable shear layer downstream of the edges. For higher frequencies, the wave propagation is mainly affected by convective effects. Differences in properties for the 2D and the 3D cases are also explored.

3:40

1pEAa9. Design of a built-in electroacoustic resonator for active noise reduction. Romain Boulandet, Etienne Rivet, and Hervé Lissek (Laboratoire d'ElectroMagnétisme et Acoustique, Ecole Polytechnique Fédérale de Lausanne, ELB Station 11, Lausanne CH-1015, Switzerland, romain.boulandet@gmail.com)

The paper focuses on the design of a built-in electroacoustic resonator for active noise reduction purposes. This concept basically encompasses a loudspeaker connected to a synthetic electrical load that enhances the ability of the transducer to dissipate a certain part of the incoming acoustic energy. The strategy is therefore to control the dynamics of boundaries in closed sound spaces (such as room, cavity, etc.) rather than targeting a global control that requires significant input of additional acoustic energy. The main attraction of the proposed methodology is its ability to achieve broadband sound absorption while bypassing the use of sensors, the sensing of sound field information being incorporated within the synthetic electrical load admittance (current/voltage transfer function). Computational and experimental results are provided to illustrate the benefits and potential of a built-in electroacoustic resonator compared to other options. Concluding remarks and discussions on foreseen future developments are then provided.

4:00

1pEAa10. The threshold of the difference between a mathematical model applied to active noise control and data recorded. Ricardo A. Quintana and Adriana P. Gallego (Universidad Distrital Francisco José de Caldas, Calle 7A # 73 - 98 Apto 503 Int 4, Bogotá 11001000, Colombia, rquintana@raqsacoustic.com)

Nowadays, there are many methods used to obtain mathematical models applied to active noise control, especially when the transfer function is required. Inside rooms, the global active sound control has bad results due to the reflections and the diffuse field. Then, authors have applied system identification to find more complex mathematical models based on measured data. Also, the number of system identification methodologies is increasing and it carries to having many unexplored models. In order to know which models are useful for global active noise control inside rooms, a relationship between the sound pressure level decreased and the error of the mathematical model is presented. First, the concept of "a useful mathematical model" is defined under any context based on an analysis of the error (FIT). In addition, this concept is delimited to the active noise control context. Finally, an example is presented.

Invited Paper

4:20

1pEAa11. Upgrade of a multi-channel active noise control system for an industrial stack. Louis-Alexis Boudreault, André L'Espérance, and Alex Boudreau (Soft dB Inc., 1040, Ave. Belvedere, Ste. 215, Quebec, QC G1S 3G3, Canada, la.boudreault@softdb.com)

Active noise control has been studied in the 1990s as an innovative way to reduce the noise in specific situations. Some applications are well known today and found commercial success like noise-canceling headphones. However, the use of active noise control in industrial applications is more complex, thus being an uncommon solution in this field. The use of active noise control for industrial stack noise is one of these applications. One of the first large-scale implementation has been set up at the end of the 1990s. This system was a 10-channel active noise control system installed on a 1.8 m wide chimney to attenuate a 320 Hz pure tone. At that time, an 8 dB noise reduction was achieved at error microphones. Fifteen years later, it has been decided to upgrade the system with the latest generation of digital signal processor (DSP) allowing a real-time optimization and better tracking speed. This paper describes the overall system and the updated multi-channel active noise controller developed for this application. It also presents the improvements, the achieved noise reduction, and the associated environmental benefits.

MONDAY AFTERNOON, 3 JUNE 2013 512BF, 1:00 P.M. TO 5:00 P.M.

Session 1pEAb

Engineering Acoustics: Transduction, Transducers, and Energy Harvesting

Stephen Butler, Chair
NUWC, Newport, RI 02841

Contributed Papers

1:00

1pEAb1. Ultra-low frequency underwater acoustic projectors: Present status and future trends. Bertrand Dubus, Pascal Mosbah (ISEN, IEMN UMR CNRS 8520, 41 Boulevard Vauban, Lille cedex 59046, France, bertrand.dubus@isen.fr), Jean-Rémi Hartmann, and Jacky Garcin (Techniques Navales /SDT/SCN/LSM/DSM, DGA, Toulon, France)

Ultra-low frequency (ULF) underwater transducers, used in the 10–400 Hz frequency range, have usually radiating surfaces the dimensions of which are small with respect to the acoustic wavelength. To radiate a high acoustic power with a monopolar ULF transducer, a large volume velocity is required to counterbalance the low radiation resistance. Three transduction technologies are available to realize compact high power ULF transducers: hydroacoustic, electromagnetic, and active material-based. In the latter case, piezoelectric ceramics and magnetostrictive rare-earth alloys are often associated to flexural vibration such as found in flextensional transducers. Compared to these materials, piezoelectric single crystals, which exhibit lower stiffnesses and produce higher strains together with higher energy densities, are potential active materials for future ULF underwater transducers. In this work, ULF transducers are analyzed in terms of their working frequencies, acoustic powers, and masses. Thirty-two ULF underwater projectors build during the last 25 years are considered. For single

crystal transducers, prototypes working at higher frequencies as well as transducers modeled with finite element method are taken into account. Using these data and classical scaling laws, abacuses displaying acoustic power-frequency curves for given masses are constructed for each technology. They show that single crystals transducers could provide more compact and powerful solutions for frequencies above 40 Hz.

1:20

1pEAb2. Performance of transducers with segmented piezoelectric stacks using materials with high electromechanical coupling coefficient. Stephen C. Thompson, Richard J. Meyer, and Douglas C. Markley (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16803, sct12@psu.edu)

Abstract underwater acoustic transducers often include a stack of thickness polarized piezoelectric material pieces of alternating polarity interspersed with electrodes and bonded together. The stack is normally much shorter than a quarter wavelength at the fundamental resonance frequency, so that the mechanical behavior of the transducer is not affected by the segmentation. When the transducer bandwidth is less than a half octave, as has conventionally been the case, stack segmentation has no significant effect on the mechanical behavior of the device. However, when a high coupling coefficient material such as PMN-PT is used to achieve a wider bandwidth, the difference between a segmented stack and a single piezoelectric piece with the same overall dimensions can be significant. This paper investigates the effects of stack segmentation on the performance of wideband underwater acoustic transducers, particularly tonpilz transducer elements. Included is discussion of transducer designs using single crystal piezoelectric material with high coupling coefficient compared with more traditional PZT ceramics.

1:40

1pEAb3. On the spatial distributions for randomly spaced arrays. Jenny Au and Charles Thompson (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, jenny_au@student.uml.edu)

In this work, we examine the statistical characteristics of the randomly spaced transducer arrays. Approaches to realizing the linear and two-dimensional arrays are considered. The array elements are equally weighted in amplitude and their contribution to received signal is by virtue of adjustment by their spatial location. The cumulative distribution for the number of transducers as a function of transducer spacing and its relationship to the spatial objective function is given. The case for Dolph–Chebychev objective function is shown in detail and in closed-form. The statistical effect spatial binning of transducer elements is examined.

2:00

1pEAb4. Electroacoustic metamaterials: Achieving negative acoustic properties with shunt loudspeakers. Herve Lissek (LEMA, Ecole Polytechnique Federale de Lausanne, STI IEL LEMA, Station 11, Lausanne 1015, Switzerland, herve.lissek@epfl.ch)

Acoustic metamaterials constitute a new class of structures that exhibit acoustic properties not readily available in nature. These properties can be a negative mass density, expressing the opposition of the acceleration of a particle to the application of pressure, or a negative bulk modulus, signifying the rarefaction of the particle in reaction to a compression (resp. a condensation in reaction to a depression). However, these artificial behaviors result from a periodic arrangement of passive unit-cells (such as membranes and side holes), and not from individual "meta-properties" of each unit-cell. It is however possible to observe such intrinsic metamaterial properties out of a passive electroacoustic resonator. This concept encompasses a loudspeaker, connected to a specific electric load, thus altering the acoustic dynamics of the loudspeaker diaphragm when subject to an exogenous sound source. It is especially possible to achieve negative acoustic impedance at its diaphragm, thanks to the connection of passive electric shunt circuits, such as simple RLC series resonators. This paper aims at highlighting the metamaterial nature of such electroacoustic resonators through computational and experimental results, followed by discussions on ongoing developments.

2:20

1pEAb5. Loudspeaker for low frequency signal driven by four piezoelectric ultrasonic motors. Juro Ohga (Ohga Acoust. Lab., 2-24-3 Tamanawa, Kamakura 247-0071, Japan, johga@nifty.com), Hirokazu Negishi (MIX Corp., Yokosuka, Japan), Ikuo Oohira (I. Oohira and Assoc., Yokohama, Japan), Hiroya Saito, Kunio Oishi (Tokyo Univ. of Technol., Hachioji, Japan), and Kazuaki Maeda (TOA Corp., Takarazuka, Japan)

The authors are developing a completely new direct-radiator loudspeaker as an alternative of the conventional electrodynamic loudspeaker. It is driven by continuous revolution of piezoelectric ultrasonic motors, It is useful for radiation of very low frequency signal because it shows almost flat phase frequency characteristics in low frequency region. A preliminary model, named dual-motor, de-spin (DMDS) model, included co-axial two ultrasonic motors. Stator of one motor is fixed to the base and of the other is connected to the cone radiator. Velocity modulation for any motor induces driving force for the cone radiator. Output sound at low frequency range (for example, 30–120 Hz) by this model was excellent because it has no significant resonance in this frequency region. However, its operation was occasionally instable. At this Congress, a highly improved model named quad-motor, de-spin (QMDS) model is presented. It uses two co-axial DMDS mechanisms. The experimental model has a cone radiator of 46 cm in diameter and an enclosure of 400 L. Its working frequency range is same as DMDS model. Harmonic distortions included in the output signal are improved to be less than 10%. Its sound quality is excellent.

2:40

1pEAb6. Influence of nonlinear parameters in Mirror filter to compensation performance of nonlinear distortions. Natsuki Uesako and Yoshinobu Kajikawa (Faculty of Eng. Sci., Kansai Univ., 3-3-35 Yamate-cho, Suita-shi, Osaka 564-8680 Japan, natsuki.uesako@gmail.com)

Mirror filter is used for the compensation of nonlinear distortions for electro-dynamic loudspeaker systems and is based on the nonlinear differential equations. The design of Mirror filter requires the estimated parameters of a target loudspeaker system. If you obtain the corresponding parameters of a target loudspeaker system and arrange Mirror filter designed using those parameters in front of the loudspeaker, then the nonlinear distortions can be compensated. Hence, the estimated parameters are very important to achieve high compensation performance. In this paper, we therefore examine the effects of the estimated parameters to the compensation performance. Concretely, we clarify the effects by varying each nonlinear parameter in Mirror filter. Simulation and experimental results demonstrate that the compensation performance for the second order nonlinear distortions depends on a nonlinear parameter of the force factor in loudspeaker systems.

3:00–3:20 Break

3:20

1pEAb7. A new loudspeaker for low frequency radiation by linear motion type piezoelectric ultrasonic actuators. Hiroya Saito (School of Comput. Sci., Tokyo Univ. of Technol., 1401-1,Katakura, Hachioji 192-0982, Japan, hirosaito12@gmail.com), Hirokazu Negishi (MIX Corp., Yokosuka, Japan), Juro Ohga (Shibaura Inst. of Technol./MIX Corp., Kamakura, Japan), Ikuo Oohira (Self-Employee, Yokohama, Japan), Kazuaki Maeda (TOA Corp., Takarazuka, Japan), and Kunio Oishi (School of Comput. Sci., Tokyo Univ. of Technol., Hachioji, Japan)

The authors had proposed new direct-radiator loudspeaker constructions with a conventional paper cone radiator driven by ultrasonic motors (USM), as a substitution for voice-coil motor. However, those models needed a revolution to linear motion conversion mechanism, and avoiding zero region non-linearity, like class A amplifier. These complications came from the conventional USM, since it is a rotational and having zero region non-linearity inherently. Here, the authors would propose a new mechanism by using new ultrasonic linear actuators, called longitudinal-bending multilayered transducers with independent electrodes (LBMTIE). The beauty of LBMTIE is linear and to control vertical motion and horizontal motion independently, hence zero region non-linearity avoided. Therefore, it is possible to substitute the voice-coil motor directly, which avoids the complicated mechanisms mentioned above. In this LBMTIE driven loudspeaker, vertical

movement voltage be fixed and horizontal voltage is driven by audio signal, like voice-coil motor. In addition, there is a big contrast against conventional voice-coil motor, which is a typical transducer, as its electrical input and sound pressure output are direct proportion each other. This is because LBMTIE driven loudspeaker may behave a sort of modulator, which is not direct proportion in between input electric power and output sound pressure level.

3:40

1pEAb8. Velocity control with class D amplifiers. Robert C. Randall (Ship and Torpedo Electron., Raytheon, 188 Hanover St. Apt 3, Fall River, Massachusetts 02720, bobrandall81@gmail.com) and David A. Brown (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Fall River, MA)

A SONAR array's radiation pattern is affected by the acoustic interactions, which may limit the effectiveness of beamforming algorithms when transmitting. A negative feedback system with a velocity sense signal fed back to the power amplifier can mitigate the array interactions proportional to the loop gain, and be effective across a broad frequency range without requiring *a priori* knowledge of the input signals. The tradeoffs between motional current velocity control and accelerometer based velocity control are summarized. Class D switching amplifiers can achieve greater than 90% efficiency and are increasingly being used to drive SONAR arrays. When a velocity control system is used with a Class D amplifier, feedback stability becomes a significant concern due to obtaining the feedback signal after the amplifier's LC output filter. The array equations are still obtained by converting the amplifier into a Thevenin equivalent force and output impedance, which now includes the amplifier's output filter and the synthesized impedance due to the velocity control loop. Sample beampatterns both with and without velocity control are shown, concluding that velocity control is well suited for use with complex dynamic transmit beamforming.

4:00

1pEAb9. A system for ultrasonic transmission of power and signal to an implanted hearing aid. Hugo Vihvelin, Jeffrey Leadbetter, Jeremy A. Brown, and Robert Adamson (School of Biomedical Eng., Dalhousie Univ., P.O. Box 15000, Halifax, NS B3H4R2, Canada, hugo.vihvelin@gmail.com)

We will report on development of a system for efficiently powering implanted hearing aids by transmitting ultrasonic acoustic energy across the skin. As compared to traditional magnetic induction coil power delivery systems, ultrasound-based systems offer a more compact form factor for the same power handling capability and lower electrical loss. Part of the challenge of building such a system for implanted hearing aids is developing efficient modulation and demodulation electronics that can deliver both electrical power and an acoustic frequency signal to the implanted device. We present the design and implementation of an amplitude modulated system in which the power is delivered on the carrier and signal in the modulation sidebands. The transmitter consists of an efficient PWM encoder driving an LC resonator tuned to the ultrasound transducer resonance frequency. The receiver consists of an efficient rectifying demodulator that provides supply voltages to internal electronics as well as the acoustic signal. Power loss mechanisms, form factor considerations, linearity, and overall system performance will be discussed.

4:20

1pEAb10. The design of ultrasonic lead magnesium niobate-lead titanate composite transducers for power and signal delivery to implanted hearing aids. Jeff Leadbetter, Jeremy Brown, and Rob Adamson (Dalhousie Univ., 1276 South Park St., Rm. 3189, Dickson Bldg., VG Site, Halifax, NS B3H 2Y9, Canada, jeff.leadbetter@dal.ca)

We present a system for efficiently powering implanted hearing aids by transmitting an ultrasonic signal across the skin. The use of ultrasound as method for power and signal transfer is known for embedded systems in industrial applications and has more recently been investigated for use with other medical implants. In our application, ultrasonic transducers are investigated as they offer substantially reduced size relative to traditional magnetic induction coil power delivery. The developed transducers use lead magnesium niobate-lead titanate (PMN-PT) piezoelectric material in a 1–3 composite formulation. PMN-PT offers an electromechanical coupling factor (kt, an indicator of maximum efficiency) that is up to 60% greater than traditional piezoceramics, while the use of composite transducers removes geometric constraints that can limit the achieved efficiency. The fabrication methods for the transducers are detailed. Experimental results are presented to show the composite transducers achieve a kt of 0.86 (out of 1.00), and a power transmission efficiency that is improved by 38% relative to a similar non-composite transducers. It is also demonstrated that these transducers offer sufficient bandwidth for amplitude or frequency modulation schemes to transmit data signals along the power carrier beam.

4:40

1pEAb11. Biocompatible wireless power transferring and charging based on ultrasonic resonance devices. Sung Q Lee, Woosub Youm, and Gunn Hwang (Nano Convergence Sensor, Electron. and Telecommunications Res. Inst., 161 gajungro yousung, Daejon 305-350, South Korea, hermann@etri.re.kr)

To increase application area of implantable devices for medical treatment including implantable cardiac defibrillator or deep brain stimulator, the rechargeable battery module is highly requested. The previous Li-type battery has limited current sources, so that the patient is forced to have surgery just for changing battery. Previous technologies such as magnetic resonance and induction coupling have limited applications because of its short transfer distance compared to device size and magnetic field intensity limitation for the safety of body exposure. As an alternative, the biocompatible wireless power transferring and charging technology is proposed using ultrasonic resonance devices. For the high efficient power transferring, optimal transfer frequency is calculated based on the acoustic radiation and damping effect. Then, the optimal load resistance is selected for matching power condition in receiver. And, transmitter is designed to match the optimal transfer frequency. The ultrasonic resonance transmitter and receiver are manufactured with the size of 20 mm diameter, 6.0 mm height. The energy conversion efficiency from input electrical power of transmitter and output power of receiver is about 25.6% at 10 cm distance, experimentally. The maximum transferring power is up to 15 mW. This result is quite high considered with the device size and the power transfer distance.

Session 1pMU

Musical Acoustics and Psychological and Physiological Acoustics: Player/Instrument Coupling

Gary Scavone, Cochair
McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada

Tamara Smyth, Cochair
Music, Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093

Invited Papers

1:00

1pMU1. A morphological and acoustic study on the effect of a trumpet player's vocal tract. Tokihiko Kaburagi (Grad. School of Design, Kyushu Univ., Shiobaru 4-9-1, Minami-ku, Fukuoka 815-8540, Japan, kabu@design.kyushu-u.ac.jp), Naoyuki Yamada (Nagareyama City Hall, Nagareyama, Japan), Takashi Fukui (Shikumi Design Co., Ltd., Fukuoka, Japan), and Eriko Minamiya (Yamaha Corp., Hamamatsu, Japan)

A morphological and acoustic study is presented to examine the role of the vocal tract in playing the trumpet. Preliminary results obtained from one professional player are shown, and the effectiveness of the method is demonstrated. Images of the vocal tract with a resolution of 0.5 mm (2 mm in thickness) were recorded with magnetic resonance imaging to observe tongue posture and to estimate the vocal-tract area function during actual trumpet performance. The input impedance was then calculated for the player's air column, including both the supra- and sub-glottal tracts, using an acoustic tube model that also considers wall losses. Finally, a time-domain blowing simulation was performed with a lip vibration model (Adachi and Sato, J. Acoust. Soc. Am. **99**, 1200–1209, 1996). In this simulation, the oscillating frequency of the lips was slightly affected by using different shapes of the vocal tract measured for the player. In particular, when the natural frequency of the lips was gradually increased, the transition to higher modes occurred at different frequencies for different vocal-tract shapes. Furthermore, simulation results showed that the minimum blowing pressure required to attain lip oscillation can be reduced by properly adjusting the vocal-tract shape.

1:20

1pMU2. Simulating different upstream coupling conditions on an artificial trombone player system using an active sound control approach. Vincent Fréour (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., Music Technol. Area, Schulich School of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, vincent.freour@mail.mcgill.ca), Thomas Hélie, Nicolas Lopes, René Caussé (IRCAM - CNRS UMR 9912, UPMC, Paris, France), and Gary P. Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., Montreal, QC, Canada)

Recent research suggests that the ability to finely tune vocal-tract resonances during trombone playing may constitute an important aspect of performance expertise. Artificial player systems, designed to reproduce the behavior of a real player, often neglect this component by not providing any control of upstream resonances. However, they offer great experimental platforms for quantitative studies on sound production mechanisms, allowing independent adjustment of certain control parameters. An active sound control method was designed to improve high tone support and investigate different conditions of coupling between the artificial lips, the downstream air-column, and the upstream cavity during sustained tones played by an artificial valve-trombone player system. Upstream input impedance at the fundamental frequency was controlled through real-time adjustment of the phase and amplitude ratio between the acoustic pressure generated on both sides of the lips. The phase difference between the upstream and downstream pressures was swept linearly while maintaining different conditions of upstream energy and fixed trombone fingering. Observations during this procedure included: (1) significant fundamental frequency variations in the neighborhood of a downstream impedance peak; and (2) variation of the downstream energy and optimal phase tuning with regard to the mechanical efficiency of the lip-valve system suggested at the energy maximum.

1:40

1pMU3. Saxophone modeling and system identification. Tamara Smyth (Music, Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, trsmyth@ucsd.edu) and Marjan Rouhipour (Computing Sci., Simon Fraser Univ., Vancouver, BC, Canada)

In this work, saxophone instrument frequency responses are estimated at both the mouthpiece corresponding to the input impedance and outside the bell, using acoustic measurement and post signal processing. The measurement technique is based on one previously developed for measuring the acoustic properties of instrument bells but is adapted to account for the fact that the saxophone bell does not easily separate from the instrument bore and must be measured as a single unit. Furthermore, measurements are taken of the instrument configured with all possible fingerings covering the playable range of the B-flat tenor saxophone, and instrument reflection and transmission functions are estimated for, and applied to a waveguide model of, each tone-hole configuration. Having the instrument frequency responses at both the mouthpiece and the bell for every possible fingering allows for an improved parametric synthesis, but also allows for saxophone system identification, inverse modeling, and estimation of player-instrument control parameters during real-time performance.

2:00

1pMU4. Measuring lips control on flute-like instruments using active vision. Benjamín Carriquiry, Patricio de la Cuadra (Centro de Investigación en tecnologías de audio, Music department, Pontificia Universidad Católica de Chile, Jaime Guzman Errazuriz 3300, Providencia, Santiago 07866, Chile, pcuadra@uc.cl), and Benoit Fabre (LAM-IJLRA, Université Pierre et Marie Curie (Paris 6), Paris, France)

Flute-like instruments operate under a feedback system between an air jet and a resonator. The characteristics of the air jet and its interaction with the resonator are defined by construction in some flute-like instruments such as the organ pipes, or completely determined by the musician's control like in the Shakuhachi, panpipe, or transverse flute. In this article, a 3D lips detection system based on active vision using laser grid triangulation has been designed and implemented. Simple musical gestures from transverse flute players have been measured and analyzed, outlying the strategies used to coordinate the parameters available, such as lips to labium distance, jet speed, and jet shape, to orchestrate an adequate sound. The analysis is presented in a non-dimensional representation capable of generalizing to other members of the flute family. The measurement system developed is compared with previous strategies and reveals a promising tool to further understand the complexity of the human control in this popular family of instruments.

2:20

1pMU5. Analysis of bow-change strategies. Knut Guettler (Retired, Eilins vei 20, Jar 1358, Norway, kg@knutsacoustics.com)

One of the most important skills of the accomplished bowed-string player is the smooth bow change. Smooth changes are often necessary in order to keep a phrase flowing, and equally important in situations where the bow is too short for the duration of the given note, the latter requiring a bow change of least possible audibility. The problem arises from the fact that a change of bowing direction requires the rotation of the Helmholtz corner to be reversed, and the phases of the string-velocity frequency components thus to be shifted 180°. In between the two states, there exists no transition that can fully maintain the sound flow without introducing undesirable noises. However, by choosing the right bowing strategy and gesture, the tradeoff between transition time and noise content can be optimized for the purpose. In practice, different players solve this problem in a number of ways. The present study, which is mainly based on numeric simulations, analyzes the sounding outcome of a variety of possible bowing parameters.

2:40

1pMU6. Perception and production of complex bowing movements in violin performance. Erwin Schoonderwaldt (Inst. of Music Physiol. and Musicians' Medicine, Hanover Univ. of Music, Drama and Media, Emmichplatz 1, Hannover, NDS 30175, Germany, schoondw@kth.se), Matthias Demoucron (Inst. for Psychoacoustics and Electron. Music, Ghent Univ., Ghent, Belgium), Eckart Altenmüller (Inst. of Music Physiol. and Musicians' Medicine, Hanover Univ. of Music, Drama and Media, Hanover, Germany), and Marc Leman (Inst. for Psychoacoustics and Electron. Music, Ghent Univ., Ghent, Belgium)

In bowed-string instruments, the primary function of bowing movements is to control the parameters that govern the stick-slip interaction between the bow and the string, giving the performer control of the sound. Not less importantly, bowing movements have to be planned ahead in order to anticipate future events. In fast, repetitive bowing movements involving string crossings and bow changes the primary and anticipatory movements become integrated, forming an overall, in the simplest case circular movement pattern. The relative timing of string crossings and bow changes is an inherent property of the shape of these patterns, which therefore has an important influence on the quality of the note transitions. We will present two complementary studies that provide insight in this coordination phenomenon. A perceptual study has been conducted using a virtual violin, in which the participants could influence the relative timing between string crossings and bow changes by a simple slider, giving insight in the perception of such transitions and typical temporal constraints. Analyses of bowing movements show in detail how the coordination is realized in performance, and how the performer adapts her/his movement patterns to performance constraints, such as tempo and dynamic level.

3:00–3:20 Break

3:20

1pMU7. Time-domain simulation of the bowed cello string: Dual-polarization effect. Hossein Mansour (Music, McGill Univ., Ste. 500- 550 Sherbrooke o, Montreal, QC H3A 1B9, Canada, hossein.mansour@mail.mcgill.ca), Jim Woodhouse (Engineering, Cambridge Univ., Cambridge, United Kingdom), and Gary P. Scavone (Music, McGill Univ., Montreal, QC, Canada)

A detailed time-domain simulation is implemented to model the bowed cello string. Building on earlier simulation models, several new features have been added to make the model more realistic: in particular, both polarizations of the string motion are included, as well as the longitudinal vibrations of the bow hair. These additional features can be turned on and off in the model to evaluate their relative importance. In all previous simulations, the bow-hair was assumed stiff enough to suppress any motion of the string perpendicular to the bowing direction. High-speed video recordings, on the contrary, have suggested that the amplitude of this motion is not negligible compared to the motion of the string in the bowing direction. The major source of this motion is tracked down to the X-Y coupling through the bridge. Although this extra dimension of vibration may not necessarily contribute much to the radiated sound by itself, it can modulate the effective bow force and hence affect the stick-slip motion of the string. The longitudinal vibration of the bowhair is also included in our model. The compliance of the bowhair was accounted for in previous studies in a crude way, but without enough detail to capture the difference between different bows.

3:40

1pMU8. Characterization of bowing strokes in violin playing in terms of controls and sound: Differences between bouncing and on-string bow strokes. Alfonso Perez Carrillo (Schulich School of Music, McGill Univ., 555 Sherbrooke West, Montreal, QC H3A, Canada, alfonso.perezcarrillo@mail.mcgill.ca)

Bowing is the main element in sound production during a violin performance and one of the most basic and important expressive resources for the musician. In the lowest level, control parameters such as force, velocity, or bow-bridge distance are directly determining the characteristics of the sound. In a higher level, bowing strokes constitute one of the main mechanisms for structuring the

performance. There are many different kinds of bowing strokes, and they are commonly classified into on-string, if the attack happens with the bow on the string and off-string, if the bow is bouncing. From a database of violin performances containing multimodal data including sound and gestures, a set of spectral features and instrumental controls is extracted and the database is segmented into intra-note segments (attack, sustain, and release). A characterization of bowing strokes and a comparison between bouncing and on-string strokes in terms of bowing controls and sound at the intra-note segments is presented.

4:00

1pMU9. Analysis/synthesis of bowing control applied to violin sound rendering via physical models. Esteban Maestre (Music Technol. Group, Roc Boronat 138, Barcelona 08018, Spain, esteban.maestre@upf.edu)

A prominent challenge in instrumental sound synthesis is to reproduce the expressive nuances naturally conveyed by a musician when controlling a musical instrument. Despite the flexibility offered by physical modeling synthesis, appropriately mapping score annotations to sound synthesis controls still remains an interesting research problem, especially for the case of excitation-continuous instruments. Here we present our work on modeling bowing control in violin performance, and its application to sound synthesis via physical models. Minimally invasive sensing techniques allow for accurate acquisition of relevant timber-related bowing control parameter signals. The temporal contours of bowing control parameters (bow velocity, bow force, and bow–bridge distance) are represented as sequences of low-order polynomial curves. A database of parametric representations of real performance data is used to construct a generative model able to synthesize bowing controls from an annotated score. Synthetic bowing controls are then used to render realistic performances by driving a violin physical model.

Contributed Papers

4:20

1pMU10. On the relation between gesture, tone production, and perception in classical cello performance. Magdalena Chudy (Ctr. for Digital Music, School of Electron. Eng. and Comput. Sci., Queen Mary Univ. of London, Mile End Road, London E1 4NS, United Kingdom, magdalena.chudy@eecs.qmul.ac.uk), Alfonso Pérez Carrillo (Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., Montreal, QC, Canada), and Simon Dixon (Ctr. for Digital Music, School of Electron. Eng. and Comput. Sci., Queen Mary Univ. of London, London, United Kingdom)

On bowed string instruments such as violin or cello, the quality of sound depends mostly on the performer's bowing technique, which determines the interaction between the bow hair and the string. An accomplished string player has numerous ways of shaping the spectrum of a desired sound. This research investigates the combination of bowing gestures necessary for production of a rich tone. In particular, bowing control parameters such as bow force, bow velocity, and bow–bridge distance captured by a dedicated sensing system are analyzed and compared against audio features. Using audio and gesture measurements of six advanced cellists recorded on two different instruments of a luthier class, we characterize a sound palette and respective bowing control patterns of each player in performed music excerpts and scales. We especially focus on how performers adjust their bowing technique to control the timber of an instrument on which they have never practiced before. Observed differences between the players on the measured audio features show consistency with the bowing parameters adapted for balancing the timbral changes due to instrument, string, and fingering position. To perceptually evaluate the recorded samples, expert musicians were asked to rank the players in terms of sound quality and tone richness.

4:40

1pMU11. Differences in technique and sound in beginner and expert cello performances and use of acoustic information to provide support for performance techniques. Taichi Sato, Shoichi Miyagawa, and Hiromi Yamatani (Tokyo Denki Univ., Adachi-ku, 5 Senju-Asahi-cho, Tokyo 120-8551, Japan, taichi@mail.dendai.ac.jp)

We studied the differences in performance between beginners and expert musicians by taking cello performance as the object of our research. By principal component analysis of the beginning part of a performance—the part when a performer just begins to play—we were able to distinguish between performances by beginners and those by experts. We showed that the process of mastery of an instrument, in which the sound of a beginning performer becomes better, can be evaluated using principal component analysis. We studied the characteristics of the ability to skillfully draw the bow across the cello strings, and we created acoustic information based on these characteristics. By giving this acoustic information to beginners, the beginners learned to become almost as skillful as expert cellists in their ability to draw the bow skillfully across the strings, even though they were only beginners.

Session 1pNSa

Noise: Advanced Hearing Protection and Methods of Measurement II

Jeremie Voix, Cochair
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Elliott H. Berger, Cochair
Occupational Health & Env. Safety Div., 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650

Invited Papers

1:00

1pNSa1. Improving speech intelligibility in active hearing protectors and communication headsets with subband processing.
Anthony J. Brammer, Gongqiang Yu, Eric R. Bernstein, Martin G. Cherniack, and Donald R. Peterson (Ergonomics Technol. Ctr., Univ. of Connecticut Health Ctr., 263 Farmington Ave., Farmington, CT 06030-2017, brammer@uchc.edu)

Parallel subband processing, in which the full bandwidth of environmental noise and a communication channel are processed separately in contiguous, restricted frequency bands, has been proposed as a means to improve speech intelligibility in noise for active hearing protectors and communication headsets (Bernstein *et al*., Int. J. Ind. Ergonom., in press). An active, adaptive feed-forward control structure has been employed to improve the audibility of sounds in the communication channel of a circumaural hearing protector / headset while active noise reduction (ANR) is used to complement the passive attenuation of the device from 50 to 800 Hz. The communication channel subbands have been implemented as octave bands from 125 Hz to 8 kHz. The performance of the device has been evaluated in a diffuse field when worn by human subjects. Word intelligibility in industrial noise was evaluated when the active system was not operating, when the device was operating as a fullband ANR system with fixed communication channel gain, and as a subband ANR system with adaptive gain of the communication channel signal to improve the speech signal-to-noise ratio. A significant improvement in speech intelligibility can be obtained with the subband system. [Work supported by NIOSH 5R01OH 008669.]

1:20

1pNSa2. Advanced hearing protection and auditory awareness in individuals with hearing loss.
Christian Giguere, Chantal Laroche, and Véronique Vaillancourt (Audiol./SLP Program, Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H8M5, Canada, cgiguere@uottawa.ca)

In-ear and earmuff-type electronic protection devices are rapidly being introduced into the marketplace and deployed in noisy industrial workplaces and military settings. In these environments, workers must be sufficiently protected from noise while being able to maintain good communication abilities and situational awareness. Features such as level-dependent attenuation or amplification, user-adjustable talk-through circuitry, noise reduction, and remote communication capabilities are commonly found in advanced devices. The benefits of these features depend on their complex interaction with the signal and noise characteristics, the hearing status and language proficiency of the workers, and the nature of the auditory task. Yet, detailed electro-acoustical specifications are rarely reported by manufacturers. Measurement standards are also lacking, though the advent of ANSI/ASA S12.42 should in part address this situation. In this paper, the characteristics of two advanced devices are reported over a test battery including measurements of the passive sound attenuation, level-dependent talk-through sound transmission, and speech recognition in noise for listeners with a wide range of hearing profiles. Electronic level-dependent attenuation provided superior speech recognition performance than passive attenuation for all groups of listeners, and often exceeded unprotected speech recognition performance.

1:40

1pNSa3. Supplemental text messaging for the resolution of auditory overload.
Sharon M. Abel (Individual Behaviour and Performance Section, Defence R&D Canada - Toronto, 1133 Sheppard Ave. W., Toronto, ON M3K 2C9, Canada, sharon.abel@drdc-rddc.gc.ca), Geoffrey Ho (Human Systems Integration Section, Defence R&D Canada - Toronto, Toronto, ON, Canada), Ann Nakashima, and Ingrid Smith (Individual Behaviour and Performance Section, Defence R&D Canada - Toronto, Toronto, ON, Canada)

Military signal operators listen, transcribe, and respond to audio traffic over multiple audio channels, in high-level noise from vehicles and weapons. The messages typically overlap in time and may be difficult to disentangle. Two studies were carried out to determine the benefit of supplemental texting. Normal-hearing participants were tested in a mock up of a military command post. Brief messages were played simultaneously over a communications headset (dichotic) and a loudspeaker array, either in quiet or in a background of vehicle noise. The at-ear speech-to-noise ratio was 5 dB. Only those messages beginning with a pre-assigned call sign were encoded. Mean scores of 84% or better were observed for messages presented over the headset, although there was a clear right ear advantage in noise. Messages coming over the loudspeakers were more difficult to understand but a visual cue directing attention to the source of an incoming targeted message resulted in a significant improvement of 7%. Replacing audio messages over the loudspeakers in noise with visual or audiovisual presentations resulted in an improvement from 71% to 96% that did not negatively affect headset performance. The data suggest that texting is a viable option for communication in cases of degraded audio.

Contributed Paper

2:00

1pNSa4. Investigation of the role for noise canceling headphones to assist speech recall in noise. Marion Burgess and Brett Molesworth (School of Eng. and Information Technol., Univ. NSW, Canberra, Northcott Dr., Canberra, ACT 2612, Australia, m.burgess@adfa.edu.au)

There are many situations where it is necessary to hear, understand and be able to recall spoken information in less than ideal listening conditions. For example within an aircraft cabin where, despite improvements in modern passenger aircraft, noise generated from aircraft engines and aerodynamic airflow make it difficult to hear important on-board safety announcements such as the preflight safety brief. The benefits of headphones that incorporate active noise control in such environments are the focus of a series of research studies. In this paper, we discuss the techniques developed to investigate the use of active noise control headphones on the intelligibility and recall of speech generated outside the headsets in noise typical of that in a commercial aircraft cabin. The initial studies were directed towards assessing the effects on the recall for safety announcements. These studies have been extended to investigate if there are any benefits for those for which English is a second language. The results suggest that the use of active noise control headphones can minimize communication errors in a range of situations and this paper will discuss the methodology adopted and summarize the outcomes.

Invited Papers

2:20

1pNSa5. Integration of a distance sensitive wireless communication protocol to hearing protectors equipped with in-ear microphones. Rachel E. Bou Serhal, Tiago Falk, and Jérémie Voix (Universite de Quebec, Ecole de Technologie Superieure, 1280 Rue Saint Marc Apt. PH3, Montreal, QC H3H 2G1, Canada, rachel.bou.serhal@etsmtl.ca)

Using radio communication in noisy environments is a practical and affordable solution allowing communication between workers wearing hearing protection devices (HPD). However, typical radio communication systems have two main limitations when used in noisy environments: first, the background noise is disturbing the voice signal picked-up and transmitted, and second, that voice signal goes to all listeners on the same radio channel regardless of their physical proximity. A new concept of a so-called "Radio-Acoustical Virtual Environment" (RAVE) addressing these two issues is presented. Using an intra-aural instantly custom molded HPD equipped with both an in-ear microphone and miniature loudspeaker, undisturbed speech is captured from inside the ear canal and transmitted over the wireless radio to the remote listener. The transmitted signal will only be received by listeners within a given spatial range, such range depending on the user's vocal effort and background noise level. This paper demonstrates the technological challenges to overcome and the methodology involved in the implementation of RAVE.

2:40

1pNSa6. Sensorial substitution system from vision to audition using transparent digital earplugs. Damien Lescal, Jean Rouat (GEGI, Université de Sherbrooke, Sherbrooke, QC, Canada), and Jérémie Voix (Génie mécanique, École de Technologie Supérieure, 1100, rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, jeremie.voix@etsmtl.ca)

Since the Tactile Vision Substitution System (TVSS) developed by Bach-Y-Rita in 1960's, several sensorial substitution systems have been developed. In general, the so-called "sensorial substitution" system transform stimuli characteristic of one sensory modality (for example, vision) into stimuli of another sensory modality (for example, audition). These systems are developed to help handicapped persons. We developed a sensorial substitution system from vision to audition. An artificial neural network is used to identify the important parts in the image. The virtual acoustic space technic is used to generate localizable sounds. A sound is associated to each important parts of the image. The entire real-time system has been implemented on iOS platforms (IPhone/IPad/IPod Touch). We associated our system with transparent digital earplugs. This way the user is aware of the audio scene happening around him. The system has been tested on non-blind persons and the results are presented.

3:00–3:20 Break

3:20

1pNSa7. An active hearing protection device for musicians. Antoine Bernier and Jérémie Voix (École de Technologie Supérieure, 1100 Rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, antoine.bernier@ens.etsmtl.ca)

Professional musicians have to deal with two great problems when wearing hearing protection devices (HPDs): the occlusion effect and the isolation effect. The occlusion effect is an unnatural and annoying perception of one's own voice when wearing HPDs. It will affect all musicians whose instrument induces vibrations to the skull, including a singer's vocal tracts and instruments mechanically coupled to the head, such as a trumpet or a violin. The isolation effect is the unnatural sensation of being isolated from a given sound environment. It is caused by a non-uniform attenuation of the HPD over the frequency spectrum and the absence of compensation for psychoacoustic factors, such as uneven loudness perception. These two effects cause a shift of perception between the musician's perception and the audience's perception and therefore compromise the musician's ability to offer a good performance to his audience. This paper presents the design and implementation of an active musician's HPD featuring a feedback active noise control system for occlusion effect reduction as well as psychoacoustic compensations for isolation effect reduction. The proposed test procedure and preliminary performance assessments are presented to validate both the test procedure and the system for future subjects trials on a larger scale.

3:40

1pNSa8. A case-study on the continuous use of an in-ear dosimetric device. Kuba Mazur and Jeremie Voix (Universite du Quebec, Ecole de Technologie Superieure, 1100, rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, kuba.mazur@etsmtl.ca)

In order to further understand the combined effects of occupational and recreational noise exposure with regards to noise induced hearing loss (NIHL), an in-ear dosimeter prototype meant for continuous use was developed. The device acts as a hearing protection device (HPD) and can measure and log effective in-ear sound pressure level as well as unprotected levels. To enable its continuous use, this HPD is also equipped with a bypass feature for "transparent" hearing, input for music or communication devices and interfaces with Android smartphones. The proposed device allows for the implementation of an algorithm accounting for the auditory fatigue recovery rate, providing a true representation of the current accumulated noise dose. This allows for 24 h dosimetry and avoids having the user manually reset the dose back to 0% on the next day and thus assuming complete fatigue recovery has occurred. This paper details the proposed recovery algorithm, presents collected field data, and discusses the benefits as well as real-world challenges of using such a device.

4:00

1pNSa9. Estimation of noise exposure level for subjects wearing hearing protector devices. Cécile Le Cocq (Génie mécanique, École de technologie supérieure, 1100 Notre Dame Ouest, Montréal, QC H3C1K3, Canada, cecile.lecocq@etsmtl.ca), Hugues Nélisse, Jérôme Boutin (Prévention des risques mécaniques et physiques, Institut de recherche Robert-Sauvé en santé et en sécurité du travail, Montréal, QC, Canada), Jérémie Voix, and Frédéric Laville (Génie mécanique, École de technologie supérieure, Montréal, QC, Canada)

Industrial workers are exposed to noise levels that could damage their hearing. The Field-MIRE (F-MIRE) method has been developed to quantify earplug and earmuff attenuations with two microphones located under and outside of the HPD. This technique has been designed to be used in the field, but does not give a direct access to the noise exposure level, that is, the noise level at the head location without the subject. In this article, we present a combination of the F-MIRE method with a proposed technique to estimate the sound pressure level without subject, in order to quantify both the ambient and protected noise exposure levels and deduce the effects on worker hearing. Several experiments have been conducted on four subjects with three types of earplugs and with five types of earmuffs. First, the best location for the microphone outside of the HPD has been determined. Second, correction factors that need to be applied on the outside microphone measurement to estimate the sound pressure level without subject have been quantified. Finally, the proposed technique has been validated with measurements taken in a simulated workplace.

4:20

1pNSa10. Improved hearing conservation in industry: More efficient implementation of distortion product otoacoustic emissions for accurate hearing status monitoring. Annelies Bockstael, Hannah Keppler, and Dick Botteldooren (Ghent Univ., Sint-Pietersnieuwstraat 41, Gent 9000, Belgium, annelies.bockstael@intec.ugent.be)

Preventing occupational hearing damage requires close monitoring of workers' hearing. Implementing Distortion Product Otoacoustic Emissions (DPOAEs) in-field is a sensitive and feasible approach provided that a combination of minimal measuring time and infrequent false-positives—i.e., cases where elevated background compromises DPOAEs—is found. This paper investigates how measurement time can be reduced by carefully selecting the tested frequency span and resolution, and how false-positives are minimized by comparing DPOAEs acquired in noise with DPOAEs previously obtained in optimal test conditions. To test this, DPOAEs have been registered with a 1/8-octave band resolution from 841 Hz to 8 kHz for 60 subjects, in quiet conditions and in white noise levels ranging from 54 dB(A) to 90 dB(A). Measurement accuracy is confronted to decrease the measurement time as a function of frequency resolution and range. Diagnostic importance and sensitivity to background noise is addressed for different frequency regions. Within-subject variation of DPOAEs in noisy conditions is assessed both between different noise conditions and between subsequent probe placement. Obtained test–retest statistics quantify normal variability and allow within normal working routines to select for further investigation persons with DPOAEs falling outside this range.

4:40

1pNSa11. Use of passive hearing protectors and adaptive noise reduction for field recording of otoacoustic emissions in industrial noise. Vincent Nadon (École de technologie supérieure, 6080 rue Laurendeau, appartement 2, MONTREAL, QC H4E3X5, Canada, vincent.nadon.1@ens.etsmtl.ca), Annelies Bockstael (INTEC, Ghent Univ., Ghent, Belgium), Hannah Keppler (Dept. of Oto-Rhino-Laryngology and Logopaedic-Audiol. Sci., Ghent Univ., Ghent, Belgium), Dick Botteldooren (INTEC, Ghent Univ., Ghent, Belgium), Jean-Marc Lina, and Jérémie Voix (École de technologie supérieure, Montreal, QC, Canada)

Distortion product otoacoustic emissions (DPOAEs) can detect noise-induced hearing loss in-field, but their data extraction is very sensitive to background noise. This paper investigates how passive and active noise reduction enhance DPOAE recording based on data collected in white noise from 54 dB(A) to 90 dB(A). Despite considerable high-frequency attenuation from a proper placed DPOAE probe, 54 dB(A) background noise deteriorates the test outcome substantially. More low-frequency attenuation by an extra passive earmuff enables measurements in white noise levels of 70 dB(A). The relationship between external sound level and noise recorded by the DPOAE system has been statistically modeled. Additionally, the upper limits of attenuation improvement are analyzed by quantifying residual physiological noise. Furthermore, for an earplug integrating microphone and speakers of the DPOAE measurement probe, adaptive noise reduction processing on the DPOAE signal is used to improve the signal-to-noise ratio. The adaptive noise reduction (ANR) is implemented using the NLMS algorithm to filter out the ambient noise, measured by the first microphone measuring the DPOAE signal, with a second miniature microphone mounted flush with the external faceplate of the isolating DPOAE probe. Simulated data show that DPOAE response extraction is possible in an environment with noise levels exceeding 70 dB(A).

Session 1pNSb

Noise: Community Noise

Eric L. Reuter, Chair
Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801

Invited Papers

1:00

1pNSb1. Quantifying the ambient community noise environment for optimal industry siting. Tim C. Wiens, Gordon L. Reusing, Slavi Grozev, and Zachary Zehr (Noise and Vib. Services, Conestoga-Rovers & Assoc., 651 Colby Dr., Waterloo, ON N2V 1C2, Canada, twiens@craworld.com)

A road traffic noise model was developed to approximate the ambient noise levels present within a 200 km^2 urban center. Road corridors that included highways, city streets, and country side-roads with varying traffic volumes were modeled to evaluate the existing ambient conditions within the project area. To calculate noise levels, an acoustical model and Traffic Noise Model (TNM) calculation standard was used to account for a variety of real-world variables such as daily average traffic counts, turning counts, speed limits, road composition, elevation, road width, and traffic composition. The model generated noise contours that were used to identify areas of elevated ambient noise levels within the project area that may prove suitable for a medium-sized industrial facility. This modeling technique and ambient community noise analysis allowed for the identification of an optimal site within the project area and also proved to be an approach that can be used to industry's advantage. Urban noise is an emerging issue for growing communities. Locating new facilities within urbanized areas with elevated ambient conditions may minimize community noise impacts, reduce post-construction noise abatement costs, and ultimately promote complementary adjacent land use and sustainable urban densification.

1:20

1pNSb2. Proposed standard—Guidance for developing state noise regulations and local noise ordinances. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com)

The American National Standards Institute (ANSI) Accredited Standards Committee S12 (Noise) Working Group (WG) 41 has been developing a draft standards document for over 12 years. The current document is in Draft #9, now under development. This document represents the consensus of many stakeholders in the community noise arena, including industry, government, consulting, and the public. The purpose of the document is to provide guidance to government officials, acoustical consultants, and other interested persons on how to develop a community noise ordinance or regulation, which is appropriate for the existing local circumstances. The document addresses issues such as public and government priorities and values, and available resources, and also provides the technical basis to manage the local sound environment. The keys to the effectiveness of the document are that it provides a menu of options for the user, discusses the trade-offs involved for decisions that must be made by government officials, and emphasizes that enforcement of a community noise ordinance is crucial to its success. A description of the current draft is presented.

Contributed Papers

1:40

1pNSb3. Do recent findings on jet noise answer aspects of the Schultz curve? Micah Downing (Blue Ridge Res. and Consulting, 15 W. Walnut St., Ste. C, Asheville, NC 28801, micah.downing@blueridgeresearch.com), Kent Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Sally Anne McInerny (Dept. of Mech. Eng., Univ. of Louisiana at Lafayette, Lafayette, LA), Tracianne Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Michael James (Blue Ridge Res. and Consulting, Asheville, NC)

Recent research efforts on nonlinear propagation from high performance jet aircraft have revealed an interesting challenge to predicting community response. This challenge focuses on receiver perception of these unique acoustical signals, which contain acoustical shocks that appear to increase their relative loudness and/or noisiness. This current finding suggests a need for an improved description of a receiver perception of the loudness of these signals in order to improve the assessment of noise impacts from these aircraft. Looking backwards, an interesting question emerges: did the earlier low bypass jet engines on commercial and transport aircraft also include these acoustical shocks? If they did contain these features, then the perceptual differences observed between aircraft and other transportation noise sources may be partially explained.

2:00

1pNSb4. Acoustical indicator of noise annoyance due to tramway in in-curve operating configurations. Arnaud Trollé, Catherine Marquis-Favre, and Achim Klein (Lab. of Bldg. and Civil Eng., Univ. of Lyon, Labex CELYA, National School of State Public Works, Rue Maurice Audin, Vaulx-en-Velin 69518 CEDEX, France, arnaud.trolle@entpe.fr)

Tramway gives rise to annoyance. Particularly, tramway in an in-curve operating configuration often emits squeal noises leading to inhabitants' complaints. Noise exposure levels were not sufficient to account for annoyance. Other acoustical factors could explain noise annoyance. A laboratory experiment is carried out in order to identify and characterize the influential acoustical factors of in-curve tramway noises. Subjects are asked to rate the short-term annoyance caused by 14 tramway pass-by noises, recorded *in situ* for various in-curve operating configurations. A psychoacoustical analysis shows that the overall perceived noise level, the irregular character and the treble character of tramway noises influence noise annoyance. These

acoustical features are taken into account through the following indices: the mean loudness, the variance of time-varying A-weighted pressure normalized by RMS A-weighted pressure, and a psychoacoustical index, constructed to account for squeal noise, expressed by the total energy of the tonal components within critical bands from 12 to 24 Barks. A multilevel regression analysis reveals that a combination of these indices proves to be satisfactory for predicting short-term annoyance due to in-curve tramway noises. These results are consistent with those from a previous experiment that implied 61 tramway pass-by noises corresponding to different operating situations.

2:20

1pNSb5. Rail noise and vibration in Australia—A case study. Vincent Chavand (Air & Noise Service Line, GHD, Level 3, GHD Tower, 24 Honeysuckle Dr., Newcastle, NSW 2300, Australia, vincent.chavand@ghd.com)

This paper reviews the various stages of a major rail project undertaken in New South Wales (NSW), Australia, between 2010 and 2012. This case study involves 40 km of new track adjacent to existing railway lines in the Hunter Valley, NSW. The project is located within a mixture of rural and urban settings and had the potential to impact on a large number of sensitive receivers during both the construction and operational phases. The project approvals required compliance with a number of relatively new noise and vibration guidelines and policies, which provides an opportunity for the author to reflect on the recent evolutions in noise and vibration control practices and policing in Australia. This paper reviews the project from the approvals process to its commissioning phase from a noise and vibration point of view. It explores the construction and operational noise modeling methodologies, reviews the design process and adopted mitigation measures and, in doing so, it discusses the practical challenges met through the course of the works.

2:40–3:00 Break

3:00

1pNSb6. Noise pollution in urban settings of the Western Amazonia and an approach to cope with. Stephan Paul (Undergrad. Program Acoust. Eng., Fed. Univ. of Santa Maria, Av. Roriama 1000, Camobi, Santa Maria, RS 97105-900, Brazil, stephan.paul@eac.ufsm.br), Isabel C. Kuniyoshi (Dept. of Speech and Hearing Sci., Faculdade Sao Lucas, Porto Velho, Brazil), Flávio André M. de Araújo (Office of the Federal Public Prosecutor of Rondonia State, Porto Velho, Brazil), and Lucinara Camargo (Dept. of Environmental Protection SEMA, Municipality of Porto Velho, Porto Velho, Brazil)

The Amazon Region is usually well known for its nature and not for the noise pollution that recently took over in many cities, both major ones such as Porto Velho, Manaus, or Belém, as well as minor cities. Transparent facades with large openings of places like restaurants and bars, heavy traffic, and young adults competing with high power car sound systems are the most common sources of noise pollution and annoyance. Results from noise measurements done at different moments at several "hot spots" in Porto Velho, capital of the state of Rondônia, will be presented. A-weighted equivalent sound pressure levels at all measurement points were found to be in exceedance of 70 dBA during several hours at night, causing a lot of annoyance to the population. Besides the results of measurements some results of the state authorities approach to combine a campaign to raise noise awareness and noise policy enforcement will be presented.

3:20

1pNSb7. Sound power level of speaking people. Marco Caniato (Univ. of Trieste, via valerio 6/a, Trieste 34100, Italy, mcaniato@units.it), Federica Bettarello (Acusticamente - Designing Team, Conegliano, Italy), and Michele Taffarel (M&T Eng., Conegliano, Italy)

In restaurants and cafés many sound sources are present: music, refrigerant, and cooling equipment and people speaking. The smoking prohibition law did move out people creating a lot of aggregation areas outdoor, both in summer and in winter time. As a matter of facts many cafés open on the outer part a stallage in order to provide beverages to outside costumers. In this way, the "people speaking" source became a common problem to deal with and solve. In order to characterize this particular sound source in terms of sound power level of a typical stallage situation full of speaking people, sound pressure power measurements according to ISO 3446 standard were carried out. The results confirm the first investigation achievements provided by Sepulcri *et al*. with a non-direct method.

3:40

1pNSb8. New athletic fields for Saint Joseph's University: A community noise battle. Felicia Doggett (Metropolitan Acoust., LLC, 40 W. Evergreen Ave., Ste. 108, Philadelphia, PA 19118, felicia@metropolitanacoustics.com)

Saint Joseph's University, located in Philadelphia, purchased a vacated private school campus adjacent to theirs. This new campus, which resided in neighboring Lower Merion Township, included 15 acres of open fields that the university desired to turn into NCAA (National Collegiate Athletic Association) baseball, softball, and field hockey fields. Proposed improvements to the fields included permanent bleacher seating, press boxes, dugouts, batting cages, artificial turf playing surfaces, re-grading of the fields, and sound reinforcement systems among other changes. The surrounding community vehemently opposed the proposed changes largely due to the potential noise generated from cheering crowds and sound systems. Through a year and a half of township hearings with testimony provided by a plethora of expert witnesses, Saint Joseph's University finally received approval with various restrictions to build their fields. This presentation explores the research, measurements, and modeling methods undertaken to quantify the acoustical implications on the surrounding community.

4:00

1pNSb9. Noise control for rooftop chiller units: An application in Istanbul. David Meredith (Kinetics Noise Control, 6300 Irelan Place, Dublin, OH 43065, dmeredith@kineticsnoise.com), Hakan Dilmen, Merve Çay (Pro-Plan Proje Mühendislik San. ve Tic. Ltd, İstanbul, Turkey), H. Temel Belek, and Ahmet Arisoy (İstanbul Teknik Üniversitesi Makina Fakültesi, İstanbul, Turkey)

A telecommunication corporation located in Gayrettepe, Istanbul, installed 20 rooftop chillers on their six-story building to air condition their newly introduced client Internet server hall in spring 2010, causing annoyance in the mostly residential neighborhood during the summer months. After a series of night-time acoustic measurements were performed to characterize the noise emitted by the chillers, a three-dimensional noise model of the area was created. Using this model, a ventilated noise control barrier was designed to bring down the contribution of the chiller units to the overall environmental noise level. Following the manufacturing and installation of the barrier, a series of night-time measurements were performed anew, which demonstrated that the application has mitigated the contribution of the noise emitted by the rooftop units to within regulation limits.

Session 1pPAa

Physical Acoustics and Biomedical Acoustics: Acoustics in Microfluidics and for Particle Separation IIb: Bubbles and Drops

Martin Wiklund, Cochair
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Yong-Jae Kim, Cochair
Mech. Eng., Texas A&M Univ., College Station, TX 77843

Contributed Papers

1:00

1pPAa1. Acoustic bubble behavior in a standing wave field. Cyril Desjouy, Pauline Labelle, Bruno Gilles, Jean-Christophe Bera, and Claude Inserra (U1032, LabTAU, Inserm, 151 Cours Albert Thomas, Lyon cedex 03 69424, France, cyril.desjouy@inserm.fr)

This paper focuses on the experimental and numerical studies of acoustic cavitation induced micro bubbles in a standing waveguide filled with water. It is shown that the cylindrical geometry of the system used in this study allows the micro bubbles to self organize into particular patterns. At high pressure amplitudes, the cavitation bubbles tend to aggregate into well known cluster patterns and at relatively low pressure amplitudes, the cavitation micro bubbles aggregate into ring patterns. This study highlights that the shape of these ring patterns is directly related to the Bjerknes force distribution in the resonator. It is also shown both experimentally and numerically that cavitation bubbles may exhibit spiraling behavior around this ring pattern. This spiraling phenomenon is numerically studied and the conditions for which a single cavitation bubble follows an orbital trajectory in the cylindrical waveguide are established, and the influences of the acoustic pressure amplitude and the initial bubble radius are investigated.

1:20

1pPAa2. The roles of acoustic cavitations in the ultrasonic cleansing of fouled micro-membranes. Yuanxiang Yang (School of Civil and Environmental Eng. and DHI-NTU Ctr., Nanyang Technolog. Univ., Nanyang Ave. 50, N1-B1-3a, Maritime Res. Ctr., Singapore 639798, Singapore, yang0250@e.ntu.edu.sg), Qianxi Wang (School of Mathematics, The Univ. of Birmingham, Birmingham, United Kingdom), and Soon Keat Tan (Nanyang Environment and Water Res. Inst. and Maritime Res. Ctr., Nanyang Technolog. Univ., Singapore, Singapore)

This paper concerned the experimental studies on the cleansing mechanism of acoustic cavitation bubbles near the fouled micro-membranes. The existence of the membrane created asymmetry in the flow field, which forced the cavitation bubble to oscillate non-spherically and finally brought forth the jet impact. The oscillations and micro-jets of the cavitation bubbles enhanced the dynamic features of the fluid nearfield and improved the capability of removing fouling. The study on the acoustic multi-bubble system was quite complicated, so first, we focused on the individual bubble dynamics near the membrane. A succession of individual cavitation bubbles were created by Q-switched Nd: YAG laser pulses and observed using a high-speed camera (up to 100 000 frames per second). The jet flow hit against the membrane surface with velocity above 100 m/s, which was strong enough to remove the adherent fouling. We compared the cleansing effects of the cavitation bubbles with different laser energies and stand-off distances from the membrane surface. And then based on the individual bubble dynamics, we can deduce the influence of cavitations in the ultrasonic cleansing of micro-membranes.

1:40

1pPAa3. Waves of acoustically induced transparency in bubbly liquids: Theory and experiment. Nail A. Gumerov (Inst. for Adv. Comput. Studies, Univ. of Maryland, 115 A.V. Williams Bldg., College Park, MD 20742, gumerov@umiacs.umd.edu), Claus-Dieter Ohl (School of Physical and Mathematical Sci., Div. of Phys. and Appl. Phys., Nanyang Technolog. Univ., Singapore, Singapore), Iskander S. Akhatov (Mech. Eng., North Dakota State Univ., Fargo, ND), Sergei P. Sametov, and Maksim V. Khazimullin (Ctr. for Micro- and Nanoscale Dynam. of Dispersed Systems, Bashkir State Univ., Ufa, Russian Federation)

The theory of self-organization of bubbles in acoustic fields predicts formation and propagation of waves of self-induced acoustic transparency. This is a strongly nonlinear effect, which is a result of a two-way coupling of the sound field with the bubble distribution. We are challenging the theory with an experiment. Here, a homogeneous distribution of gas bubbles is first generated and then an ultrasonic field is switched on. The pressure waves are below the cavitation threshold and in a frequency range from 50 kHz to 200 kHz, mostly above the linear resonance frequency of the bubbles. The ultrasound leads to a rapidly propagating bubble wave away from the transducer. The dynamics is observed with a high-speed camera and analyzed. Interestingly, this transparent region is propagating through the bubbly liquid at velocities substantially higher than the bubble rise velocity due to the gravity. A simplified theoretical model of this acoustically induced transparency is developed. Both, analytical and numerical solutions are obtained. A comparison of the experimental data with the model is presented and the underlying physics of the problem is discussed.

Session 1pPAb

Physical Acoustics and Biomedical Acoustics: Acoustics in Microfluidics and for Particle Separation III: Biological Applications

Michel Versluis, Cochair
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Lawrence A. Crum, Cochair
Appl. Phys. Lab., Univ. of Washington, Ctr. for Industrial and Medical Ultrasound, Seattle, WA 98105

Invited Papers

2:00

1pPAb1. Acoustic deformation of cells. Puja Mishra, Martyn Hill, and Peter Glynne-Jones (Faculty of Eng. and the Environment, Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, pgj98r@gmail.com)

The stretching of cells using optical tweezers has been previously demonstrated, enabling the mechanical properties of individual cells to be assessed with potential applications in, for example, identifying cancer cells and parasite infections such as malaria. We demonstrate here a system that uses acoustic radiation forces to compress levitated cells to an extent comparable to that demonstrated with optical tweezers. While the deformation of levitated droplets has been demonstrated in the past, this paper addresses the challenge of producing significant forces on objects that have low acoustic contrast with their host medium and are small compared to the acoustic wavelength. The acoustic deformation can potentially be applied to many (e.g., thousands) of cells simultaneously, opening the way to higher throughput diagnostic devices. In our system, osmotically swollen red blood cells (RBCs) are used to demonstrate the principle as they are particularly compliant due to the absence of a cytoskeleton. A resonance is formed in a square capillary of inner dimension, 100 μm. Excited by a transducer at a half-wave resonance of 7.9 MHz, cells are both levitated and focused laterally into a single line down the center of the capillary prior to the compression forces being applied. We present finite element models of the acoustic deformation, verifying our code against known results for droplet deformation.

2:20

1pPAb2. Application of acoustic radiation pressure to align cells in a commercial flow cytometer. Gregory Kaduchak and Michael D. Ward (Molecular and Cell Essentials, Life Technologies, 29851 Willow Creek Rd., Eugene, OR 97402, greg.kaduchak@lifetech.com)

Forces derived from acoustic radiation pressure can be used to replace or partly replace hydrodynamic forces to align cells and particles in flow cytometry. The ability to focus cells into a tight line without relying on hydrodynamic forces allows many new possibilities for sample delivery. Dilute samples can be processed quickly. Flow velocities can be varied allowing control of particle delivery parameters such as laser interrogation time and volumetric sample input rates. Recently, a commercial flow cytometer that directs particles into the laser interrogation region using acoustic radiation pressure in a 200 □m channel has been developed. In this talk, the application of acoustic radiation pressure in flow cytometry systems from fundamental principles to implementation details will be presented. Data will be shown for both the operational implementation of the acoustic focusing device as well as demonstrating its ability to perform for complex biological assays.

2:40

1pPAb3. On-chip acoustic sample preparation for cell studies and diagnostics. Martin Wiklund, Ida Sadat Iranmanesh, Athanasia E. Christakou, Mathias Ohlin (Appl. Phys., Royal Inst. of Technol., KTH-Albanova, Stockholm 10691, Sweden, martin@biox.kth.se), Aman Russom (Biotechnology, Royal Inst. of Technol., Stockholm, Sweden), and Björn Önfelt (Appl. Phys., Royal Inst. of Technol., Stockholm, Sweden)

We describe a novel platform for acoustic sample preparation in microchannels and microplates. The utilized method is based on generating a multitude of acoustic resonances at a set of different frequencies in microstructures, in order to accurately control the migration and positioning of particles and cells suspended in fluid channels and chambers. The actuation frequencies range from 30 kHz to 7 MHz, which are applied simultaneously and/or in sweeps. We present two devices: A closed microfluidic chip designed for pre-alignment, size-based separation, isolation, up-concentration, lysis of cells, and an open multi-well microplate designed for parallel aggregation and positioning of cells. Both devices in the platform are compatible with high-resolution live-cell microscopy, which is used for fluorescence-based optical characterization. Two bioapplications are demonstrated for each of the devices: The first device is used for size-selective cell isolation and lysis for DNA-based diagnostics, and the second device is used for quantifying the heterogeneity in cytotoxic response of natural killer cells interacting with cancer cells.

3:00

1pPAb4. Numerical modeling for analyzing microfluidic acoustophoretic motion of cells and particles with application to identification of vibro-acoustic properties. Zhongzheng Liu (Mech. Eng., Texas A&M Univ., College Station, TX), Han Wang, Arum Han (Elec. and Comput. Eng., Texas A&M Univ., College Station, TX), and Yong-Joe Kim (Mech. Eng., Texas A&M Univ., 3123 TAMU, College Station, TX 77843, joekim@tamu.edu)

Microfluidic, acoustophoretic cell/particle separation has gained significant interest recently. In order to analyze the motion of cells/particles in the acoustophoretic separation, a one-dimensional (1-D) analytical model in a "static" fluid medium has been widely used, while the effects of acoustic streaming, viscous boundary layers, and 2-D and 3-D geometries are usually not considered. Therefore, it is not sufficient to accurately predict the cell/particle motion. Thus, a numerical modeling procedure for analyzing the acoustophoretic cell/particle motion is presented to include the aforementioned effects. Here, the first-order acoustic pressure and the second-order acoustic streaming velocity are first calculated by using a high-order finite difference method. Then, acoustophoretic force is calculated based on the acoustophoretic force equation proposed by Gorkov and is applied to the Newton's equation of motion to calculate the motion of cells/particles. Through various simulations, the effects of acoustic streaming on the motion of cells/particles are studied. Since the acoustophoretic motion depends on the vibro-acoustic properties (e.g., density, compressibility, and size) of particles/cells, the vibro-acoustic properties can be estimated by optimally fitting the experimental and simulated trajectories. The properties obtained from experimental results with polystyrene beads show good agreement with the data reported in literature.

3:20

1pPAb5. Acoustic radiation force to reposition kidney stones. Michael Bailey, Yak-Nam Wang, Julianna C. Simon, Bryan W. Cunitz (Ctr.Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, bailey@apl.washington.edu), Jonathan D. Harper, Ryan S. Hsi (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Frank Starr, Marla Paun, Barbrina Dunmire (Ctr.Industrial and Medical Ultrasound, Applied Physics Lab, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Seattle, Washington), Lawrence A. Crum (Ctr.Industrial and Medical Ultrasound, Applied Phys. Lab., Univ. of Washington, Seattle, WA), and Mathew D. Sorensen (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA)

Our group has introduced transcutaneous ultrasound to move kidney stones in order to expel small stones or relocate an obstructing stone to a nonobstructing location. Human stones and metalized beads (2–8 mm) were implanted ureteroscopically in kidneys of eight domestic swine. Ultrasonic propulsion was performed using a diagnostic imaging transducer and a Verasonics ultrasound platform. Stone propulsion was visualized using fluoroscopy, ultrasound, and the ureteroscope. Successful stone movement was defined as relocating a stone to the renal pelvis, ureteropelvic junction (UPJ), or proximal ureter. Three blinded experts evaluated for histologic injury in control and treatment arms. All stones were moved. 65% (17/26) of stones/beads were moved the entire distance to the renal pelvis (3), UPJ (2), or ureter (12). Average successful procedure per stone required 14 ± 8 min and 23 ± 16 pushes. Each push averaged 0.9 s in duration. Mean interval between pushes was 41 ± 13 s. No gross or histologic kidney damage was identified in six kidneys from exposure to 20 1-s pushes spaced by 33 s. Ultrasonic propulsion is effective with most stones being relocated to the renal pelvis, UPJ, or ureter. The procedure appears safe without evidence of injury. [Work supported by NIH DK43881, DK092197, and NSBRI through NASA NCC 9-58.]

Contributed Papers

3:40

1pPAb6. Macro-scale acoustophoretic separation of lipid particles from red blood cells. Brian P. Dutra (FloDesign Sonics Inc., 380 Main St., Wilbraham, MA 01095, b.dutra@fdsonics.com), Michael Rust (Biomedical Eng., Western New England Univ., Springfield, MA), Daniel Kennedy (Pharmacology, Western New England Univ., Springfield, MA), Louis Masi (FloDesign Sonics Inc., Wilbraham, MA), and Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA)

Autologous blood salvage is frequently used in cardiac surgery. However, shed mediastinal blood contains lipid particles ranging in size from 10 to 60 μm. Lipid emboli flow subsequently lodge in the brain capillaries resulting in strokes, leading to neurocognitive dysfunction and death. A novel acoustophoretic filtration system has been developed to separate the lipids from the red blood cells (RBCs). The system works at the macro-scale, supporting flow rates in excess of 2 L/h. The system is designed such that the acoustic radiation force is able to overcome the combined effects of fluid drag and buoyancy forces. Both RBCs and lipid particles are therefore trapped in the ultrasonic standing wave. Due to the opposite contrast factors of lipids and RBCs, the two components separate at opposite nodes within the standing wave, with lipids concentrating at pressure anti-nodes and RBCs at pressure nodes. Subsequent gravitational separation is used to separate the lipids and RBCs. Preliminary results were obtained with a suspension of 10x diluted bovine blood mixed with a 0.75% safflower oil emulsion. Measurements indicate a 15 fold increase in hematocrit of the captured RBCs when compared to the original sample solution, and an excellent separation of the oil droplets.

4:00

1pPAb7. Removal of living cells from biosensing surfaces in droplet-based microfluidics using surface acoustic waves. Adrien Bussonnière, Alan Renaudin (Université de Sherbrooke, Pavillon 3IT Parc innovation, 3000 boulevard de l'université, Sherbrooke, QC J1K 0A5, Canada, adrien.bussonniere@usherbrooke.ca), Yannick Miron, Michel Grandbois (Pharmacologie, Université de Sherbrooke, Sherbrooke, QC, Canada), Michaël Baudoin (FILMS, IEMN, Lille, France), and Paul Charette (Université de Sherbrooke, Sherbrooke, QC, Canada)

Removal of living biological cells from surfaces is a critical process for many applications in the area of biosensing and lab-on-a-chip. Trypsin is one of the most effective biochemical tools used to cleave the cells proteins that are responsible for bonding cells to surfaces [K. A. Walsh, Meth. Enzymol. **19**, 41 (1970)]. We propose a method using Rayleigh-type (20 MHz) surface acoustic wave (SAW)-based mixing [Renaudin *et al*., Lab Chip **10**, 111 (2010)] as an accelerator for trypsin-mediated removal of living cells from surfaces. In the experiments, a 10 μL droplet of Hank's Balanced Salt Solution (HBSS)-trypsin is placed on a piezoelectric substrate covered with human embryonic kidney cells (HEK293). Using phase contrast microscopy, cells removal time for different acoustic power levels and trypsin concentrations is measured. Results from validation experiments show that a minimum of 180 s is necessary to completely release surface-bonded cells covered by the 10 μL droplet without the use of SAW (negative control). By using microstreaming flow in the droplets generated by the SAW, cells are released from the surface in less than 8 s. This work will contribute to improved lab-on-a-chip devices based on living cell biosensing.

4:20

1pPAb8. Acoustophoretic force-based compressibility measurement of cancer cells having different metastatic potential. Han Wang (Dept. of Elec. and Comput. Eng., Texas A&M Univ., College Station, TX), Zhongzheng Liu (Dept. of Mech. Eng., Texas A&M Univ., College Station, TX), Dong Moon Shin, Georgia Chen (Dept. of Hematology and Medical Oncology, Emory Univ. School of Medicine, Atlanta, Texas), Younghak Cho (Dept. of Mech. System Design Eng., Seoul National Univ. of Sci. and Technol., Seoul, South Korea), Yong-Joe Kim (Dept. of Mech. Eng., Texas A&M Univ., College Station, TX), and Arum Han (Dept. of Elec. and Comput. Eng., Texas A&M Univ., 309C WERC, TAMU 3128, College Station, TX 77843-3128, arum.han@ece.tamu.edu)

Mechanical properties of cells such as compressibility are regarded to be different as cancer cells progress into metastatic state. Traditional methods for measuring mechanical properties of single cells such as AFM and micropipette aspiration require labor-intensive procedures and can cause damage to cells due to direct contact, thus unsuitable for high-throughput measurement. Acoustophoretic force exerted on particles under acoustic-standing-waves depends on the particle and medium's vibro-acoustic properties. Thus, cells with different mechanical properties show different mobility under acoustic resonant field, which can be analyzed to decipher the mechanical properties of cells. Here we present a high-throughput, single-cell-resolution, cell compressibility measurement approach based on acoustic-standing-wave-induced force, and the finding that head and neck cancer cells having different metastatic capacities show noticeable differences in compressibility. The acoustophoresis chip has a straight flow channel with a piezoelectric transducer attached at the bottom. Trajectories of moving cells in the channel under acoustic standing wave excitation in the absence of flow are recorded. By using a microfluidic acoustophoretic model, the simulated trajectories of cells are calculated. The mechanical properties of cells are estimated by fitting the experimental and simulated trajectories thereby. Cells with highest metastatic capacity showed highest compressibility, consistent with previously reported clinical observations.

4:40

1pPAb9. Acoustic separation of sub-micron particles in gases. Etienne Robert, Ramin Imani Jajarmi (Mechanics, Kungliga Tekniska Högskolan (KTH), Osquars Backe 18, Stockholm 100 44, Sweden, etienne@mech.kth.se), Jan B. Pettersson (Chemistry and Molecular Biology, Göteborg Univ., Göteborg, Sweden), and Klas Engvall (Chemical Technol., Kungliga Tekniska Högskolan (KTH), Stockholm, Sweden)

In several areas of science and technology, there is a strong need for concentrating, separating, and sorting small particles suspended in gaseous flows. Acoustic fields can be used to accomplish this task, an approach extensively used in liquid phase microfluidics that has great potential for aerosol treatment. This paper presents an experimental investigation of acoustophoresis for very small particles in gases, with sizes ranging from tens to hundreds of nanometers. The phenomenon is studied in a rectangular channel with variable height in which a standing acoustic field is created by a broadband electrostatic transducer operated in the 50–100 kHz range. The flow can either be seeded with particles with a known size distribution or ambient laboratory air can simply be circulated in the channel. Downstream of the separation channel, the flow is separated into enriched and depleted streams with adjustable slits for analysis. The particle number density and size distribution is measured with a scanning mobility particle sizer (SMPS) as a function of position in the standing wave pattern. From these measurement, the separation efficiency is determined as a function of the particle size, excitation frequency, bulk flow velocity, and number of nodes in the channel. Further analysis yields an estimation of the force acting on the particles, which for very small particles yields novel information on the magnitude of acoustophoretic forces in the transition and molecular flow regimes.

MONDAY AFTERNOON, 3 JUNE 2013 — 514ABC, 1:00 P.M. TO 5:00 P.M.

Session 1pPPa

Psychological and Physiological Acoustics: Binaural Hearing and Binaural Techniques I

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

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

Invited Papers

1:00

1pPPa1. Applications of models of binaural hearing. Jens P. Blauert (Inst. of Commun. Acoust., Ruhr-Univ. Bochum, Bochum D-44780, Germany, jens.blauert@rub.de) and Jonas Braasch (Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Models of binaural hearing have been proposed for more than 60 decades, but it is only recently that they are ready for technological application. Recognizing this situation, 15 laboratories in Europe and in the United States have founded a research group (AABBA) with the aim of setting in place "Aural Assessment by Means of Binaural Algorithm." Now, after its first 4-years term of activity, the group has published their recent results in a chapter book (Springer)—including a software package for building customized binaural models. In this talk, the group's most relevant results will be described. The results concern the following areas of model application, among others: binaural analysis of aural scenes, binaural de-reverberation, binaural quality assessment of audio channels, loudspeakers and performance spaces, binaural perceptual coding, binaural processing in hearing aids and cochlea implants, binaural systems in robots, binaural/tactile human–machine interfaces, speech-intelligibility prediction in rooms, and/or multi-speaker scenarios. Current members of AABBA are research labs at Universities/Techn. Universities in Helsinki, Boston, Cardiff, Oldenburg, Lyon, Troy, NY (Rensselaer), Bochum, Berlin, Copenhagen, Dresden, Eindhoven, Munich, Paris (Pierre et Marie Curie), Toulouse, and at the Austrian Academy's Acoustics Research Inst. in Vienna.

1:20

1pPPa2. An improved method for head-related transfer function interpolation and grid matching. Aussal Matthieu (DIGITAL MEDIA SOLUTIONS, 45, grande allée du 12 février 1934, Noisiel 77186, France, matthieu.aussal@dms-cinema.com), Alouges François (CMAP, ECOLE POLYTECHNIQUE, Palaiseau, France), and Katz F. Brian (LIMSI-CNRS, Orsay, France)

Today, there exists a growing number of HRTF datasets available with each set often proposing a unique variation on the spatial discretization measurement grid. These differing grids, typically determined by the mechanical system employed, result in datasets, which are not directly comparable or exploitable. To alleviate the limitation of incompatible grids and assist in the adaptation of measurements performed on one grid to another, facilitating the inter-exchange of HRTF sets, a fixed radius HRTF interpolation method is proposed. The approach is based on decomposition of the sound field using spherical harmonics, allowing for a global spatial recomposition. The frequency domain HRTF is separated into its complex components and interpolations are performed independently before reconstitution. Spherical harmonic truncation order is chosen to provide a system which is roughly square, improving matrix inversion with Tikhonov regularization. A high spatial density HRTF was used as a test case for evaluating the interpolation method. A series of measures are employed to quantitatively compare the quality of the interpolation as compared to traditional interpolation methods in both the time and frequency domains.

1:40

1pPPa3. Statistical analysis of head related transfer function data. Yuancheng Luo, Dmitry N. Zotkin (Comput. Sci. and UMIACS, Univ. of Maryland, College Park, MD), and Ramani Duraiswami (VisiSonics Corp., A.V. WIlliams Bldg., #115, College Park, MD 20742, ramani@umiacs.umd.edu)

The head related transfer function (HRTF) is a function that characterizes the response of a given individual to sound from a particular location in an egocentric coordinate system. The range dependence is often neglected, and the HRTF is approximated as a function of frequency and direction, $H(\theta,\varphi,\omega)$. The HRTF displays considerable inter-personal variability, and a major open problem is the development of a generative model for the HRTF from anthropometry. Further, the sampling used in measuring HRTF data varies widely from database to database, and moreover often there are no measurements for elevations below the subject. This raises associated questions of optimal sampling, interpolation, hole-filling, and others. In this work, we model the HRTF via a non-parametric, data-driven, Gaussian Process Regression model. We develop efficient regression techniques to perform inference using this model on measured HRTF data. We then suggest methods for HRTF interpolation, HRTF extrapolation, feature extraction, sampling, and personalization. The methods are tested on the CIPIC database and results presented. [Partial NSF support is gratefully acknowledged.]

2:00

1pPPa4. Toward optimal functional representation of head-related transfer functions in the horizontal plane. Wen Zhang, Thushara D. Abhayapala, Rodney A. Kennedy (Res. School of Eng., Australian National Univ., Canberra, ACT 0200, Australia, wen.zhang@anu.edu.au), and Mengqiu Zhang (School of Elec. Eng., KTH Royal Inst. of Technologyity, Stockholm, Sweden)

Head-related transfer function (HRTF) individualization using principle component analysis (PCA) modeling rely on the empirical data to reduce HRTF dimensionality for an optimal representation and to achieve HRTF personalization by tuning the model weights with the subject anthropometric parameters. However, for these representations, the basis is discrete and data dependent, which can limit its usefulness in universal HRTF representation. This paper studies the optimal functional representation of magnitude HRTF of 45 subjects for sound sources in the horizontal plane. We first use circular harmonics to extract the subject-independent HRTF angular dependence. The remaining spectral components of 45 subjects are then modeled by PCA and two standard functions, i.e., Fourier series and Fourier Bessel series. The metric to evaluate the model efficiency is the expansion weights cumulative variance. We identify that individual magnitude HRTFs over 20 kHz range could be modeled adequately well by a linear combination of only 19 Fourier series; this is a near optimal representation in comparison with the statistical PCA model. Further analysis of the model weights with subjective anthropometric measurements will provide a promising method for HRTF individualization, especially considering the nature of data independent continuous basis functions employed in the proposed functional representation.

2:20

1pPPa5. A three dimensional children head database for acoustical research and development. Stine Harder, Rasmus R. Paulsen (Informatics and Mathematical Modelling, Tech. Univ. of Denmark, Richard Petersens Plads., Bldg. 321, Office 215, Kgs. Lyngby 2800, Denmark, rrp@imm.dtu.dk), Martin Larsen (Oticon A/S, Smørum, Denmark), and Søren Laugesen (Oticon Res. Ctr. Eriksholm, Snekkersten, Denmark)

Most computational–acoustic work within spatial hearing relies on head-related transfer functions from databases of measurements taken on adult humans or dummy heads. We aim to provide a set of 3D digital heads including children, from which head-related transfer functions can be computed instead of measured. However, current volumetric scanning techniques do not have sufficient resolution for accurately scanning the external ear, and computed tomography also involves radiation. In this paper, we propose a framework for scanning, stitching, and meshing complete human heads. The process starts by acquisition of multiple 3D surface scans of the same subject using a high-resolution photogrammetric scanner. Second, the scans are semi-automatically aligned and noise and incoherence is removed. This is followed by an iterative process where a volumetric implicit representation of the head is optimized. The process consists of a regularized surface-reconstruction step followed by an alignment step. Finally, a surface representation of the entire head is extracted using a triangulation of the zero-level iso-surface of the implicit volume. The process has been used to reconstruct the heads of children aged 10 months to 9 years. The data and the associated reconstruction algorithms will be made publicly available for use in acoustical research and development.

2:40

1pPPa6. Consistency among the head-related transfer functions from different measurements. Xiao-li Zhong and Bo-sun Xie (Acoustic Lab., Phys. Dept., School of Sci., South China Univ. of Technol., Bldg. No.18, Wu Shan Rd. No.381, Guangzhou, Guangdong 510641, China, xlzhong@scut.edu.cn)

Empirical measurement is a common approach to obtaining head-related transfer functions (HRTFs). Due to differences in experimental conditions and possible errors, some deviations exist among the data from different measurements even for the same subject. This work aims to evaluate deviations of HRTFs from different measurements. Five sets of KEMAR HRTFs from three groups including MIT-media Lab, CIPIC Interface Lab, and our lab are used. A free-field equalization is applied to the original data so as to eliminate the influence caused by the difference in the response of electro-acoustic measuring chain. The deviations among HRTF magnitudes for different measurements are specified by spectral distortion. Results indicate that the magnitude deviation increases with increasing frequency and reaches more than 10 dB above the frequency of 6 kHz. Salient deviations often occur in contralateral source directions or in ipsilateral source directions near the frequency of pinna-related notch. Nevertheless, most deviations can be effectively reduced after an auditory filter smoothing. This work suggests that the inconsistency in measured HRTF data and its impact on auditory perception should be taken into account when comparing and standardizing HRTFs from different measurements. [Work supported by National Nature Science Fund of China Grant No. 11004064]

3:00–3:20 Break

3:20

1pPPa7. Conventional and spatial principal component analysis on near-field head-related transfer functions. Bo-sun Xie and Cheng-yun Zhang (Acoust. Lab., Phys. Dept., School of Sci., South China Univ. of Technol., Bldg. No.18, Wu Shan Rd. No. 381, Guangzhou 510641, China, phbsxie@scut.edu.cn)

Head-related transfer functions (HRTFs) vary with both frequency and source position. The near-field HRTFs with source distance less than 1.0 m are particularly complicated due to their distance dependence. Principal component analysis (PCA), which is conventionally carried out in the frequency or time domain, has been widely applied to reduce the dimensionality of far-field HRTF data with source distance greater than 1.0 m. The present work first extends the conventional PCA, and then proposes a spatial PCA method in the spatial domain rather than in the frequency or time domain to reduce the dimensionality of near-field HRTF data. An illustrative case indicates that near-field HRTF magnitudes at 9 distances with 493 directions for each distance can be approximately represented by the weighted sum of 15 spectral shape basis functions using the conventional PCA or the weighted sum of 15 spatial basis functions using the spatial PCA. Both representations account for more than 98% energy variation of the original data, and reduce the dimensionality of the original data to about a quarter. The proposed method is also applicable to the head-related impulse responses in the time domain. Furthermore, the spatial PCA scheme is potentially applicable to simplify near-field HRTF measurement.

3:40

1pPPa8. Spatially continuous model of the broadband time-of-arrival in the head-related transfer functions. Piotr Majdak and Harald Ziegelwanger (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14, Wien 1040, Austria, piotr@majdak.com)

Head-related transfer functions (HRTFs) describe the filtering of the incoming sound by the human anatomy. They contain the so-called broadband time-of-arrivals (TOAs), which interaural differences yield the well-known interaural time differences used to estimate the lateral position of sound sources by the human auditory system. The TOAs are essential for a time-synchronous binaural rendering of multiple virtual sound sources or for interpolation of the timing information in an existing HRTF set. Estimation of the TOA is usually done separately for each spatial direction and is thus prone to errors and directional outliers. A method for a robust estimation of spatially continuous TOA function from a set of listener-specific HRTFs is presented. The method relies on a geometric model of the HRTF-measurement setup represented by parameters like head position, radius, and ear position. The model parameters were fit to HRTFs of a sphere numerically calculated under various conditions, and to measured HRTFs of 160 listeners. The resulting model parameters and TOA functions corresponded well with the measurement geometry and manually derived TOAs, respectively. The model parameters were further compared to those resulting from a simplified model which assumes the listener being in the center of the HRTF-measurement setup, demonstrating the impact of the usually neglected aspect of listener position on the HRTF timing quality.

4:00

1pPPa9. Calculation of listener-specific head-related transfer functions: Effect of mesh quality. Harald Ziegelwanger, Piotr Majdak (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14, Vienna, Vienna A-1040, Austria, harald.ziegelwanger@oeaw.ac.at), and Andreas Reichinger (Zentrum für Virtual Reality und Visualisierung Forschungs-Gmbh, Wien, Austria)

The geometry of head and ears defines the listener-specific directional filtering of the incoming sound. The filtering is represented by the head-related transfer functions (HRTFs), which provide spectral features relevant for the localization of sound-sources. HRTFs can be acoustically measured or numerically calculated based on a geometric representation of the listener. While the acoustically measured HRTFs usually provide localization performance similar to that obtained in free-field listening, the performance obtained with numerically simulated HRTFs, however, heavily depends on the quality of the geometric and acoustic model of the listener used for the simulation. In this study, we show how to calculate listener-specific HRTFs with spectral features similar to that from acoustically measured HRTFs for the entire audible frequency range. We review the boundary-element method coupled with the fast-multipole method and we present details on the prerequisites like the geometry-capture technique, acoustical parameters, and the numerical algorithms. Further, the effect of the mesh quality on the HRTFs was investigated by systematically varying the average edge length from 1 to 5 mm. The HRTF amplitude spectra were analyzed and evaluated by visual comparison and in a localization model. The optimal average edge length for a fast calculation of HRTFs yielding potentially good localization performance is discussed.

4:20

1pPPa10. Quantitative assessment of spatial sound distortion by the semi-ideal recording point of a hear-through device. Pablo Hoffmann, Flemming Christensen, and Dorte Hammershøi (Acoustics, Aalborg Univ., Fredrik Bajers Vej 7B5, Aalborg 9220, Denmark, pfh@es.aau.dk)

A hear-through device combines a microphone and earphone in an earpiece so that when worn, one per ear, it can work as an acoustically transparent system allowing for simultaneous individual binaural recording and playback of the real sound field at the ears. Recognizing the blocked entrance to the ear canal as the ideal recording point—i.e., all directional properties of the incident sound field are recorded without distortion—it is critical for such device to be sufficiently small so that it can be completely inserted into the ear canal. This is not always feasible and the device may stretch out from the ideal position and thus distort the captured spatial information. Here we present measurements that quantify by how much the directional properties of the sound field are distorted by semi-ideal hear-through prototypes built by mounting miniature microphones on the outer part of selected commercial insert earphones. This includes an analysis of the magnitude by which spatial information is distorted and the extent to which these distortions are direction dependent. Potential strategies for compensating these distortions are also considered.

4:40

1pPPa11. Measuring pressure and particle velocity along the human ear canal. Marko Hiipakka and Ville Pulkki (Aalto Univ., Otakaari 5 A, Espoo 02150, Finland, Marko.Hiipakka@aalto.fi)

A non-invasive method of measuring or estimating accurately the head-related transfer functions (HRTFs) and headphone transfer functions (HpTFs), i.e., the pressure at the eardrum rather than at the blocked ear canal entrance is called for. In this work, a miniature-sized acoustic pressure–velocity sensor is used to measure both pressure and velocity along the ear canals of human test subjects. The measurements are used to study the applicability of a recently proposed method of estimating the pressure at the eardrum from pressure–velocity measurements made at the ear canal entrance. The measurement results are compared to results from computational modeling with human ear canal parameters. In addition, the effect of the PU-sensor itself on the pressure at the eardrum is studied. It is shown that the estimation method is reliable and accurate for most human subjects. The diameter and the shape of the ear canal affect the results in such a way that the best results are obtained with wide and straight ear canals whereas less accurate results are obtained with narrow and curved ear canals. It is concluded that the estimation method facilitates the obtaining of individual HRTFs and HpTFs at the eardrum using non-invasive measurements.

MONDAY AFTERNOON, 3 JUNE 2013 | 516, 1:00 P.M. TO 5:00 P.M.

Session 1pPPb

Psychological and Physiological Acoustics: Psychoacoustics and Perception (Poster Session)

Elin Roverud, Chair
Purdue Univ., West Lafayette, IN 47906

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.

1pPPb1. Effects of compression on the use of onset time differences to detect one tone in the presence of another. Sara M. Madsen and Brian Moore (Psychology, Cambridge Univ., Downing St., Desborough, Cambridge CB2 3EB, United Kingdom, smkm2@cam.ac.uk)

It is easier to hear one of two notionally "simultaneous" tone complexes if the onset of the masker complex is delayed relative to that of the signal. However, the ability to use onset asynchrony as a cue may be reduced when using amplitude compression, due to distortion of the onset of sounds (overshoot effects). We assessed how fast- and slow-acting five-channel compression affects the ability to use onset asynchrony to detect one (signal) complex tone when another (masking) complex tone is played almost simultaneously. A 2:1 compression ratio was used with normal-hearing subjects and individual compression ratios and gains recommended by the CAM2 hearing aid fitting method were used for hearing-impaired subjects. For the normal-hearing subjects, performance improved with increasing onset asynchrony in all conditions. The improvement was greatest with fast compression and least with no compression. Preliminary results for the hearing-impaired subjects indicate smaller but similar effects of onset asynchrony and a greater benefit of compression. The benefit of compression probably occurs because compression increases the level of the part of the signal that occurs before the masker relative to the masker. [Work supported by Starkey (U.S.A.) and the MRC (UK).]

1pPPb2. Study on effects of presence of cue-tone on psychophysical tuning curves. Shunsuke Kidani, Ryota Miyauchi, and Masashi Unoki (School of Information Sci., JAIST, 1-1, Asahidai, Nomi, Ishikawa 923-1292, Japan, kidani@jaist.ac.jp)

Our previous study indicated that tunings of the auditory filter were sharpened by the presence of a cue-tone [Kidani *et al*. (2012), ISH2012]. It is unclear, however, whether the variation of the auditory filter due to the cue-tone is caused by excitation or suppression, because tip of the filter is normalized at 0 dB. Psychophysical tuning curves (PTCs) can show that the detection threshold is decreased at the probe frequency or increased around the probe by the presence of cue-tone, indicating excitation and suppression respectively. PTCs, because, are measured as masked threshold of probe by narrow-band noise. This study aims to consider the effect of the presence of cue-tone by measuring of PTCs. In present study, PTCs were measured in

simultaneous masking in the absence and presence of cue-tone for four probe frequencies. The probes were presented at 10 dB above each absolute threshold. The frequency and level of the cue-tone were same as the probe. The result revealed that filter-Q, regarded as the sharpness of tuning, was increased by the presence of cue-tone when the probe frequencies were 1.0 and 2.0 kHz, while the filter-Q was not changed when the probe frequencies were 0.5 and 4.0 kHz.

1pPPb3. Optimizing the simultaneous estimation of frequency selectivity and compression using notched-noise maskers with asymmetric levels. Tomofumi Fukawatase, Toshio Irino, Ryuichi Nisimura, Hideki Kawahara (Faculty of Systems Eng., Wakayama Univ., 930 Sakaedani, Wakayama 640-8510, Japan, tomofuka0522@gmail.com), and Roy D. Patterson (Dept. of Physiol., Development and Neurosci., Cambridge Univ., Cambridge, United Kingdom)

It is important for the development of hearing aids and other audio devices to make accurate estimates of the frequency selectivity and compression of the auditory filter. Previously, we reported a technique for estimating the compression of the auditory filter that combined data from a simultaneous notched-noise experiment and a temporal masking curve (TMC) experiment. Unfortunately, the TMC data derived for individual listeners in forward masking is not stable; the cue to the presence of the signal is not entirely clear in forward masking. In this paper, we report attempts to make the traditional simultaneous notched-noise technique more sensitive to the effects of cochlear compression by varying the relative levels of the noise bands. Asymmetric-level maskers (ALMs) make it possible to estimate the filter shape and compression of the auditory filter simultaneously and reliably; the slope of the input–output function is substantially lower than with symmetric-level maskers. We also describe a procedure for incorporating a sensitivity analysis into the filter-fitting process to determine the minimum number of notched-noise conditions required to produce reliable estimates of selectivity and compression, in hopes of being able to employ the technique with hearing impaired listeners.

1pPPb4. Reliability of procedures used for scaling loudness. Walt Jesteadt and Suyash N. Joshi (Psychoacoust. Lab., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, Walt.Jesteadt@boystown.org)

In this study, 16 normally hearing listeners judged the loudness of 1000-Hz sinusoids using magnitude estimation (ME), magnitude production (MP), and categorical loudness scaling (CLS). Listeners in each of four groups completed the loudness scaling tasks in a different sequence on the first visit (ME, MP, CLS; MP, ME, CLS; CLS, ME, MP; CLS, MP, ME), and the order was reversed on the second visit. This design made it possible to compare the reliability of estimates of the slope of the loudness function across procedures in the same listeners. The ME data were well fitted by an inflected exponential (INEX) function, but a modified power law was used to obtain slope estimates for both ME and MP. ME and CLS were more reliable than MP. CLS results were consistent across groups, but ME and MP results differed across groups in a way that suggested influence of experience with CLS. Although CLS results were the most reproducible, they do not provide direct information about the slope of the loudness function because the numbers assigned to CLS categories are arbitrary. This problem can be corrected by using data from the other procedures to assign numbers that are proportional to loudness. [Work supported by NIH.]

1pPPb5. Sequential dependencies in magnitude scaling of loudness. Suyash N. Joshi and Walt Jesteadt (Psychoacoust. Lab., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, Suyash.Joshi@boystown.org)

Ten normally hearing listeners used a programmable sone-potentiometer knob to adjust the level of a 1000-Hz sinusoid to match the loudness of numbers presented to them in a magnitude production task. Three different power-law exponents (0.15, 0.30, and 0.60) and a log-law with equal steps in dB were used to program the sone-potentiometer. The knob settings systematically influenced the form of the loudness function. Time series analysis was used to assess the sequential dependencies in the data, which increased with increasing exponent and were greatest for the log-law. It would be possible, therefore, to choose knob properties that minimized these dependencies. When the sequential dependencies were removed from the data, the slope of the loudness functions did not change, but the variability decreased. Sequential dependencies were only present when the level of the tone on the previous trial was higher than on the current trial. According to the attention band hypothesis [Green and Luce, Perception Psychophys., 1974] these dependencies arise from a process similar to selective attention, but observations of rapid adaptation of neurons in the inferior colliculus based on stimulus level statistics [Dean *et al*., Nature Neurosci. (2005)] would also account for the data. [Work supported by NIH.]

1pPPb6. Effect of musical training on static and dynamic measures of spectral-pattern discrimination. Stanley Sheft (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 S. Paulina St., 1012 AAC, Chicago, IL 60612, ssheft@gmail.com), Kirsten Smayda (Univ. of Texas at Austin, Austin, TX), Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL), W. Todd Maddox, Bharath Chandrasekaran, (Univ. of Texas at Austin, Austin, TX)

Both behavioral and physiological studies have demonstrated enhanced processing of speech in challenging listening environments attributable to musical training. The relationship, however, of this benefit to auditory abilities as assessed by psychoacoustic measures remains unclear. Using tasks previously shown to relate to speech-in-noise perception, the present study evaluated discrimination ability for static and dynamic spectral patterns by 49 listeners grouped as either musicians or nonmusicians. The two static conditions measured the ability to detect a change in the phase of a logarithmic sinusoidal spectral ripple of wideband noise with ripple densities of 1.5 and 3.0 cycles per octave chosen to emphasize either timbre or pitch distinctions, respectively. The dynamic conditions assessed temporal-pattern discrimination of 1-kHz pure tones frequency modulated by different 5-Hz lowpass noise samples with thresholds estimated in terms of either stimulus duration or signal-to-noise ratio. Musicians performed significantly better than nonmusicians on all four tasks. Discriminant analysis showed that group membership was correctly predicted for 84% of the listeners with the structure coefficient of each measure greater than 0.46. Results suggest that enhanced processing of static and dynamic spectral patterns defined by low-rate modulation may contribute to the relationship between musical training and speech-in-noise perception. [Work supported by NIH.]

1pPPb7. Spectral uncertainty and similarity effects in informational masking identically related to the Simpson-Fitter metric of target-masker separation. An-Chieh Chang, Jacob Stamas (Commun. Sci. and Disord., Dept. Commun. Sci. and Disord., Univ. of Wisconsin, Madison, WI 53705, achang5@wisc.edu), Inseok Heo (Elec. and Comput. Eng., Univ. of Wisconsin - Madison, Madison, WI), Lynn Gilbertson, and Robert Lutfi (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, Madison, WI)

Evidence is provided suggesting a primary dependence of informational masking (IM) on the stochastic separation of target and masker given by Simpson-Fitter's da [Lutfi *et al*. J. Acoust. Soc. Am. **132**, EL109-113 (2012)]. The stimuli were synthesized impact sounds of plates played in sequence as masker-target-masker triads. Their spectra varied independently and at random on each presentation as would correspond to changes in plate size. In the 2IFC procedure the listener's task was to choose the larger-sized target. The effect of spectral uncertainty regarding the masker was examined by measuring d' performance for different values of the variance in masker size. The effect of spectral similarity of target and masker was examined by measuring performance for different values of the mean difference between target and masker size. The functions relating d' to da in both cases were identical and of similar slope across listeners. Identical functions were also obtained, though with shallower slopes, when listeners judged the target hit with greater impact force. The results are considered in terms of their implications for the development of a model of IM that emphasizes the statistical properties of signals over loosely defined concepts of target-masker similarity and masker uncertainty.

1pPPb8. Spatial uncertainty and proximity effects in informational masking identically related to the Simpson–Fitter metric of target-masker separation. Jacob Stamas (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, Univ. of Wisconsin, Madison, WI 53705, jstamas@wisc.edu), Inseok Heo (Elec. and Comput. Eng., Univ. of Wisconsin - Madison, Madison, WI), Lynn Gilbertson, An-Chieh Chang, and Robert Lutfi (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, Madison, WI)

Further evidence is provided suggesting a primary dependence of informational masking (IM) on the stochastic separation of target and masker given by Simpson-Fitter's da [Lutfi *et al*. J. Acoust. Soc. Am. **132**, EL109-113 (2012)]. The stimuli were brief bursts of Gaussian noise or words played

in sequence as masker-target-masker triads. The apparent position of bursts (words), from left to right, was varied independently and at random on each presentation using KEMAR HTRFs. In the 2IFC procedure, the listener's task was to choose the target positioned further to the right. The effect on performance of spatial uncertainty regarding the masker was examined by manipulating the position variance of the masker. The effect on performance of spatial proximity of target to masker was examined by manipulating the position mean difference between target and masker. In both cases, the data were well described by a single linear function relating d' performance to da; intercepts differed across listeners, but slopes were similar. Comparable results presented at this meeting for the effects of spectral uncertainty and similarity of target and masker suggest that the statistical properties of signals may be a more significant determinant of IM than their specific acoustic properties.

1pPPb9. Toward a model of informational masking: The Simpson–Fitter metric of target-masker separation. Lynn Gilbertson, An-Chieh Chang, Jacob Stamas (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, Madison, WI), Inseok Heo (Elec. and Comput. Eng., Univ. of Wisconsin - Madison, Madison, WI), and Robert Lutfi (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, 1410 E. Skyline Dr., Madison, WI 53705, ralutfi@wisc.edu)

Informational masking (IM) is the term used to describe masking that appears to have its origin at some central level of the auditory nervous system beyond the cochlea. Supporting a central origin are the two major factors associated with IM: trial-by-trial uncertainty regarding the masker and perceived similarity of target and masker. Here preliminary evidence is provided suggesting these factors exert their influence through a single critical determinant of IM, the stochastic separation of target and masker given by Simpson–Fitter's da [Lutfi *et al*., J. Acoust. Soc. Am. **132**, EL109-113 (2012)]. Target and maskers were alternating sequences of tones or words with frequencies, F0s for words, selected at random on each presentation. The listener's task was to discriminate a frequency-difference in the target tones or identify the target words. Performance in both tasks was found to be constant across conditions in which the mean difference (similarity), variance (uncertainty), or covariance (similarity) of target and masker frequencies were selected to yield the same value of da. The results are discussed in terms of their implications for the development of a model of IM that emphasizes the statistical properties of signals over loosely defined concepts of masker uncertainty and target-masker similarity.

1pPPb10. Extending Schroeder-phase masking: Influence of direction and shape of masker instantaneous frequency. Evelyn M. Hoglund, Yonghee Oh, Joseph F. Hribar, Kelsi J. Wittum, Megan L. Strang, and Lawrence L. Feth (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd, Columbus, OH 43204, hoglund.1@osu.edu)

Schroeder (1970) devised an algorithm to produce low peak factor signals. Schroeder signals with equal amplitude spectra, but reversed phase spectra, reveal large differences in masker effectiveness for listeners with normal hearing [Smith *et al*. (1986)]. Results reported here extend previous work to include detection of multiple bursts of the same frequency, and multiple bursts that increase or decrease in frequency. Signal frequencies were selected to correspond with harmonics in the maskers. Results indicate that changing the frequency of the signal amplifies the difference between the Schroeder-phase maskers, but the direction of the change does not. When a frequency modulated tone is substituted for the Schroeder maskers, masked threshold depends on the shape of the instantaneous frequency (IF) function, as well as the direction of change. For linear FM (with IF similar to the Schroeder maskers), masked thresholds are comparable to the Schroeder-phase maskers. However, for logarithmic FM, IF changes at a constant ERB/s and directional differences are much smaller. A modified channel model [Oh (2012)] shows substantial differences in the basilar membrane response to these different maskers. [Research supported by a grant from the Office of Naval Research #N000140911017.]

1pPPb11. Multicomponent signal detection: Tones in noise and amplitude modulation detection. Eric R. Thompson (Ball Aerosp. & Technol. Corp., 2610 7th St., Bldg. 441, Wright Patterson, OH 45433, eric.thompson.ctr@wpafb.af.mil), Brian D. Simpson, and Nandini Iyer (Air Force Res. Lab., Wright Patterson, OH)

In order to predict the detectability of broadband acoustic signals, a model must include a means of integrating information across frequencies. There have been several previous studies measuring the detectability of multicomponent signals, but it is still not clear what the best model is when signal components are not equally detectable. Some researchers have proposed that thresholds are driven by the most detectable component (max-d' model), while others have found that the best model for their data is a statistical summation model, where component sensitivities are combined using a Pythagorean sum. In the present study, detection thresholds were collected in broadband noise for single tones at three frequencies and three signal-to-noise ratios (SNRs) and for all combinations of SNRs for the three tones presented together. Also, amplitude modulation (AM) detection thresholds were measured for a 16-Hz AM signal imposed on 300-Hz-wide noise bands centered at three frequencies at three modulation depths for each band individually, for combinations of two bands and for all three bands presented together. While both models (max-d' and Pythagorean sum) can predict the general trend of the multicomponent data from the single component data, neither model fits the data very well.

1pPPb12. Investigating the effects of intensity on the bandwidth of peripheral filtering in an amplitude-modulation notch detection task. Matthew L. Richardson, Allison I. Shim, and Bruce G. Berg (Cognit. Sci., UC Irvine, 159 St. Vincent, Irvine, CA 92618, mlrichar@uci.edu)

The effect of intensity on the effective bandwidth of auditory temporal processing is investigated. Thresholds for detecting sinusoidal amplitude-modulation of a 200-Hz wide band of noise centered at 1000 Hz are measured in the presence of a notched noise masker. The masker consists of two, 200-Hz wide, unmodulated bands of noise placed at frequencies above and below the modulated band. Thresholds for a modulation rate of 10 Hz are estimated for different notch bandwidths ranging from 100 Hz to 2740 Hz. The use of a slow modulation frequency aims to avoid possible central limitations of temporal processing at higher modulation frequencies. Intensity is varied across two conditions, with all three bands of noise presented at either 40 dB SPL or 85 dB SPL. Threshold functions for the two intensity levels are essentially identical. The maximum notch width at which an effect of the masker is observed is approximately 500 Hz. The results are consistent with a hypothesis that the filtering characteristics of temporal processing (e.g., envelope model) and spectral processing (e.g., power spectrum model) are different.

1pPPb13. Thresholds of tone pitch contour discrimination for English listeners. Rachael C. Gilbert (Linguistics, The Univ. of Texas at Austin, 4812 Ave. H, Apt B, Austin, TX 78751, rachaelgilbert@gmail.com) and Chang Liu (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

This study aims to provide psychophysical data on English language listeners' ability to discriminate tone pitch contours. Just noticeable differences (JND) of F0 contour changes were measured in six native listeners of American English. Previous work in our lab found English listener thresholds for offsets of falling tones to be significantly lower than those for onsets of rising tones. To what extent this difference is due to the position of the F0 shift versus the direction of the F0 contour is unclear. In this study, we control for four experimental factors: stimulus type (speech, nonspeech), position of F0 shift (onset, offset), direction of shift (upward, downward), and F0 contour direction (falling, rising). Preliminary results reveal that English listeners had significantly lower psychophysical thresholds for F0 shifts at the offset than at the onset. No significant difference was found for F0 shift direction, F0 contour direction, or stimulus type. The current data suggest that the F0 shift position was the primary determinant in our previous study and replicate other findings showing that English listeners perceive tones on a psychophysical base. Future work will examine these results in relation to those of native tone language listeners.

1pPPb14. An automated procedure for detecting human frequency-following responses to voice pitch. Fuh-Cherng Jeng and Jiong Hu (Commun. Sci. and Disord., Ohio Univ., 1 Ohio University Dr., Grover Ctr. W224, Athens, OH 45701, jeng@ohio.edu)

The frequency-following response (FFR) to voice pitch has been widely examined in research laboratories and has demonstrated its potential to be transformed into a useful tool for patients with hearing, speech, and language disorders in the clinic. During the past decade, many aspects of the FFR have been reported. The presence of such a response, however, still

relies on subjective interpretation of the observer. Aside from a recent study reporting two algorithms for detecting such a response, there has been limited number of studies reporting the development of an automated procedure for FFR. The purpose of this study is (1) to develop an automated procedure that utilizes the statistical properties of the temporal and spectral energy distributions in the recorded waveforms and (2) to explore the effectiveness, accuracy, and efficiency of the automated procedure and compare them with those obtained from conventional algorithms and human judgments.

1pPPb15. Infants' ability to perceive the pitch of unresolved harmonics. Bonnie K. Lau and Lynne A. Werner (Univ. of Washington, 523 Broadway East Unit 217, Seattle, WA 98102, bonniekwlau@gmail.com)

An important phenomenon for models of pitch perception is that adult listeners' can extract pitch from complexes containing only unresolved harmonics. Although 3-month-olds discriminate resolved harmonics on the basis of missing fundamental (MF) pitch, their ability to discriminate unresolved harmonics is unknown. This study investigated the ability of adults, 7- and 3-month-olds, to perceive the pitch of unresolved harmonics using an observer-based method. Stimuli were MF complexes that were bandpass filtered with a -12 dB/octave slope, combined in random phase, and presented at 70 dB SPL for 650 ms with a 50 ms rise/fall and with a pink noise to mask distortion products. The experiment consisted of two conditions: (1) "low" unresolved harmonics between 2500 and 4500 Hz based on MFs of 160 Hz (H17–H26) and 200 Hz (H13–H22) and (2) "high" unresolved harmonics between 4000 and 6000 Hz based on MFs of 190 Hz (H22–H31) and 200 Hz (H20–H29). To demonstrate MF pitch discrimination, participants were required to ignore spectral changes in complexes with the same fundamental and to respond only when the fundamental changed. Interestingly, variable performance in the "high" condition was observed with adult participants. However, nearly all infants tested categorized complexes by MF pitch in both conditions, suggesting discrimination of unresolved harmonics at 3 months.

1pPPb16. Mistuning detection in a complex of unresolved harmonics: Effects of age. Sara K. Mamo (Div. of Speech and Hearing Sci., Univ. of North Carolina - Chapel Hill, 075 MacNider Hall, CB #7070, Chapel Hill, NC 27599-7070, smamo@med.unc.edu) and John H. Grose (Dept. of Otolaryngol.-Head/Neck Surgery, Univ.of North Carolina - Chapel Hill, Chapel Hill, NC)

Older adults experience speech perception difficulties that are not explained by audiometric thresholds. One hypothesis is that temporal processing deficits contribute to these speech-in-noise difficulties. To test this, threshold for mistuning was measured for a component within a complex of unresolved harmonics—a cue that likely depends on sensitivity to envelope perturbations. The complex comprised harmonics 12–16 of a 100-Hz or 200-Hz fundamental, and the duration was either 170-, 340-, or 680-ms. Presentation level was 70 dB SPL. The starting phases of all components were randomized on each presentation. The 3AFC procedure adaptively varied the mistuning of harmonic 14 to obtain the frequency-shift threshold. Younger and older listeners with audiometrically normal hearing participated. The expectation was that older adults would require greater mistuning to detect changes in envelope periodicity and that this would be more evident at shorter durations. Preliminary results support an overall effect of age as well as effects of fundamental frequency and duration. Results will be considered in the context of parallel speech-evoked ABR studies being undertaken in this population that point to age-related deficits in envelope processing, possibly driven by poor encoding of unresolved harmonics. [Work supported by NIDCD 1-F32-DC012217-01A1 & 5-R01-DC001507.]

1pPPb17. Retention of gap length in normal-hearing listeners. Meghan M. Smith, Dennis Ries, and Audra Woods (Commun. Sci. and Disord., Grover Ctr. W241, 1 Ohio Univ., Athens, OH 45701, ms802311@ohio.edu)

Listeners' ability to retain information about gap length within a noise burst was studied. JND for gap length between the target and comparison stimuli was obtained using the single-interval adjustment-matrix procedure for retention intervals that were silent, included four noise bursts, or included four noise bursts with discrete gaps. JNDs for retention intervals containing either type of noise burst did not differ significantly from those obtained for the silent retention interval. This result differs from that found for retention of pitch and loudness. This might occur as more cortical resources are used for retention of auditory temporal information.

1pPPb18. The influence of feature detection on working memory in complex auditory fields. AnneMarie Chiodi, Aurora Weaver, and Dennis Ries (Commun. Sci. and Disord., Grover Ctr. W241, 1 Ohio Univ., Athens, OH 45701, ac175509@ohio.edu)

Research reveals that feature detectors may enhance listener performance in auditory discrimination and detection tasks involving frequency modulation (FM) [Cusack and Carlyon, J. Exp. Psychol. **29**, 713–725 (2003)]. The influence of these detectors on retention of auditory information is unknown. This study investigated the impact of FM on listener performance in auditory, delayed-comparison tasks for conditions that differed in the number of background stimuli within two perceptual windows separated by various retention intervals. The background stimuli within both windows were either all modulated sinusoids or pure-tones for a given trial. An additional stimulus was presented in each window that could differ in its modulation state (FM or unmodulated sinusoid) across the two windows. The temporal placement and frequencies of all stimuli within the first window were assigned randomly for a given trial and the second window followed these parameters. Listeners were to determine whether the two windows were the same or different. Preliminary results show that same-different judgments of target modulation state was easier in a field of four unmodulated sinusoids than vice versa. This result occurred regardless of retention interval length. The further influence of field complexity, retention interval length, and working memory span will be discussed.

1pPPb19. Boundary effects on the illusory continuity of and interrupted glide through a notched noise. Valter Ciocca (School of Audiol. and Speech Sci., UBC, 2177 Wesbrook Mall, Vancouver, BC V6T 1Z3, Canada, vciocca@audiospeech.ubc.ca) and Nicholas Haywood (MRC Inst. of Hearing Res., Nottingham, United Kingdom)

This study investigated the illusory continuity of an interrupted frequency glide through a notched-noise burst. A 2I-2AFC procedure was used to measure detection of the (target) portion of the frequency glide that overlapped in time with the noise. The portions of the glide preceding and following the noise (flankers) could be present or absent. The center frequency of the notch coincided with either the frequency end-point of the flanker that preceded the noise, or the onset frequency of the flanker that followed the noise. A control condition with a wide-band noise burst (absent notch) was also included. Performance was poorest in the absent notch condition and was significantly poorer with present than with absent flankers. This suggests that listeners perceptually restored the missing target when flankers were present. Performance was also less accurate (indicating stronger illusory continuity) when the notch was centered on the end-point of the flanker that preceded the noise. These results suggest that the masking of the onset of the flanker following the noise provides a stronger cue to the perception of continuity than the masking of the offset of the flanker that precedes the noise.

1pPPb20. Sensory consonance of two simultaneous sine-tones. Reinhart Frosch (ETH and PSI (retired), Sommerhaldenstrasse 5B, Brugg 5200, Switzerland, reinifrosch@bluewin.ch)

In Chapter 4 of my book "Musical Consonance and Cochlear Mechanics" (vdf, Zurich, 2012), four psychoacoustic experiments on the sensory consonance of two simultaneous sine-tones are described. In each of those experiments, the deeper-tone frequency f_d was kept fixed, at $f_d = 132$, 264, 528, or 1056 Hz. Each experiment was done twice, at sound-pressure levels of 50 and 70 dB (SPL). The resulting consonance curves (sensory consonance versus higher-tone frequency f_h) exhibit consonance minima at beat-rates $b_{m.d.} = f_h - f_d$ (where "m.d." = "most dissonant") ranging from 13 Hz (at $f_d = 132$ Hz) to 39 Hz (at $f_d = 1056$ Hz). In Section 15.1 of the above-mentioned book, these most dissonant beat rates are shown to agree well with the following empirical law: $b_{m.d.} = (1.07s^{-0.5}) * \mathrm{sqrt}(f_{avg})$, where $f_{avg} = f_d + b_{m.d.}/2$. The present study was prompted by the comments of a reader; the just described empirical law is unsatisfactory because in the underlying experiments the deeper-tone frequency f_d [rather than the average frequency $(f_d + f_h)/2$] was kept constant. It was found that the data agree equally well with the following modified empirical law: $b_{m.d.} = (1.09s^{-0.5}) * sqrt(f_d)$. This modification does not affect the validity of the complex-tone consonance theories described in Chapters 15 and 16 of the mentioned book.

1pPPb21. Dependency of tonality perception on frequency, bandwidth, and duration. Armin Taghipour, Bernd Edler, Masoumeh Amirpour, and Jürgen Herre (Int. Audio Lab. Erlangen, Am Wolfsmantel 33, Erlangen 91058, Germany, armin.taghipour@audiolabs-erlangen.de)

Psychoacoustic studies show that a narrowband noise masker exhibits a stronger simultaneous masking effect than a tonal masker with the same signal power placed at the noise center frequency. Consequently, perceptual audio codecs commonly incorporate some sort of tonality estimation as part of their perceptual model. However, common tonality estimation techniques do not necessarily reflect the perception of tonality by human listeners. As long as the tone and narrowband noise signals are long enough, they are easily distinguishable for normal hearing listeners. However, if the stimulus duration decreases, both signal types approach the shape of impulses, and therefore, at some point become audibly identical. Consequently, at a given frequency and noise bandwidth, there is a duration threshold below which the signals cannot be distinguished. A series of so-called "2-AFC 3-step up-down" psychoacoustic tests are designed and carried out to investigate the frequency and bandwidth dependency of these duration thresholds. The test results, collected from 32 listeners, are statistically evaluated and confirm a decreasing threshold for increasing center frequency and bandwidth. These results can be used to improve psychoacoustic models for audio codecs by using tonality estimators with frequency and bandwidth adapted temporal resolution.

1pPPb22. Reflection orders and auditory distance. Catarina Mendonça (Dept. of Signal Process. and Acoust., Aatlo Univ., Otakaari 5, Espoo FI-02150, Finland, mendonca.catarina@gmail.com), João Lamas (Centro Algoritmi, Univ. of Minho, Guimaraes, Portugal), Tom Barker (Dept. of Signal Process., Tampere Univ. of Technol., Tampere, Finland), Guilherme Campos, Paulo Dias (Departamento de Electrónica, Telecomunicações e Informática, Univ. of Aveiro, Aveiro, Portugal), Ville Pulkki (Dept. of Signal Process. and Acoust., Aatlo Univ., Espoo, Finland), C. Silva, and Jorge Santos (Centro Algoritmi, Univ. of Minho, Guimarães, Portugal)

The perception of sound distance has been sparsely studied so far. It is assumed to depend on familiar loudness, reverberation, sound spectrum, and parallax, but most of these factors have never been carefully addressed. Reverberation has been mostly analyzed in terms of ratio between direct and indirect sound, and total duration. Here we were interested in assessing the impact of each reflection order on distance localization. We compared sound source discrimination at an intermediate and at a distant location with direct sound only, one, two, three, and four reflection orders in a 2AFC task. At the intermediate distances, normalized psychophysical curves reveal no differentiation between direct sound and up to three reflection orders, but sounds with four reflection orders have significantly lower thresholds. For the distant sources, sounds with four reflection orders yielded the best discrimination slopes, but there was also a clear benefit for sounds with three reflection orders. We discuss the results in terms of direct-to-reflected ratio, reflection directionality, and spectral information.

1pPPb23. Ventriloquism effect and aftereffect in the distance dimension. L'uboš Hládek, Christophe C. Le Dantec, Norbert Kopčo (Inst. of Comput. Sci., P. J. Safarik Univ., Jesenná 5, Košice 04001, Slovakia, lubos.hladek@student.upjs.sk), and Aaron R. Seitz (Dept. of Psychol., Univ. of California, Riverside, CA)

When an auditory target is presented simultaneously with a spatially displaced visual target, the perceived auditory target location shifts toward the visual target. This effect, known as the *ventriloquism effect* or *visual capture*, has been extensively studied in the horizontal dimension, but not in distance. Here, we measured distance localization performance in a reverberant room. Stimuli were either audio-visual (AV) 300-ms broadband noise bursts presented synchronously with spatially congruent or incongruent visual stimuli/LEDs, or auditory-only (A-only) noise bursts. One of eight speakers (distance 70 cm to 203 cm directly ahead of the listener) presented a stimulus on each trial. During adaptation runs, the AV stimuli were presented with the V-component closer or further by 30% than the A-component (displacement direction fixed within session). The ventriloquism effect was observed for both V-closer and V-further AV stimuli, with slightly stronger shifts induced by the V-closer stimuli. Ventriloquism aftereffect, assessed by presenting A-only trials interleaved with the adaptation-AV trials, was also observed, but was weaker than the ventriloquism effect. The results suggest that visual targets do capture auditory targets in the distance dimension, but visual modulation might be asymmetrical with respect to distance. [Work supported by EU FP7-247543, VEGA-1/0492/12, NSF (BCS-1057625).]

1pPPb24. An assessment of virtual auditory distance judgments among blind and sighted listeners. Andrew J. Kolarik (Dept. of Psychol., Cambridge Univ., Downing St., Cambridge, CB2 3EB, United Kingdom, ak771@cam.ac.uk), Silvia Cirstea, Shahina Pardhan (Vision and Eye Res. Unit, Anglia Ruskin Univ., Cambridge, United Kingdom), and Brian Moore (Dept. of Psychol., Cambridge Univ., Cambridge, United Kingdom)

Auditory distance perception is a crucial component of blind listeners' spatial awareness. Many studies have reported supra-normal spatial auditory abilities among blind individuals, such as enhanced azimuthal localization [Voss *et al*. (2004)] and distance discrimination [Kolarik *et al*. (in press)]. However, it is not known whether blind listeners are better able to use acoustic information to enhance judgments of distance to single sound sources, or whether lack of visual spatial cues prevents calibration of auditory distance information, leading to worse performance than for sighted listeners. Blind and sighted listeners were presented with single, stationary virtual sound sources between 1.22 and 13.79 m away in a virtual anechoic environment simulated using an image-source model. Stimuli were spoken sentences. Sighted listeners systematically underestimated distance to remote virtual sources, while blind listeners overestimated the distance to nearby virtual sources and underestimated it for remote virtual sources. The findings suggest that blind listeners are less accurate at judging absolute distance, and experience a compression of the auditory world, relative to sighted listeners. The results support a perceptual deficiency hypothesis for absolute distance judgments, suggesting that compensatory processes for audition do not develop among blind listeners when estimating the distance to single, stationary sound sources.

Session 1pSA

Structural Acoustics and Vibration: Measurement and Modeling of Structures with Attached Noise Control Materials II

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Invited Papers

1:00

1pSA1. Visco-thermal dissipations in heterogeneous porous media. Fabien Chevillotte, Luc Jaouen, and François-Xavier Bécot (Matelys, 1 rue Baumer, Vaulx-En-Velin 69120, France, fabien.chevillotte@matelys.com)

Semi-phenomenological models have been widely used since the 1990's for modeling visco-thermal dissipations of acoustical energy through porous media. These dissipations are taken into account by two complex frequency-dependent functions (the dynamic density ρ_{eq} (ω) and the dynamic bulk modulus K_{eq} (ω)), which are analytically derived from macroscopic parameters. Other models were derived for modeling perforated plates [J. Sound Vib. **303** (2007)], double porosity media [J. Acoust. Soc. Am. **114**(1) (2003)] or, more recently, porous composites made of porous inclusions in a substrate porous media [Acta Acoust. **96** (2010)]. So far, this latter model is not able to consider the shape of the host and the client media. This model can neither be extended to limiting cases of perforated plate model nor double porosity model. Based on a modified equivalent fluid model, this work proposes a unified model which accounts, analytically, for the shape of the inclusions, might they be porous or not. This model enables to describe the acoustic behavior of any kind of composite media from perforated plates to arbitrarily shaped porous composites including configurations of porous inclusions in solid matrix or double porosity media. In addition, possible pressure interactions between the substrate material and the inclusions are accounted for.

1:20

1pSA2. Prediction of acoustic properties of parallel assemblies by means of transfer matrix method. Kévin Verdière, Raymond Panneton, Saïd Elkoun (GAUS, Université de Sherbrooke, 2500 bd de l'université, Sherbrooke, QC J1K2R1, Canada, kevin.verdiere@usherbrooke.ca), Thomas Dupont, and Philippe Leclaire (DRIVE, ISAT, Université de Bourgogne, Nevers, France)

The Transfer Matrix Method (TMM) is used conventionally to predict the acoustic properties of laterally infinite homogeneous layers assembled in series to form a multilayer. In this work, a parallel assembly process of transfer matrices is used to model heterogeneous materials such as patchworks, acoustic mosaics, or a collection of acoustic elements in parallel. In this method, it is assumed that each parallel element can be modeled by a 2x2 transfer matrix, and no diffusion exists between elements. The method is validated by comparison with finite element (FE) simulations and acoustical tube measurements on different configurations at normal and oblique incidence. Then, an overview of the possibilities, such as the combination of series and parallel matrices, the sound absorption coefficient, and the transmission loss of a parallel array of resonators or three-dimensional geometries is presented and discussed.

1:40

1pSA3. Investigations on the sensitivity of the relationships between sound absorption characteristics and microstructure related parameters for polyurethane foams. Morvan Ouisse (Appl. Mech., FEMTO-ST Univ. of Franche-Comté, 24 rue de l'épitaphe, Besançon 25000, France, morvan.ouisse@univ-fcomte.fr), Olivier Doutres, Noureddine Atalla (GAUS, Dept. of Mech. Eng., Université de Sherbrooke (Qc), Sherbrooke, QC, Canada), and Mohamed Ichchou (LTDS, Ecole Centrale de Lyon, Ecully, France)

Straightforward semi-phenomenological models have been developed for highly porous polyurethane foams to predict the macroscopic non-acoustic parameters involved in the classical Johnson-Champoux-Allard model (i.e., porosity, airflow resistivity…) from microstructure properties (i.e., strut length, strut thickness, and reticulation rate). These microstructure properties are measured using sophisticated optical methods (i.e., optical microscope, SEM) and a large variability can be observed due to great complexity of the 3D microstructure; variability also depends on the precision of the measurement device. This work investigates how the variability associated with the model inputs affects the model outputs (i.e., non-acoustic parameters, surface impedance, and sound absorption coefficient). The sensitivity analysis is based on the Fourier Amplitude Sensitivity Test (FAST). It helps quantify the correlation between the input parameters and identify the parameters contributing the most to output variability, thus requiring precise measurement. This study illustrates the preponderant impact of the reticulation rate (i.e., open pore content) on acoustic performances and guides the user on the required optical measurement device.

2:00

1pSA4. Multi-scale acoustics of partially open cell poroelastic foams. Minh Tan Hoang (Faurecia Interior System, Marne-la-Vallée, France), Guy Bonnet, and Camille Perrot (Laboratoire Modélisation et Simulation Multi Echelle, MSME UMR 8208 CNRS, Université Paris-Est, 5, Boulevard Descartes, Bâtiment Lavoisier, Bureau D13, Champs-sur-Marne 77454, France, camille.perrot@univ-paris-est.fr)

The present paper reports on the modeling of linear elastic properties of acoustically insulating foams with unit cells containing solid films or membranes at the junction between interconnected pores from a numerical homogenization technique. It combines fluid-flow induced microstructure identification with simulations of the effective Young's modulus and Poisson ratio from a mixture of routinely available laboratory measurements (porosity, permeability, cell size) and finite element calculations when the boundary conditions of the periodic unit cell take particular symmetric forms. This combination results in microstructural determination of the macroscopic coefficients entering into the Biot-Allard theory of wave propagation and dissipation through porous media. Precise control over pore morphology and mechanical properties of the base material renders this multi-scale approach particularly suitable for various advanced applications.

2:20

1pSA5. Modeling of the acoustic absorption of bi-modal polylactide foams. Shahrzad Ghaffari Mosanenzadeh (Mech. and Industrial Eng., Univ. of Toronto, No. 5, King's College Rd., Toronto, ON M5S 3G, Canada, shahrzad.ghaffari@yahoo.com), Olivier Doutres (Groupe d'Acoustique de Vibrations, Univ. of Sherbrooke, Sherbrooke, QC, Canada), Hani E. Naguib, Chul B. Park (Mech. and Industrial Eng., Univ. of Toronto, Toronto, ON, Canada), and Noureddine Atalla (Groupe d'Acoustique de Vibrations, Univ. of Sherbrooke, Sherbrooke, QC, Canada)

In this study, highly porous bi-modal structures were designed and fabricated from polylactide (PLA) as the main structure and utilizing polyethylene glycol (PEG) to form micro pores by compression molding combined with particulate leaching technique. The pore size of the foam structure was controlled by salt particulates and higher interconnectivity was achieved by the co-continuous blending morphology of PLA matrix with water-soluble PEG. This fabrication method makes it possible to control pore geometry and interconnectivity closely and therefore is an ideal approach to study the relation between microstructure and acoustic properties of the foams. PLA is a bio-based thermoplastic polymer derived from renewable resources. Therefore, the resulting acoustic foams are benign and environmentally friendly. Fabricated foams were characterized based on cellular, acoustic, and mechanical properties. The acoustic performance of the foams was studied by measuring the normal incident absorption coefficient in accordance with the ASTM E1050 standard. An analytical model based on Johnson–Champoux–Allard model was used to numerically simulate the acoustic performance of foams under study. Numerical results predict the absorption behavior of PLA foams with high accuracy. Through this research, open porosities close to 90% were achieved, and the effect of water soluble polymer on cellular properties, acoustic and mechanical performance of polylactide foams was studied.

2:40

1pSA6. Improving the sound absorbing efficiency of closed-cell foams using shock waves. Olivier Doutres, Noureddine Atalla (GAUS, Université de Sherbrooke, 2500 Boul. de l'Université, Sherbrooke, QC J1K 2R1, Canada, olivier.doutres@usherbrooke.ca), Martin Brouillette, Christian Hébert (Dept. of Mech. Eng., Shock wave Lab., Sherbrooke, QC, Canada), and David Begg (Woodbridge Foam Corp., Woodbridge, ON, Canada)

Producing closed-cell foams is generally cheaper and simpler than open-cell foams. However, the acoustic efficiency of closed-cell foam materials is poor because it is very difficult for the acoustic waves to penetrate the material. A method to remove the membranes closing the cell pores (known as reticulations) and thus to improve the acoustic behavior of closed-cell foam material is presented. The method is based on the propagation of shock waves inside the foam aggregate where both the shock wave generator and the foam are in air at room conditions. Various shock treatments have been carried out on a polyurethane foams, and the following conclusions were drawn: (1) the reticulation rate increases and thus the airflow resistivity decreases while increasing the amplitude of the shock treatment; (2) the softness of the foam increases; (3) the process is reliable and repeatable; (4) obtained acoustic performance is comparable to classical thermal reticulation; and (5) the process can be used to control the reticulation rate along the thickness.

3:00

1pSA7. Full-band exact homogenization of one-dimensional elastic metamaterials. Min Yang, Zhiyu Yang, and Ping Sheng (Physics, Hong Kong Univ. of Sci. and Technol., Dept. of Phys., HKUST, Clear Water Bay, Kowloon, Hong Kong 852, Hong Kong, erwinstu@ust.hk)

Metamaterials extend the realm of materials' properties by carefully designed structural inclusions. By targeting the extraction of effective properties from composite materials, homogenization theory plays an important role for metamaterials in their design and characterization. However, conventional homogenization methods are limited to the long wavelength limit. Here, we introduce an exact homogenization scheme valid for one-dimensional metamaterials over the full frequency band. In this scheme, with the aid of eigenstates' characterization, a set of explicit formulas for effective mass density and effective elastic modulus are obtained by matching the surface responses properties of a metamaterial's single structural unit with a piece of effectively homogenized material. In the frequency regimes beyond the conventional homogenization theory, new features, such as the imaginary parts of the effective parameters, have been found. Applying this scheme on a layered structure, the predicted transport properties and displacement fields from the effective parameters show excellent agreement with numerical simulations.

3:20

1pSA8. Noise control using lightweight acoustic metamaterials. Christina J. Naify (National Res. Council, 4555 Overlook Ave. SW, Washington, DC 20375, christina.naify.ctr@nrl.navy.mil)

Metamaterials have emerged as promising solutions for manipulation of sound waves in a variety of applications. Negative dynamic mass has been explored in metamaterial applications to improve sound insulation in both three-dimensional (ball-in-rubber), and two-dimensional (membrane-type) approaches. Noise control utilizing locally resonant acoustic materials (LRAM) resulted in improved

sound insulation by 500% over acoustic mass law predictions at peak transmission loss (TL) frequencies. The LRAM contribute minimal added mass, making them appealing for weight-critical applications such as aerospace structures. In this study, an overview of LRAM for noise control applications will be presented, including potential issues associated with scale-up of the structure. TL of single-celled and multi-celled LRAM was measured using an impedance tube setup with systematic variation in geometric parameters to understand the effects of each parameter on acoustic response. Finite element analysis (FEA) was also performed to predict TL as a function of frequency for structures with varying complexity, including stacked structures and multi-celled arrays. [Work supported by the Office of Naval Research.]

3:40

1pSA9. Omnidirectional acoustic absorber with a porous core—Theory and measurements. Olga Umnova, Andy Elliott, and Rodolfo Venegas (Univ. of Salford, The Crescent, Salford m5 4wt, United Kingdom, o.umnova@salford.ac.uk)

An omni-directional acoustic absorber consisting of a porous core and the impedance matching metamaterial layer has been designed and tested in the laboratory. Semi-analytical and numerical models have been developed and validated. The numerical model takes into account the viscous losses in the matching layer. A 1.5 m demonstrator has been built and tested under acoustic and weak shock excitation. Testing with acoustic excitation showed good agreement between measurement and model, with near perfect absorption between 400 and 1000 Hz. Testing against weak single-pulse shock in an anechoic chamber also confirmed a significant reduction in peak pressure levels when compared to a conventional porous absorber without matching layer. The findings suggest that structure is equally effective when wrapped around an object like a column, pipeline, or the underside of a vehicle, as it would be when entirely filled with an absorbing porous material.

4:00

1pSA10. Reflexion of flexural waves at the end of a tapered beam of quadratic profile covered with a thin viscoelastic layer. Vivien Denis, Julien Poittevin, Adrien Pelat, Benjamin Elie, and Francois Gautier (Laboratoire d'Acoustique de l'Université du Maine, Université du Maine, Avenue O. Messiaen, Le Mans 72000, France, francois.gautier@univ-lemans.fr)

Flexural waves propagating in a beam can be efficiently absorbed if one extremity is tapered with a power law profile and covered by a very thin viscoelastic layer [Krylov, JSV **274**, 605–619 (2004)]. Such a terminaison induces an effect known as "the acoustic black hole effect" (ABH), which is resulting from properties of propagation of flexural wave in beams having non homogeneous thicknesses: if the thickness decreases locally, flexural waves slow down and the amplitude of the displacement field increases, leading to efficient energy dissipation if an absorbing layer is placed where the thickness is minimum [Georgiev *et al*., JSV 330, 2497–2508 (2011)]. Absorption of the ABH terminaison is estimated, thanks to the direct measurement of the reflexion coefficient, using a wave decomposition technique. Experimental modal analysis of a ABH beam can be performed using a "high resolution" technique, which permits to estimate the modal density. Analysis of these experimental results is performed, thanks to a model based on the finite difference method. It is shown that local transverse modes are playing an important role in the absorption properties of ABH.

4:20

1pSA11. Importance of container geometry of an elastomer particulate damper called Enidamp^{TM} into the damping of vibrations. Marcelo Bustamante, Samir N. Gerges, Júlio A. Cordioli, Ovdalio P. Martin (Vib. and Acoust. Lab., UFSC, João de Deus Machado 224, Ap. 3 - Trindade, Florianópolis, SC 88040-900, Brazil, marcelo.bustamante@lva.ufsc.br), Jeffrey N. Weisbeck, and Mark Ott (Engineering, ITT/Enidine Inc., Orchard Park, NY)

Research testing has been conducted to development of an innovative damping treatment, called Enidamp^{TM}. This treatment can add considerable damping to a structure by leading a vibration through a rigid connection, to a set of elastomer particles, which behaves as a damper. It is of interest to study the parameters involved in this mechanism of energy dissipation and to achieve optimal performance of the Enidamp^{TM} system. Particularly, this paper experimentally analyzes the importance of the container geometry which houses the elastomer particles. For this purpose, the fluidization point at which the elastomeric particles become optimally excited to maximize damping is found for different depths and widths of the Enidamp^{TM} container keeping the volume constant. Important conclusions from this experiment guide future studies for prototype improvements.

4:40

1pSA12. Structural-borne sound mitigation in small wind turbines using constrained viscoelastic layer. Baruch Pletner (Intelligent Dynam. Canada, LTD, 11 Acadia St., Dartmouth, NS B2Y 2N1, Canada, baruch.pletner@iptrade.com), Nic Strum, David Sampson, and Ali Kheirabadi (Seaforth Energy, Inc., Halifax, NS, Canada)

As the growing acceptance of small wind turbines operating in suburban and rural communities coincides with increasingly stringent regulations on the sound emitted by these turbines, the need for sound mitigation solutions becomes urgent. Small turbines need to be affordable for small business use, and thus, proposed solutions must be cost-effective and low maintenance. Easy retrofit to existing turbines is also desirable. Wind turbines generate sound via two main mechanisms: structural borne sound generated by the gearbox and generator and transmitted through the nacelle structure and aeroacoustic sound generated by the interaction of the airstream with the rotating blades and other turbine components. Current study focused on the mitigation of structural-borne sound in a 50 kW wind turbine using a constrained viscoelastic layer. The viscoelastic layer comprised of multiple tiles with normal force to the nacelle structure provided by ratcheting bands. Optimal value for the normal force was empirically determined, and the resulting reductions in generated sound were documented both in the laboratory and on a working turbine under a number of operating conditions. The result is a cost-effective solution with zero cost of ownership and easy installation on a wide range of small to medium-size wind turbines.

Session 1pSCa

Speech Communication: Mixed Effects Modeling: Applications and Practice in Speech Research

Christian DiCanio, Chair
Haskins Lab., 300 George St., Ste. 900, New Haven, CT 06511

Chair's Introduction—12:55

Invited Papers

1:00

1pSCa1. Modeling multi-level factors using linear mixed effects. Cynthia G. Clopper (Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, clopper.1@osu.edu)

Linear mixed-effects models of 2x2 designs are readily interpretable using treatment contrast coding, although their interpretation is not directly comparable to the interpretation of more traditional ANOVAs. In particular, the interpretation of the "main effect" term for one factor holds only for the baseline level of the other factor. Interpreting models of designs involving factors with more than two levels and/or interactions involving more than two factors is more complex and even less comparable to familiar interpretations of ANOVAs. Alternative methods for analyzing these more complex designs include using different contrast coding (e.g., sum, Helmert, or custom), selecting specific baseline levels for the factors, and running multiple models of the same data set with different baseline levels of comparison. These methods may return quite different results, however, such that a significant factor with treatment contrast coding may not be significant with sum contrast coding or a significant interaction term in one model may not be significant when the baseline levels of the relevant factors are changed. Thus, although linear mixed effects provide an opportunity to model complex designs with multiple sources of variability, this modeling requires careful consideration of model parameters to achieve the most appropriate interpretation of the data.

1:20

1pSCa2. Multilevel models, covariates, and controlled factors in experimental speech research: Unified analyses of highly structured data. Noah H. Silbert, Jared A. Linck (Ctr. for Adv. Study of Lang., Univ. of Maryland, 7005 52nd Ave., College Park, MD 20742, nsilbert@umd.edu), and Mark VanDam (Speech & Hearing Sci., Washington State Univ., Spokane, WA)

Experimental speech research often makes use of complex experimental designs, but even when multiple experimental factors are manipulated, measured outcomes may be influenced by non-controlled and incompletely controlled factors. Multilevel models (of which mixed-effect models are a special case) enable unified analysis of the relationships between, on the one hand, trial-level data and, on the other, experimental factors and potentially important non-controlled variables. Fitted multilevel models allow us to draw inferences simultaneously about group-level experimental effects and covariates (the typical focus of experimental work) as well as individual subject and item properties (both of which can be important in applied research). The utility of multilevel models will be illustrated with analyses of data from a number of studies. We present models of phonological structure, gender differences, and within-gender subject variability in the acoustics of spoken English consonants; simultaneous modeling of experimental factors, subject and item variability, and second language proficiency in bilingual lexical processing; and modeling of the effects of age, hearing loss, phonological/lexical properties, subject and item variability, and multiple vocabulary-related covariates in early language development.

1:40

1pSCa3. Experimentally elicited productions: Differences and similarities between mixed effects and ANOVA analyses. Matthew Goldrick (Linguistics, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, matt-goldrick@northwestern.edu)

Currently, many experimental studies of speech production use fully counterbalanced designs to examine variation in categorical (e.g., correct/incorrect) or relatively continuous measures (e.g., reaction times, voice onset times). These data present several challenges to ANOVA analyses. Some of these issues are well known to be the speech community; for example, the non-normality of dependent variables such as proportion correct. Others have been less extensively addressed; for example, many speech studies account for participant- but not item-specific contributions to variance. I'll discuss the opportunities and challenges in using linear mixed effects models to address these issues. I'll review some of the common issues that arise in using such models and discuss how to interpret and report their results.

2:00

1pSCa4. The use of mixed effects models in quantifying the dynamics of speech. Khalil Iskarous (Linguistics, Univ.of Southern California, 301P Grace Ford Salvatory, USC, Los Angeles, CA 90089-1693, kiskarou@usc.edu)

Mixed-effects models have been used often in quantifying the variability in data from experiments in speech and language. In most of these experiments, the dependent variable is measured at some landmark of a kinematic or decision process. However mixed-effects models are increasingly being used to quantify the variability of dependent variables that vary in time, such as articulator movements, formant transitions, and eye tracking data. This presentation will first provide a tutorial introduction to the use of the mixed-effects model, especially the growth-curve variant, for quantifying variability where time is an essential independent variable. It will then be argued that the model coefficients can be interpreted as dynamic coefficients of differential equations that describe the dynamics of the underlying processes.

2:20–2:40 Panel Discussion

Session 1pSCb

Speech Communication: Autocorrelation-Based Features for Speech Perception

Yoichi Ando, Cochair
Ando Yoichi, Kobe Univ., 1-4-132-105 Hiyodoridai, 1-4-132-105 Hiyodoridai, Kobe Kita 657-1123, Japan

Peter Cariani, Cochair
Biomedical Eng., Boston Univ., 629 Watertown St., Newton, MA 02460

Chair's Introduction—12:55

Invited Papers

1:00

1pSCb1. Autocorrelation function mechanism for pitch salience and cross-correlation function mechanism for sound localization revealed by magnetoencephalography. Yoshiharu Soeta (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol. (AIST), 1-8-31 Midorigaoka, Ikeda, Hyogo 563-8577, Japan, y.soeta@aist.go.jp), Ryota Shimokura (Dept. of Otorhinolaryngol. - Head and Neck Surgery, Nara Medical Univ., Kashihara, Japan), and Seiji Nakagawa (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol. (AIST), Ikeda, Osaka, Japan)

A model of primary sensations and spatial sensations is proposed by Ando (2001). The model of the auditory-brain system includes the autocorrelation function (ACF) and the interaural cross-correlation function (IACF) mechanisms. At present, environmental noises are evaluated by sound level such as equivalent continuous A-weighted sound pressure level (LAeq). However, we sometimes feel annoyed with sound with low sound level because of the quality. Sound quality can be characterized by factors obtained from ACF and IACF of sound. For example, pitch and pitch strength can be characterized by delay time and amplitude of the maximum peak of the ACF. Directional sensation can be characterized by delay time and amplitude of the maximum peak of the IACF. To verify the model, we investigated how ACF and IACF factors are coded in our human brain. The results indicated that delay time and amplitude of the maximum peak of the ACF and IACF are coded by the latency and strength of brain activity. In addition, we applied the model to analyze a Buddhist sutra chanted in temples. The results indicated that some characteristics of the sutra could be characterized by the ACF and IACF factors.

1:20

1pSCb2. Autocorrelation-based features for speech representation. Yoichi ando (Kobe Univ., 1-4-132-105 Hiyodoridai, 1-4-132-105 Hiyodoridai, Kobe Kita 657-1123, Japan, andoy@cameo.plala.or.jp)

This study investigates autocorrelation-based features as a potential basis for phonetic and syllabic distinctions. The work comes out of a theory of auditory signal processing based on central monaural autocorrelation and binaural crosscorrelation representations. Correlation-based features are used to predict monaural and binaural perceptual attributes that are important for the architectural acoustic design of concert halls: pitch, timbre, loudness, duration, reverberation-related coloration, sound direction, apparent source width, and envelopment (Ando, 1985, 1998; Ando and Cariani, 2009). The current study investigates the use of features of monaural autocorrelation functions (ACFs) for representing phonetic elements (vowels), syllables (CV pairs), and phrases using a small set of temporal factors extracted from the short-term running ACF. These factors include listening level (loudness), zero-lag ACF peak width (spectral tilt), $\tau 1$ (voice pitch period), $\varphi 1$ (voice pitch strength), τe (effective duration of the ACF envelope, temporal repetitive continuity/contrast), segment duration, and $\Delta\varphi 1/\Delta t$ (the rate of pitch strength change, related to voice pitch attack-decay dynamics). Times at which ACF effective duration τe is minimal reflect rapid signal pattern changes that usefully demarcate segmental boundaries. Results suggest that vowels, CV syllables, and phrases can be distinguished on the basis of this ACF-derived feature set.

1:40

1pSCb3. Synthesis of the speech signals by using autocorrelation function. Shin-ichi Sato and Alejandro Bidondo (Ingeniería de Sonido, Universidad Nacional de Tres de Febrero, Varentín Gómez 4752, Caseros, Provincia de Buenos Aires 1678, Argentina, ssato@untref.edu.ar)

The running autocorrelation function (r-ACF) is obtained by the FFT method based on the Wiener–Khinchine theorem after obtaining the power density spectrum for a signal. This study attempted to reconstruct the original speech signal by using a part of its r-ACF. First, the stationary part of the vowel signals were investigated to determine until which delay time of the ACF (maximum time lag) is necessary to recognize the reconstructed signals as the original ones. Then, the continuous speech signals were investigated to determine the appropriate integration interval as well as the maximum time lag.

2:00

1pSCb4. Relationship between Intelligibility and autocorrelation factors of Japanese monosyllables. Ryota Shimokura, Sakie Akasaka, Hiroshi Hosoi, and Toshie Matsui (Dept. of Otorhinolaryngol. - Head and Neck Surgery, Nara Medical Univ., Shijo-cho 840, Kashihara, Nara 634-8522, Japan, rshimo@naramed-u.ac.jp)

Amplified speech sound by a hearing aid can improve the speech intelligibility for conductive hearing loss, while improve partly for sensorineural hearing loss. Although the indiscernible consonants in Japanese monosyllables have been identified, it is still unclear which acoustical feature in the consonants causes the difficulty on hearing. Therefore, the aim of this study is to clarify an influential factor on the speech intelligibility using autocorrelation function (ACF), which can capture temporal features of signals. The ACF factors were compared with percent articulations obtained by a Japanese speech intelligibility test ($n = 144$). The time-series factors were calculated by running ACF along a monosyllable, and the median values of the ACF factors were represented. As results, an effective duration (τ_e [ms]) of the ACF was correlated with the averaged percent articulations among the consonants ($r = 0.87$, $p < 0.01$). The τ_e indicates temporal fluctuation of speech signals including its fundamental frequency. The deteriorated perceptual function for temporal fluctuation may reduce the recognition ability of the consonants, so the application of the ACF analysis for a hearing aid may help the hearing of patients with sensorineural hearing loss.

2:20–2:40 Break

2:40

1pSCb5. Speaker recognition analysis using running autocorrelation function parameters. Alejandro Bidondo, Shin-ichi Sato, Ezequiel Kinigsberg, Mariano Arouxet, Sabater Andrés, Agustín Arias, Adrián Saavedra, and Ariel Groisman (Ciencia y Tecnología, Universidad Nacional de Tres de Febrero, Av. De Los Constituyentes 1426, Villa Maipú. San Martín. Buenos Aires 1650, Argentina, abidondo@untref.edu.ar)

The human brain's process associated with the recognition and identification of acoustic signals is regarded as the calculation of the distances between "sound vectors;" the ones listened in the present with the memorized vectors in previous listening. Matching vectors (minimal distance between them) would indicate sounds come from similar sound sources or same sound source. In this study, the statistical values of r-ACF (running autocorrelation function) microscopic parameters of 10 spoken words recordings (in Spanish) by the same and different speakers were calculated, and the aforementioned vector's distances were constructed by using the distances between the r-ACF parameters, being able to predict the degree of similarity between the speakers.

3:00

1pSCb6. Investigation of calculation methods of effective duration on autocorrelation-based Chinese speech recognition. Kun Peng Huang (GuoGuang Electric Co. Ltd., No.8 JingHu Rd., HuaDu, GuangZhou, GuangDong 510800, China, ohmycar@163.com) and Yoichi Ando (Grad. School of Sci. and Technol., Kobe Univ., Kobe, Japan)

The factors extracted from running autocorrelation function have phonetic and syllabic meanings, and they can be used for Chinese speech recognition. One of the factors is the effective duration of ACF envelope τe, which reveals the temporal repetitive continuity of the speech signals. The values of τe vary when different calculation methods are adopted. By adopting different calculation methods to several sets of Chinese vowels and characters, which are recorded from a group of speakers, a preferred calculation method is decided by the uniqueness of τe, that is the τe calculated by such method can mostly distinguish different vowels, characters, or even speakers.

Contributed Paper

3:20

1pSCb7. The role of normalization in phoneme recognition and speaker definition. Roy D. Patterson (Physiology, Development and Neurosci., Univ. of Cambridge, Downing Site, Cambridge, Cambridgeshire CB22 5LW, United Kingdom, rdp1@cam.ac.uk) and Toshio Irino (Faculty of Systems Eng., Wakayama Univ., Wakayama, Japan)

There is size information in speech sounds because the vocal tract and the vocal cords both grow as a child develops into an adult. Specifically, average pitch and mean formant frequency decrease as speaker size increases. Nevertheless, human speech recognition is effectively size invariant across the full range of sizes in the normal population of speakers and well beyond. It is also the case that listeners can discriminate speaker size with great accuracy; indeed, with greater accurately than they can discriminate the loudness of sound or the brightness of light. The first part of this talk describes how the peripheral auditory system normalizes speech sounds automatically to produce a size invariant representation for speech recognition. The second part presents a model of how the central auditory system transforms information in the cochlea into our perception of who is speaking and what they are saying. The model suggests that the system combines information about vocal resonator size with a small amount of contextual information to determine what the person is saying (at the phonological level), and then it adds voice pitch information to determine who is speaking (in the sense of the sex and size of the speaker).

3:40–4:40 Panel Discussion

Session 1pSCc

Speech Communication: Distinguishing Between Science and Pseudoscience in Forensic Acoustics II

Geoffrey Stewart Morrison, Cochair
Forensic Voice Comparison Lab., School of Elec. Eng. & Telecommunications, Univ. of New South Wales, Sydney, NSW 2052, Australia

James Harnsberger, Cochair
Univ. of Florida, 402 NW 24th St., Gainesville, FL 32607

Contributed Papers

1:00

1pSCc1. Mismatched distances from speakers to telephone in a forensic-voice-comparison case. Ewald Enzinger (Forensic Voice Comparison Lab., School of Elec. Eng. & Telecom., Univ. of New South Wales, Sydney, NSW 2052, Australia, e.enzinger@student.unsw.edu.au)

In a forensic-voice-comparison (FVC) case, one speaker (A) was talking on a mobile telephone, and another (B) was standing a short distance away. Later, B moved closer to the telephone. Shortly thereafter, there was a section of speech where the identity of the speaker was disputed. All material for training an FVC-system could be extracted from this single recording, but there was a near-far mismatch: Training data for A were near, training data for B were far, and the disputed speech was near. We describe a procedure for addressing the degree of validity and reliability of an FVC system under such conditions, prior to it being applied to the casework recording: Sections of recordings of pairs of speakers of known identity are used to train an A and a B model; multiple other sections from each of the A and B recordings are used as test data; a likelihood ratio is calculated for each test section; and system validity and reliability are assessed. Prior to training and testing, the A and B recordings were played through loudspeakers and rerecorded via a mobile-telephone network, B was rerecorded twice, once with the loudspeaker near and once with it far from the telephone.

1:20

1pSCc2. Exploring duration and spectral parameters of English /m/ for forensic speaker comparison. Colleen Kavanagh (Audio & Video Analysis Unit, RCMP, 1426 St. Joseph Blvd., Ottawa, ON K1A 0R2, Canada, colleen.kavanagh@rcmp-grc.gc.ca)

The speaker-specificity of five acoustic features of British English /m/ was explored from a forensic speaker comparison perspective. Normalized duration, center of gravity (COG), standard deviation (SD), and frequencies at peak and minimum amplitudes were measured for 30 adult male Standard Southern British English and Leeds English speakers. Spectral measurements were made in each of five frequency bands (0–0.5 kHz, 0.5–1 kHz, 1–2 kHz, 2–3 kHz, and 3–4 kHz) and calculated from a 40-ms window at the midpoint of each token. ANOVAs showed Speaker to be a highly significant factor for all variables. Discriminant analysis (DA) and likelihood ratio (LR) estimation assessed speaker discrimination with individual predictors and combinations thereof. Sample sizes limited the number of predictors in DA to eight; F-ratios were used to select the best predictors for analysis. The COG+SD (bands 1, 3, 4, 5) and Best 8 F-ratios (COG bands 1, 4, 5 + SD 1, 3, 4 + Peak 1, 4) tests achieved 53% and 49% correct classification, respectively. The Best 8 F-ratios and COG+SD tests also produced the best LR results, while COG+Peak performed similarly. DA and LR results for all predictor combinations will be presented and the most promising speaker comparison parameters highlighted.

1:40

1pSCc3. Examining long-term formant distributions as a discriminant in forensic speaker comparisons under a likelihood ratio framework. Erica Gold (Lang. and Linguist. Sci., The Univ. of York, Heslington, York YO10 5DD, United Kingdom, erica.gold@york.ac.uk), Peter French, and Philip Harrison (J P French Assoc., York, United Kingdom)

This study investigates the use of long-term formant distributions (LTFD) as a discriminant in forensic speaker comparisons. LTFD are the distributions calculated for all values of each formant for a speaker in a single recording. Spontaneous speech recordings from 100 male speakers of Southern Standard British English, aged 18–25 were analyzed from the DyViS Database (Nolan 2009). The recordings were auto-segmented to obtain a minimum of 50 s of vowels per speaker. The iCAbS (iterative cepstral analysis by synthesis) formant tracker was used to automatically extract and measure F1-F4 every 5 ms. To assess the evidential value of the LTFDs, likelihood ratios (LRs) were computed using a MatLab implementation of Aitken and Lucy's (2004) Multivariate Kernel-Density formula (Morrison 2007). It was found that LTFD performs well overall, but much better with different speaker comparisons than same speaker comparisons (97.76 % compared to 78% of comparisons providing correct support; Cllr = 0.9072 and EER = 5.47%). LTFD appears to be a good discriminant to include in forensic speaker comparison analyses and offers the added attraction of avoiding potential correlation problems between vowel phonemes.

2:00

1pSCc4. Establishing typicality: A closer look at individual formants. Vincent Hughes (Dept. of Lang. and Linguist. Sci., Univ. of York, Heslington, York YO10 5DD, United Kingdom, vh503@york.ac.uk)

Research into the forensic performance of individual formants has offered considerable evidence to support the traditional acoustic-phonetic view that whilst F1 and F2 encode broad phonetic contrast, higher formants may offer greater speaker-discriminatory potential (Peterson 1959, Ladefoged 2006, Clermont and Mokhtari 1998, Rose 2002). However, the comparative performance of formants has largely been assessed using posterior assessments of speaker-specificity (McDougall 2004, 2006; Clermont *et al.* 2008). Using quadratic polynomial fittings of F1 to F3 from spontaneous tokens of /ai/ extracted from all 100 speakers in the DyVis database (Nolan *et al.* 2009), this paper discusses issues relating to p(H|E)-based voice comparison analysis (particularly the use of discriminant analysis, DA). Further, DA performance is compared with an analysis based on likelihood ratios (LRs). LRs based on F3 are found to only marginally outperform F1 and F2 with regard to the magnitude of same-speaker and different-speaker strength of evidence, as well system performance metrics (EER and Cllr). The poorer than expected F3 LRs are assessed with regard to the distributions of values within- and between-speakers for the best performing F3 coefficient: the constant. The data go some way to establishing F3 population statistics, which may potentially be applied to voice comparison casework.

2:20–2:40 Break

2:40

1pSCc5. A likelihood ratio-based forensic voice comparison using formant trajectories of Thai diphthongs. Supawan Pingjai (CHL, Australian National Univ., Grad. House, Garran Rd., Canberra, NSW 0200, Australia, supawan.pingjai@anu.edu.au)

This study investigates the phonetic–acoustic properties of the three Thai diphthongs /i:a, ɨ:a, u:a/ within the context of forensic voice comparison. The likelihood-ratio approach is applied to the parameterized formant trajectories of each diphthong in order to evaluate their respective discriminatory power. The aim of this study is to assess to what extent such properties can be used to distinguish, in a probabilistic sense, two or more speech samples. Formant trajectories were fitted using both polynomial interpolation and the discrete cosine transform. Likelihood ratio values were derived by the multivariate kernel density (MVKD) estimation approach proposed by Aitken and Lucy (2004) and then calibrated by using the log-likelihood ratio cost function—Cllr (Brummer 2005) and the 95%-credible interval (Morrison *et al*., 2010). We have finished gathering all speech data for this study, and are currently processing the data using various computational tools. References: Aitken, C. G. G. and Lucy, D. "Evaluation of trace evidence in the form of multivariate data," App. Stat. **54**,109–122 (2004). Brümmer, N., FoCal Toolkit [software], 2005. See: http://www.dsp.sun.ac.za/~nbrummer/focal/. Morrison G. S., Thiruvaran T., and Epps J. (2010) "Estimating the likelihood-ratio output of a forensic-voice-comparison system," The Speaker and Language Recognition Workshop, Brno.

3:00

1pSCc6. Fusion of multiple formant-trajectory- and fundamental-frequency-based forensic-voice-comparison systems: Chinese /ei1/, /ai2/, and /iau1/. Cuiling Zhang (Dept. of Forensic Sci. & Technol., China Criminal Police Univ., Tawan St. NO.83, Huanggu District, Shenyang, Liaoning 110854, China, cuiling-zhang@forensic-voice-comparison.net) and Ewald Enzinger (Forensic Voice Comparison Lab.,School of Elec. Eng. & Telecommunications, Univ. of New South Wales, Sydney, NSW, Australia)

This study investigates the fusion of multiple formant-trajectory- and fundamental-frequency-trajectory-based (f0-trajectory-based) forensic-voice-comparison systems. Each system was based on tokens of a single phoneme: tokens of Chinese /ei1/, /ai2/, and /iau1/ (numbers indicate tones). Human-supervised formant-trajectory and f0-trajectory measurements were made on tokens from a database of recordings of 60 female speakers of Chinese. Discrete cosine transforms (DCT) were fitted to the trajectories and the DCT coefficients used to calculate likelihood ratios via the multivariate kernel density (MVKD) formula. The individual-phoneme systems were fused with each other and with a baseline mel-frequency cepstral-coefficient (MFCC) Gaussian-mixture-model universal-background-model (GMM-UBM). The latter made use of the entire speech-active portion of the recordings. Tests were conducted using high-quality recordings as nominal suspect samples and mobile-to-landline transmitted recordings as nominal offender samples. Fusion of the phoneme-systems with the baseline system via logistic regression did not lead to any substantial improvement in validity and reliability deteriorated.

3:20–4:00 Panel Discussion

MONDAY AFTERNOON, 3 JUNE 2013 510A, 1:00 P.M. TO 3:20 P.M.

Session 1pSPa

Signal Processing in Acoustics, Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Nearfield Acoustical Holography (NAH) Measurements and Applications

Alba Granados, Cochair
Dept. Elec. Eng., Tech. Univ. of Denmark, DTU Bldg. 352, Kongens Lyngby DK-2800, Denmark

Alan T. Wall, Cochair
Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602

Contributed Papers

1:00

1pSPa1. Regularized reconstruction of sound fields with a spherical microphone array. Alba Granados and Finn Jacobsen (Dept. of Elec. Eng., Tech. Univ. of Denmark, DTU Bldg. 352, Kongens Lyngby DK-2800, Denmark, fja@elektro.dtu.dk)

Spherical near field acoustic holography with microphones mounted on a rigid spherical surface is used to reconstruct the incident sound field. However, reconstruction outside the sphere is an ill-posed inverse problem, and since this is very sensitive to the measurement noise, straightforward implementation might lead to disastrous reconstructions. A large number of regularization tools based on singular value decomposition are available, and it has been found that the acoustic holography problem for certain geometries can be formulated in such a way that similarities to singular value decomposition become apparent. Hence, a number of regularization methods, including truncated singular value decomposition, standard Tikhonov, constrained Tikhonov, iterative Tikhonov, Landweber and Rutishauser, have been adapted for spherical near field acoustic holography. The accuracy of the methods is examined by means of simulations and measurements, which leads to practical recommendations on the use of regularization techniques regarding space and frequency.

1:20

1pSPa2. Quasi-holographic processing as an alternative to synthetic aperture sonar imaging. David J. Zartman, Daniel S. Plotnick (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, zartman.david@gmail.com), Timothy M. Marston (Naval Surface Warfare Ctr., Panama City Div., Panama City, FL), and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA)

By limiting attention to supersonic-like wavevectors, time-resolved holographic imaging was demonstrated to be helpful for identifying transient elastic responses of targets contributing to far field scattering [Hefner and Marston, Acoust. Res. Lett. Online **2**, 55–60 (2001); **3**, 101–106

(2002)]. That approach was applied to time-evolving three-dimensional wavefields scanned with a two-dimensional raster scan at a significant offset. In recent work, however, it was also demonstrated that line scan measurements of bistatic scattering could be processed quasi-holographically as an alternative to synthetic aperture sonar (SAS) imaging [Baik *et al*., J. Acoust. Soc. Am. **130**, 3838–3851 (2011)]. The present investigation broadens the line scan approach to reversible signal extraction associated with image features, and to monostatic quasi-holographic imaging, in which the source and receiver are co-located and scanned along a line. Some applications of this approach will be illustrated such as separation of signals from edge diffraction features and specular features from those of elastic features. [Work supported by ONR and the NSWC-PCD ILIR/IAR research fund.]

1:40

1pSPa3. Optimized two-dimensional imaging of transient sound fields using a hybrid transient acoustic holography. Siwei Pan and Weikang Jiang (State Key Lab. of Mech. System and Vib., Shanghai Jiao Tong Univ., 800 Dong Chuan Rd., Shanghai 200240, China, swpan@sjtu.edu.cn)

A hybrid transient acoustic holography (HTAH) is presented to visualize the two-dimensional transient sound fields radiated from planar sources with unknown locations and dimensions, by combining the time reversal source localization with the near-field acoustic holography (NAH) based on the interpolated time domain equivalent source method (TDESM). Based on the near-field measurements with a microphone array, the time reversal source focusing algorithm is used to find out the hotspots of the sound sources on the equivalent source plane, which suggests the collocation of equivalent sources. The interpolated TDESM based NAH is then employed to reconstruct and image the transient sound field on the reconstruction plane. The proposed HTAH technique can reduce the elements number of microphone array by only collocating the equivalent sources in the vicinity of the “real” sound sources. The visualization of the transient sound fields radiated from single-planar-piston and dual-planar-piston model is studied by numerical simulations, respectively. The experiments are performed in a semi-anechoic chamber by using two baffled loudspeakers. Both the simulation and experimental results revealed that this hybrid scheme can realize a better two-dimensional imaging of transient sound fields than the original interpolated TDESM based NAH in the measurement using same amount of microphones.

2:00

1pSPa4. Modified statistically optimized near-field acoustical holography for jet noise characterization. Alan T. Wall, Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, alantwall@gmail.com)

Near-field acoustical holography has been shown to be a useful tool for visualizing jet noise fields. It has been applied to a full-scale jet on an installed military aircraft with promising results, but the source characteristics in the extreme near field have been difficult to characterize because of the interference of acoustic reflections off the rigid reflecting plane beneath the jet. To provide accurate sound field reconstructions, a modified approach to statistically optimized near-field acoustical holography (SONAH) is implemented. In conventional SONAH, the sound field is represented by a matrix of elementary wave functions at all desired spatial locations. In this modified approach, advantage is taken of the property that arbitrary, user-defined functions can be selected for this matrix. Here, two sets of cylindrical wave functions, one centered on the jet centerline and one on the image source centerline, are used to obtain an accurate near-field reconstruction.

2:20

1pSPa5. A study on the effect of sound velocity estimation for underground imaging. Ryo Toh, Takuya Sakuma, and Seiichi Motooka (Dept. of Elec. Electron. and Comput. Eng., Chiba Inst. of Technol., 2-17-1, Tsudanuma, Narashino, Chiba 275-0016, Japan, liang.tao@it-chiba.ac.jp)

An efficient technique for detecting objects buried underground is expected for archaeological exploration and civil engineering. We have proposed a three dimensional underground imaging method by using an amplitude correlation synthesis processing and an electromagnetic induction type sound source. Because the sound velocity of sand and soil depends on a lot of physical parameters that varies with the area, it is needed to be measured before imaging processing. Up to now, the sound velocity employed for signal processing is measured by receivers buried underground near the area to be detected. In order to improve the efficiency of field detection, a method of estimating the underground sound velocity by the sound velocity measured from the ground surface is studied. In this study, the comparison experimental results of the velocity of underground transverse wave, the velocity of underground longitudinal wave, and the sound velocity measured from the ground surface are introduced. Furthermore, the effect on the image of underground objects brought forth by the sound velocity estimation is discussed.

2:40

1pSPa6. Improved hydrophone calibration by combining acoustic holography with the radiation force balance measurements. Sergey Tsysar (Phys. Faculty, Lomonosov Moscow State Univ., GSP-1, 1-2 leninskie Gory, Moscow 119991, Russian Federation, sergey@acs366.phys.msu.ru), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Oleg Sapozhnikov (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation)

Ultrasound sources are frequently characterized by the radiation force (RF) balance method that is based on the relation between the total acoustic power and RF on absorbing or reflecting targets. This relation is usually taken from the plane-wave approximation or with a geometrical correction for focused sources. However, real sources emit inhomogeneous acoustic beams. Acoustic holography is a method of recording the true field by measuring both pressure magnitude and phase over a 2D surface (a hologram). The hologram makes it possible to accurately calculate the radiation stress tensor on the surface of the absorbing target. Such measurements allow the relation of hydrophone sensitivity with measured RF based on the known exact expression for RF as a function of the 2D pattern of acoustic pressure and particle velocity. This suggests an improved approach for single-frequency hydrophone calibration that benefits from the inherent accuracy of mass balances as well as the fact that pressure approximately scales with one half of the accuracy of the RF measurement. In the current study this approach was used to calibrate a hydrophone by characterizing a 1-MHz flat piezoceramic source in water using acoustic holography and a RF balance. [Work supported by RFBR and NIH EB007643.]

3:00

1pSPa7. Multi-spectral acoustic imaging on object surface in air. Xinhua Guo, Yosuke Mizuno, and Kentaro Nakamura (Precision and Intelligence Lab., Tokyo Inst. of Technol., 4259-R2-26 Nagatsuta, Midori-ku, Yokohama, JAPAN, Yokohama 226-8503, Japan, guoxinhua@sonic.pi.titech.ac.jp)

Acoustical imaging has been performed using mono-frequency or a limited number of frequencies in the previous studies. The frequency dependence, however, may provide rich information on surface profiles, structures hidden under surfaces, and material properties of objects. In this study, acoustic imaging on object surfaces was conducted over a wide frequency range with a fine frequency step. A rigid surface with different profiles and a boundary between two objects composed of different materials were illuminated by sound wave swept over the frequency range from 1 kHz to 20 kHz with a 30-Hz step. The scattered sound field was recorded two-dimensionally using a scanning microphone, and the holographic method was used to reconstruct the sound pressure distribution on the surface from the recorded data. From the experimental results, the characteristics of the surfaces with respect to the shapes and material properties were demonstrated experimentally. The depth of the holes was identified by its own resonance frequency, and the two different materials were successfully distinguished by multiple images obtained at different frequencies.

Session 1pSPb

Signal Processing in Acoustics: Acoustic Feature Extraction and Characterization

Edmund J. Sullivan, Chair
Prometheus Inc., 46 Lawton Brook Lane, Portsmouth, RI 02871

Contributed Papers

3:20

1pSPb1. Classifying sonar signals with varying signal-to-noise ratio and bandwidth. Stefan Murphy (Underwater Sensing, Defence Res. and Development Canada, 9 Grove St., Dartmouth, NS B2Y 3Z7, Canada, stefan.murphy@drdc-rddc.gc.ca)

An automatic aural classifier developed at Defence Research and Development Canada has demonstrated the ability to distinguish target echoes from clutter using perceptual-based features inspired by sonar operators. Initially, the classifier was tested with echoes from explosive sources, but more recent research involved transmitting broadband waveforms from transducer sources. In sonar transducer operation, there is a trade off between source level and bandwidth, and the goal of this paper is to study how these factors affect echo classification. Source level relates to signal-to-noise ratio (SNR), which inherently affects classification since signals with low enough SNR cannot be distinguished from noise, let alone other signals. The dependence of classification performance on bandwidth is less obvious; however, the aural classification technique is based on a sub-band type of processing that mimics the basilar membrane in the human auditory system, and this model is not well adapted for narrow bands. Performance of the aural classifier is therefore expected to degrade as bandwidth is decreased. In this paper, the effect of SNR and signal bandwidth on echo classification is examined using echoes of varying SNR, and in various bands selected using band-pass filters.

3:40

1pSPb2. An automated framework for the extraction of ultrasonic echoes embedded in noise. Adam Pedrycz, Henri-Pierre Valero, Hiroshi Hori, Kojiro Nishimiya, Hitoshi Sugiyama, and Yoshino Sakata (Acoust.-Sonic Eng., Schlumberger Ltd., 2-2-1 Fuchinobe, Chuo-ku, Sagamihara-shi, Kanagawa-ken 229-0006, Japan, APedrycz@slb.com)

Proposed is an automated framework for the extraction and characterization of the arriving echo in ultrasonic signals embedded in high noise. Commonly, in order to correctly characterize the first echo hidden within a noise-ridden signal, multiple traces are stacked in a gather to improve the SNR, hence facilitating easier extraction and characterization of the recorded echo. Such first order statistical methods require multiple traces and usually fall short in the accuracy of the echo estimate when the variance of the noise does not belong to a known distribution. To mitigate this problem, a framework has been developed comprised of a multi-step procedure, i.e., pre-processing, localization, gating, and finally parameterization of the given echo. This automatic framework operates on single traces and does not require the setting of processing parameters. By means of this method, the true echo can be extracted in one-shot from other overlapping noise components. Furthermore, because the method operates on a trace-by-trace basis, it is insensitive to large non-stationarities in the baseline. Experiments conducted using synthetic as well as aluminum reflector pulse-echo lab data demonstrate the effective extraction of the true echo under the presence of noise and ringing at varying levels of severity.

4:00

1pSPb3. Extract voice information using high-speed camera. Mariko Akutsu, Yasuhiro Oikawa, and Yoshio Yamasaki (Dept. of Intermedia Art and Sci., Waseda Univ., 3-4-1 Ohkubo, Shinjuku, Tokyo, Japan, yoikawa@waseda.jp)

Conversation is one of the most important channels for human beings. To help communications, speech recognition technologies have been developed. Above all, in a conversation, not only contents of utterances but also intonations and tones include important information regarding a speaker's intention. To study the sphere of human speech, microphones are typically used to record voices. However, since microphones have to be set around a space, their existences affect a physical behavior of the sound field. To challenge this problem, we have suggested a recording method using a high-speed camera. By using a high-speed camera for recording sound vibrations, it can record two or more points within the range of the camera at the same time and can record from a distance, without interfering with the sound fields. In this study, we extract voice information using high-speed videos, which capture both a face and a cervical part of the subject. This method allows recording skin vibrations, which contain voices with individuality and extrapolating sound waves by using an image processing method. The result of the experiment shows that a high-speed camera is capable of recording voice information.

4:20

1pSPb4. Multi-stage identification for abnormal/warning sounds detection based on maximum likelihood classification. Kohei Hayashida, Junpei Ogawa, Masato Nakayama, Takanobu Nishiura, and Yoichi Yamashita (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, cm012063@ed.ritsumei.ac.jp)

In recent years, the methods utilizing environmental sounds have been increasingly employed for monitoring the safety of the elder who lives in distant place. Environmental sounds should consist of various sounds in daily life, and identified ones enables to detect abnormity. To detect abnormity, it is therefore required that abnormal/warning sounds are accurately identified among environmental sounds. In the past, environmental sound identification methods have generally utilized acoustic models constructed by each sound source for all environmental sounds. In our former research, we proposed multi-stage identification for detecting abnormal/warning sounds accurately. However, these methods design individual acoustic models from similar sounds. Therefore, the sound identification performance is degraded. To overcome this problem, in this study, we proposed environmental sound classification based on acoustic features for model construction. The proposed method classifies environmental sounds based on the difference of acoustic likelihood and designs acoustic models are constructed by classification results. Moreover, the proposed method detects the abnormal/warning sounds more accurately by combining it with the multi-stage identification. We carried out the evaluation experiment with environmental sound database. Experimental results of this experiment demonstrate that the identification performance for the proposed method is higher than that for the conventional methods.

4:40

1pSPb5. Abnormal events recognition and classification for pipeline monitoring systems based on vibration analysis and artificial neural networks. Bin Chen (School of Automation, Beijing Univ. of Posts and Telecommunications, Beijing, China) and Xiaobin Cheng (Key Lab. of Noise and Vib. Res., Inst. of Acoust., Chinese Acad. of Sci., 21, Beisihuanxilu Rd., Haidian District, Beijing 100190, China, xb_cheng@mail.ioa.ac.cn)

Pipelines have become the principal means of oil and gas transportation. However, pipeline leakage takes place due to some natural or artificial damages, which may cause loss of life and properties along with the environmental pollutions. A new pipeline detection and pre-warning system based on distributed optical fiber sensor is proposed, and the hardware has been accomplished. Now, its following key problem is how to recognize and classify the abnormal events, such as oil stealing, construction, artificial excavation, motor work, and train passing. This paper involves a study on this and proposes a solution method. First, original vibration signal is pre-processed and segmented according to threshold of energy within a narrower bandwidth. Then, event features in time and frequency domain are analyzed through statistical analysis and short-time Fourier transform (STFT). The energy coefficients at some bandwidth can distinguish different type of abnormal events, which are chosen as feature vectors. At classification, abnormal events are first divided into discrete and continuous events with single classifier, which can decrease classified event sets and improve recognition accuracy. Then, BP artificial neural network is applied to identify the type of abnormal events. Finally, proposed method will be verified with actual collection data sets.

5:00

1pSPb6. Backward waves and quasi-resonance of shells and invariants of the time reversal operator. Philippe D. Franck, Clorennec Dominique, Maximin Cès, Romain Anankine, and Claire Prada (Institut Langevin, ESPCI ParisTech, CNRS UMR 7587, 1 rue Jussieu, Paris 75006, France, claire.prada-julia@espci.fr)

Backward waves propagating on shell are guided modes with opposite phase and group velocities. For a shell in vacuum, backward modes are linked to zero group velocity modes and resonances, which have been the object of recent studies. For a shell embedded in water, the group velocity does not vanish because of the leakage into the fluid. However, the group velocity of the backward mode has a minimum associated to a quasi-resonance. These phenomena are studied on air filled steel and zircaloy hollow cylinders, using a 3 MHz linear array in pulse echo mode. The circumferential guided modes are generated and their radiation into water detected by the array. The modes are separated using the decomposition of the time reversal operator (TRO), each pair of counter-propagating modes being associated to two invariants of the TRO [Prada *et al*. J. Acoust. Soc. Am. (1998)]. Two resonances are revealed by the eigenvalues of the TRO, one is associated with the first longitudinal thickness resonance and the other, very high, occurring at a slightly lower frequency, corresponds to the minimum of the group velocity of the backward mode. The back-propagations of the eigenvectors of the TRO provide the phase velocities of these modes.

MONDAY AFTERNOON, 3 JUNE 2013 — 516, 1:00 P.M. TO 5:00 P.M.

Session 1pSPc

Signal Processing in Acoustics: Miscellaneous Topics in Signal Processing in Acoustics (Poster Session)

K. Thomas Wong, Cochair
Elec. Eng., Hong Kong Polytechnic Univ., Hong Kong, Hong Kong

Jens Meyer, Cochair
mh Acoust., 38 Meade Rd., Fairfax, VT 05454

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.

1pSPc1. Objective analysis of higher-order ambisonics sound-field reproduction for hearing aid applications. Chris Orinos and Jorg M. Buchholz (National Acoust. Lab., Australian Hearing, 126 Greville St., Sydney, NSW 2067, Australia, chris.orinos@nal.gov.au)

The evaluation of hearing aids (HAs) inside realistic sound environments is of increasing interest. Higher-order Ambisonics (HOA) has been used for loudspeaker-based sound field resynthesis and HOA recording microphone arrays are available. Although HOA has been evaluated perceptually, it is unclear how far the results can be transferred to evaluating HA technologies (particularly multi-microphone enhancement algorithms). In order to determine the minimum HOA order required for HA testing, an HOA framework was developed, simulating the entire path from sound presented in a room, picked up by a microphone array, decoded, and received at the ears of a HA-fitted dummy head. HA directivity patterns were compared between an ideal free-field and its HOA representation to evaluate the introduced error. In-room analysis was conducted to investigate the bandwidth and performance of a directional microphone in realistic situations. For a bandwidth B, the required order M_{min} was found to be $M_{min} \geq B/600$ Hz for the anechoic (worst) case scenario. The presence of reverberation introduced natural room response variations across different source-receiver locations, suggesting that the acceptable HOA error can be increased. Hence, in reverberant environments, the required HOA order is reduced, and at least 2D HOA reproduction can be used for evaluation of HA technologies.

1pSPc2. Linearized versus non-linear inverse methods for seismic localization of underground sources. Geok Lian Oh and Finn Jacobsen (Acoust. Technol., Tech. Univ. of Denmark, 25 Jalan Sempadan #03-06, Singapore 457400, Singapore, gloh@elektro.dtu.dk)

The problem of localization of underground sources from seismic measurements detected by several geophones located on the ground surface is addressed. Two main approaches to the solution of the problem are considered—a beamforming approach that is derived from the linearized inversion problem, and the Bayes nonlinear inversion method. The travel times used in the beamformer are derived from solving the Eikonal equation. In the linearized inversion method, we assume that the elastic waves are predominantly acoustic waves, and the acoustic approximation is applied. For the nonlinear inverse method, we apply the Bayesian framework where the misfit function is the posterior probability distribution of the model space. The model parameters are the location of the seismic source that we are interested in estimating. The forward problem solver applied for the nonlinear inverse method is a finite difference elastic wave-field numerical method. In this paper, the accuracy and performance of the linear beamformer and nonlinear inverse methods to localize a underground seismic source are checked and compared using computer generated synthetic experimental data.

1pSPc3. B-format for binaural listening of higher order Ambisonics.
Ryouichi Nishimura (National Inst. of Information and Commun. Technol., 2-2-2 Hikaridai, Seika-cho, Soraku-gun, Kyoto 619-0288, Japan, ryou@nict.go.jp) and Kotaro Sonoda (Nagasaki Univ., Nagasaki, Japan)

B-format is a four-channel signal capable of rendering a sound scene with spatial information. It can be regarded equivalent to first order ambisonics. Ambisonics requires a high order to contain precise spatial information, and higher order ambisonics requires an exponentially large amount of data. This limitation comes from the fact that the original aim of ambisonics is to reproduce the whole sound field. However, as mobile devices are prevalent, users often listen to sound media through earphones. Because nowadays users can hold sound contents individually, one can assume that sound contents could be produced adaptively to each user. Here we propose a way to make B-format signals more suitable for individual binaural listening. We assume that the production side can capture a sound scene with higher order ambisonics, because it may be processed for enterprise applications. Under this assumption, the binaural signal is once generated from the higher order ambisonics, and then its B-format signal is obtained by inversely processing the signal, assuming the first order ambisonics. Computer simulations show that interaural phase differences (IPDs) are improved at a frequency region where IPD dominantly affects sound localization. Results of hearing tests are also discussed.

1pSPc4. Steering for listening area of reflective audio spot with parametric loudspeaker array.
Shohei Masunaga, Daisuke Ikefuji, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is037089@ed.ritsumei.ac.jp)

A parametric loudspeaker has a high directivity by utilizing an ultrasound wave as a carrier wave. Therefore, a parametric loudspeaker can form a specific listening spot called “audio spot.” Furthermore, a parametric loudspeaker can form the “reflective audio spot” by utilizing the reflected sound. The listeners in the listening area may perceive the acoustic sound image on the reflector. However, it has the problem that the reflective audio spot with a single parametric loudspeaker is narrow area. Therefore, it is difficult for several listeners to perceive the reflective audio spot at the same time. If we can expand the area of the reflective audio spot, several listeners can perceive the reflective audio spot at the same time. Thus. in this paper, we attempt to steer the area of the reflective audio spot with parametric loudspeaker array. We carried out objective and subjective evaluation experiments to confirm the effectiveness of the proposed system in a conference room. As a result, we confirmed the proposed system can expand the are of the reflective audio spot.

1pSPc5. Steerable parametric loudspeaker with preprocessing methods.
Chuang Shi and Woon-Seng Gan (School of Elec. & Electron. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave., S2-B4a-03, DSP Lab, Singapore 639798, Singapore, shichuang@ntu.edu.sg)

The emerging applications of the parametric loudspeaker, such as 3D audio, require both directivity control and high fidelity at the audible frequency (i.e., the difference frequency of the primary frequencies generated by the parametric loudspeaker). Although the phased array techniques have been applied and proved adequate to adjust the steering angles of the parametric loudspeaker, and preprocessing methods have been studied to reduce the harmonic distortions, there is no published work on the effectiveness of the combination of the beamsteering method and the preprocessing methods for the broadband steerable sound beam system. This paper aims to investigate on this unexplored problem. First, the relation between the phases of the primary waves and the difference frequency wave is explored to prove the feasibility of achieving a broadband steerable sound beam from the parametric loudspeaker with preprocessing methods. Second, based on the derived relation, the beamsteering structure is proposed. Lastly, preprocessing methods are proposed for the steerable parametric loudspeaker using double sideband modulation (DSBAM) and square root amplitude modulation (SRAM) methods. Spatial performances of the steerable parametric loudspeaker with preprocessing methods are presented in this paper.

1pSPc6. Evaluation of different spatial windows for a multi-channel audio interpolation system.
Jorge A. Trevino Lopez (Grad. School of Information Sci./Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 9808577, Japan, jorge@ais.riec.tohoku.ac.jp), Takuma Okamoto (Universal Media Res. Ctr., National Inst. of Information and Commun. Technol., Seika, Kyoto, Japan), Yukio Iwaya (Faculty of Eng., Tohoku Gakuin Univ., Tagajo, Miyagi, Japan), and Yôiti Suzuki (Grad. School of Information Sci./Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Miyagi, Japan)

The adoption rate of multi-channel audio systems has dramatically increased in recent years. It is common to find 5.1- or 7.1-channel systems in typical home theaters. However, most users do not setup the satellite loudspeakers at the prescribed positions for aesthetic reasons or due to space constraints. Recently, we introduced a technique to optimize multi-channel contents for reproduction over non-ideal loudspeaker setups. Our proposal improves localization accuracy for sound sources reproduced by horizontal loudspeaker arrays. It can also be extended to handle full 3D contents, like those of the upcoming 22.2-channel standard. The proposed method works by applying a set of spatial windows centered at the loudspeaker positions and interpolating along the angles using the spherical harmonics. We now extend our previous results by evaluating the performance of six different spatial window functions. We consider the 5-channel distribution of ITU recommendation BS.775-2, as well as four variations that end-users are likely to deploy. Apparent sound source locations are estimated from the energy and velocity vectors at the sweet spot. Our study found that using the Slepian window with our proposal and a non-ideal loudspeaker layout leads to a reproduced sound field that is closer to that of the ideal configuration.

1pSPc7. Including frequency-dependent attenuation for the deconvolution of ultrasonic signals.
Ewen Carcreff, Sébastien Bourguignon, Jérôme Idier (IRCCyN, 1 rue de la Noë, Nantes 44321, France, ewen.carcreff@irccyn.ec-nantes.fr), and Laurent Simon (LAUM, Le Mans, France)

Ultrasonic non-destructive testing (NDT) is a standard process for detecting flaws or discontinuities in industrial parts. A pulse is emitted by an ultrasonic transducer through a material, and a reflected wave is produced at each impedance change. In many cases, echoes can overlap in the received signal and deconvolution can be applied to perform echo separation and to enhance the resolution. Common deconvolution techniques assume that the shape of the echoes is invariant to the propagation distance. This can cause poor performances with materials such as plastics or composites, in particular because acoustic propagation suffers from frequency-dependent attenuation. In geophysics, biomedical imaging or NDT, various frequency-dependent attenuation models have been proposed under different formulations. This communication compares the related possible constructions in order to account for attenuation in deconvolution methods. Especially, we introduce a discrete model for the data, that includes an attenuation matrix in the standard convolution model. Experimental data acquired from Plexiglas plates show that, for this material, attenuation varies roughly linearly with frequency, leading to a unique parameter identification. Finally, we show that such an advanced model manages a better fitting of the data, and promises improvement for the deconvolution of complex ultrasonic data.

1pSPc8. Comparative signal processing analyses of a speed-dependent problem as motivated by brake judder problem.
Osman T. Sen (Mech. Eng., Istanbul Techn. Univ., Inonu Cad. No: 65 Gumussuyu, Istanbul 34437, Turkey, senos@itu.edu.tr), Jason T. Dreyer, and Rajendra Singh (Mech. and Aerosp. Eng., The Ohio State Univ., Columbus, OH)

The goal of this paper is to investigate a transient problem using several digital signal processing techniques. First, a simple linear mathematical model, where a point mass is connected to a roller through a contact interface, is developed and the dynamic interfacial force is analytically calculated as a function of the speed. In this model, the contact interface is

described with a linear spring and viscous damper, and the system is excited with a base excitation, as defined by the undulations on the roller surface. Due to the time-varying speed characteristics of the roller, the resulting response is transient. Second, the dual-domain analyses of the calculated system response is carried out by using short-time Fourier and wavelet transforms, since single-domain representation leads to a loss of information due to signal's transient characteristics. Third, the Hilbert transform is applied and the envelope curves of the interfacial force response are successfully obtained. Finally, this problem is briefly linked to brake judder phenomenon and its source regimes are briefly explained.

1pSPc9. Fast Bayesian hierarchical inference via sparsity enforcing *a priori* for aeroacoustic source imaging. Ning Chu, Ali Djafari (Groupe de problème inverse, Laboratoire des signaux et systèmes (l2s), SUPELEC, SUPELEC, plateau de Moulon, 3 rue Joliot-Curie, 91192 GIF-SUR-YVETTE Cedex (France), Gif sur yvette, Paris 91192, France, chuning1983@gmail.com), José Picheral (Dept. Signal et Systèmes Electroniques, SUPELEC, Paris, France), and Nicolas Gac (Groupe de problème inverse, Laboratoire des signaux et systèmes (l2s), SUPELEC, Paris, France)

Aeroacoustic imaging is a technique for mapping the positions and powers of aeroacoustic sources. We propose a novel inverse solution by applying Bayesian hierarchical inference via sparsity enforcing *a priori*. We model the sparse prior of source powers by using the double exponential distribution, which can greatly improve the spatial resolution and robustness to background noise. Hyperparameters and source powers can be alternatively estimated based on the joint maximum *a priori* optimization. To accelerate the optimization, we improve the forward model of aeroacoustic power propagation by exploring the convolution operator. Finally, our approach is compared with some classical methods on simulated and real data. And our approach is feasible to apply for aeroacoustic imaging with the 2D non-uniform microphone array in wind tunnel tests, especially for near-field monopole and extended source imaging.

1pSPc10. Low latency audio coder design for high quality audio service on server-client environment. Han-gil Moon, Nam-suk Lee, and Hyun-wook Kim (DMC R&D Ctr., Samsung Electron., 416, Maetan 3-dong, Yeongtong-gu, Suwon 443-742, South Korea, hangil.moon@samsung.com)

Low latency audio coding attracts increasing attention among high quality communication applications such as video conferencing system and server-client media applications such as cloud computing based interactive A/V service. This paper presents a low latency audio coding scheme which can achieve both low delay and high subjective audio quality at the same time. In order to guarantee low delay, the proposed coding system incorporates low overlap window while preserving the window size as same as that of conventional (AAC) long window. The supplementary signal processing tool is incorporated to enhance the audio quality. The proposed coding scheme achieves the delay of 24 ms in 48 kHz and the MUSHRA score, which is comparable to that of commercialized AAC.

1pSPc11. Security screening using ultrasound. David Hutchins, Lee Davis, and Sheldon Tsen (School of Eng., Univ. of Warwick, Gibbet Hill Rd., Coventry CV4 7AL, United Kingdom, D.A.Hutchins@warwick.ac.uk)

This work will demonstrate that it is possible to produce images of hidden objects, using ultrasound transmitted through air. For example, it can be shown that a knife can be imaged, when hidden behind a layer of clothing fabric. To achieve this, it is necessary to use coded waveforms and signal recovery techniques, in order to retrieve small signals in the presence of a much larger reflection from the outer fabric surface. In addition, ultrasound can be used in through-transmission to detect hidden objects within thin packages. This and other examples of the use of air-coupled ultrasound for security work will be demonstrated.

1pSPc12. Articulatory-based speaker recognition. Luis Rodrigues (Concordia Univ., 1515 St. Catherine W, EV12.111, Montreal, QC H3G2W1, Canada, luisrod@encs.concordia.ca) and John Kroeker (Eliza Corp., Beverly, MA)

This paper presents a new methodology for computational speaker recognition based on a mathematical model of articulatory speech production. The method, based on articulatory phonology is tested on the MOCHA database for recognizing a male speaker and a female speaker. From an engineering perspective, in articulatory phonology one is interested in the trajectories over time of a set of articulators. These time trajectories are associated with the production of speech. The basic phonological unit in articulatory phonology is the articulatory gesture, which is defined as a dynamic system specified by a characteristic set of parameters. This dynamic system receives as inputs a target state and a set of parameters that tune the system to the desired action. The output is the solution of the state equation, i.e., the state trajectory, where the sate is formed by the x-y positions of the important articulators that describe human speech. The state trajectory is then mapped to the output speech waveform by emulating the human vocal tract, through the observation equation and the MFCCs frequency description. A simplification of this model will be used in this paper for speaker recognition with 100% success in recognizing a male and a female speaker.

1pSPc13. A dynamic automatic noisy speech recognition system for a single-channel hybrid noisy industrial environment. Sheuli Paul (Univ. of Kaiserslautern, Kaiserslautern, Kaiserslautern 67663, Germany, paul@eit.uni-kl.de)

A dynamic noisy speech recognition system is developed to recognize single-channel small spoken commands in a hybrid noisy industrial environment. This hybrid system has three parts: (a) hybrid pre-processing to enhance noisy speech, (b) feature extraction for perceptual speech features, (c) classification and recognition for the DANSR's result. Here, the single-channel is only one microphone, and the hybrid noise is environmental mixed noise distinguished as: (i) strong, (ii) time varying steady-unsteady, and (iii) mild. A new adaptive feature extraction technique based on local trigonometric transformation (LTT) is introduced and examined. This is adapted with psychoacoustic quantities such as Bark scaled critical band spectrum, loudness scale, and perceptual entropy. Here the spectral analysis is done by rising cut-off function, folding operation, and discrete cosine transformation (DCT-IV) instead of Fourier transform. Then, inverse DCT-IV and unfolding operation result in perceptual LTT (PLTT) features. These are recognized by hidden Markov model (HMM). The new PLTT features are more efficient and perceptually meaningful than the standard feature extraction techniques. The DANSR system is a novel solution for small commands to a long existing hybrid noise problem.

1pSPc14. A novel noise-reduction algorithm for real-time speech processing. Frederic E. Theunissen and Tyler Lee (Psychology and Neurosci., UC Berkeley, 3210 Tolman Hall, Berkeley, CA 94720, theunissen@berkeley.edu)

We developed a new noise-reduction algorithm based on a joint spectro-temporal representation of signals. The algorithm was inspired by the discovery in our laboratory of higher-level avian auditory cortical neurons that showed invariant responses to communication signals. The algorithm consists of an analysis step and a synthesis step. In the analysis step, the sound is first decomposed into narrow band signals by a frequency filter bank. These time-frequency waveforms are then further analyzed using a spectro-temporal modulation filter bank to obtain a representation that is akin to the one generated by cortical neurons. In our algorithm, this modulation filter bank was obtained from the principal component analysis of the speech signal in the time-frequency representation. We then learned which subset of the modulation filters provided the best information to extract the signal from the noise. In the synthesis step, we then used this subset of spectral-temporal modulation feature detectors to generate a set of time-varying frequency gains that could be applied directly to the original time frequency decomposition. In this manner, we were able to perform noise reduction in real time and with minimal delay. Our algorithm yielded similar noise reduction but better quality speech quality than current state-of-the-art algorithms.

1pSPc15. Causal binary mask estimation for speech enhancement using sparsity constraints. Abigail A. Kressner, David V. Anderson, and Christopher J. Rozell (School of Elec. and Comput. Eng., Georgia Inst. of Technol., 58 6th St NE Unit 2608, Atlanta, GA 30308, abbiekre@gatech.edu)

While most single-channel noise reduction algorithms fail to improve speech intelligibility, the ideal binary mask (IBM) has demonstrated substantial intelligibility improvements for both normal- and impaired-hearing listeners. However, this approach exploits oracle knowledge of the target and interferer signals to preserve only the time-frequency regions that are target-dominated. Single-channel noise suppression algorithms trying to approximate the IBM using locally estimated signal-to-noise ratios without oracle knowledge have had limited success. Thought of in another way, the

IBM exploits the disjoint placement of the target and interferer in time and frequency to create a time-frequency signal representation that is more sparse (i.e., has fewer non-zeros). In recent work (in preparation for ICASSP 2013), we have introduced a novel time-frequency masking algorithm based on a sparse approximation algorithm from the signal processing literature. However, the algorithm employs a non-causal estimator. The present work introduces an improved de-noising algorithm that uses more realistic frame-based (causal) computations to estimate a binary mask.

1pSPc16. Objective and subjective evaluation of complementary Wiener filter for speech dereverberation. Kento Ohtani, Tatsuya Komatsu (Grad. School of Information Sci., Nagoya Univ., Furo-cho,Chikusa-ku, Nagoya, Aichi 464-8601, Japan, ohtani.kento@g.sp.m.is.nagoya-u.ac.jp), Kazunobu Kondo (Corporate Res. & Development Ctr., Yamaha Corp., Nagoya, Japan), Takanori Nishino (Information Eng., Grad. School of Eng., Mie Univ., Nagoya, Japan), and Kazuya Takeda (Grad. School of Information Sci., Nagoya Univ., Nagoya, Japan)

Acoustic distortion caused by reverberation can degrade speech quality and performance of speech-based systems. Several dereverberation techniques have been proposed in the literature. For example, a dereverberation method using a complementary Wiener filter can suppress late reverberation with few computational resources. As a method for dereverberation, the method using a complementary Wiener filter has been proposed, and for the exponential decay impulse response model, it is shown theoretically that we can suppress reverberation with few computational resources. In this report, we approximate expectation of the power spectrum. which is necessary to calculate a complementary Wiener filter as exponential moving average. We conducted dereverberation experiments using actual environment room impulse response. The results of the objective evaluation show that the suppression performances of the actual environment room impulse response can approximate from the results of the exponential decay impulse response model. Additionally, we investigated the relationship between the results of objective evaluation and the results of subjective evaluation. In a small reverberation environment, we can see strong correlation between the results of objective and subjective evaluation.

1pSPc17. Evaluation of human-phonatory radiation characteristics with a polyhedron loudspeaker. Naoki Yoshimoto, Kota Nakano, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Noji-higashi, Kusatsu 525-8577, Japan, is046081@ed.ritsumei.ac.jp)

Spoken dialog systems have been studied for car navigation systems and voice search systems. For evaluating these systems, a loudspeaker is used instead of a human because these systems require various kinds of speech samples. However, the sounds radiated by loudspeaker cannot reproduce human-phonatory radiation characteristics. Therefore, the mouth simulator is utilized to reproduce human-phonatory radiation characteristics. Although it is based on the average mouth shapes, shapes of mouth are different among phonemes. Therefore, due to the hardware structure, it cannot accurately reproduce various human-phonatory radiation characteristics affected by shapes of mouth. In this study, we developed a polyhedron loudspeaker to solve this problem. It consists of 11 loudspeakers, which are independently controlled. Controlling eleven loudspeakers makes it possible to reproduce desired radiation characteristics. By utilizing this method, we try to reproduce human-phonatory radiation characteristics of Japanese five vowels and typical consonants with digital filters which were adaptively designed. We carried out an evaluation experiment in various measuring points to verify the effectiveness of the proposed method. As a result, it was confirmed that human-phonatory radiation characteristics with the proposed method could be accurately approximated compared with the conventional mouth simulator.

1pSPc18. Optimized hermetic transform beam-forming of acoustic arrays via cascaded spatial filter arrangements derived using a chimerical evolutionary genetic algorithm. Harvey C. Woodsum and Christopher M. Woodsum (Sci. and Technol., Nergetic System Dynam., LLC, 3700 N. Charles St., Unit 903, Baltimore, Maryland 21218, cwoodsu1@jhu.edu)

Hermetic transforms are complex matrices, having particular mathematical properties, that have recently been introduced to the field of acoustic array signal processing. Cascade sequences of Hermetic transform matrices have been shown to have direct utility in accomplishing spatial filtering and beam-forming of data from oversampled arrays. The present work details the adaptation of techniques previously shown to be successful in the processing of radio-wave phased-array antenna systems [Woodsum *et al.*, *16th International Conference on Cognitive and Neural Systems* (2012)] to the processing of sampled digital data from acoustic arrays. As in our earlier work, the use of a Chimerical, Evolutionary, Genetic Algorithm, having a "feature seeking" fitness function, is retained, for deriving optimal multiplicative arrangements of non-commuting elemental transform matrices. Each elemental matrix represents a spatial "pole" or "zero," and cascaded arrangements of these are utilized to create a desired spatial pattern response for the array. In terms of acoustic reception, the technique is especially successful in dealing with null placement in order to mitigate large numbers of interfering signals, and in achieving super-resolution beams for arrays that are "acoustically small." Experimental results are compared to theoretical predictions of performance.

1pSPc19. A study on acoustic imaging based on beamformer to range spectra in the phase interference method. Ryota Miyake, Kohei Hayashida, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is0041ki@ed.ritsumei.ac.jp)

Information on the distance to the target is very important to achieve the practical use of hands-free speech interfaces and nursing-care robots. Many distance measurement methods, which use the time-of-flight (TOF) of a reflected wave measured with reference to the transmitted wave, have been proposed. However, these methods cannot measure short distances because the transmitted wave, which has not attenuated sufficiently at the time of a reflected wave reception, suppresses reflected waves for short distances. We previously proposed an acoustic distance measurement method based on interference between the transmitted and reflected waves, which can be used for distance measurement over a short range using single microphone. This method is referred to the phase interference method. It can estimate the distance to target, but cannot estimate the direction of target. In the present paper, therefore, we propose to achieve acoustic imaging with the phase interference method by using microphone-array instead of single microphone. More specifically, we apply the beamformer to the range spectra calculated from observed signals at each microphone of microphone-array to obtain the spatial information. Finally, we confirm the effectiveness of the proposed method through evaluation experiments in real environments.

1pSPc20. Investigations into the human pinna shapes on head-related transfer functions in the median plane. Hajime Komatsu, Kota Nakano, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is0016rv@ed.ritsumei.ac.jp)

The binaural reproduction system requires many accurate measurements of head-related transfer functions (HRTFs) to achieve the high-precision sound localization. However, the actual measurement of HRTFs has a heavy burden for subjects. To solve this problem, personalize HRTFs have been proposed. In the personalize HRTFs, the interaural level difference (ILD) and the interaural time difference (ITD) are utilized on the sound localization in the horizontal plane, and the spectral envelope of HRTFs is utilized on the sound localization in the median plane. In the present paper, we focus on the human pinna shapes as listener's anthropometric parameters on the sound localization in the median plane. In order to reveal the effect of human pinna shapes on HRTFs in the median plane, we investigate the relationship between human pinna shapes and the spectrum envelope of HRTFs. More specifically, we crafted the dummy pinna for the dummy head. Also, we investigated the spectrum envelope of HRTFs in various shape conditions of the dummy pinna in the median plane. As a result of investigations, we confirmed correspondence relationship between human pinna shapes and the spectrum envelope of HRTFs.

1pSPc21. Acoustic echo cancelation in discrete Fourier transform domain based on adaptive combination of adaptive filters. Luis A. Azpicueta-Ruiz, Anibal Figueiras-Vidal, and Jeronimo Arenas-Garcia (Dept. of Signal Theory and Commun., Universidad Carlos III de Madrid, Av Universidad 30, Leganes, Madrid 28911, Spain, azpicueta@tsc.uc3m.es)

Acoustic echo cancelers (AECs) are vital to many of communication systems, including hands-free telephone and videoconference, among others. Recently, adaptive combination of adaptive filters has been presented

as an easy but effective method to improve its performance, alleviating different compromises inherent to adaptive filters (responsible to identify the room impulse response). The most important tradeoff is related with the selection of the step size, which involves speed of convergence and residual echo. However, AECs require long adaptive filters, forcing to employ frequency-domain techniques to reduce the computational load and to accelerate the algorithms convergence when colored inputs—such speech—are presents. In this paper, we present an AEC based on combination of filters in discrete Fourier transform domain. Considering that both the input signal and the cancelation scenario make the behavior of adaptive filters is frequency dependent, our proposal exploits the combination capabilities employing different mixing parameters to separately combine independent spectral regions of two frequency-domain adaptive filters with different step sizes. In this way, our scheme outperforms recent algorithms where only a single combining parameter mixes the overall outputs of two frequency-domain adaptive filters. These advantages are illustrated by means of realistic experiments.

1pSPc22. Adaptive active control of free space acoustic noise. Iman Tabatabaei Ardekani and Waleed H. Abdulla (Elec. and Comput. Eng., The Univ. of Auckland, Private Bag 92019, Auckland CBD, Auckland, Auckland 1142, New Zealand, i.ardekani@auckland.ac.nz)

This paper concerns adaptive active control of acoustic noise in free space. Traditional adaptive active noise control algorithms are efficient in acoustic ducts; however, they are very unstable and sensitive when being used in free space. An efficient adaptive active noise control algorithm for free space noise is derived based on a root locus analysis on the adaptation process performed in adaptive active noise control. The traditional algorithm and the proposed algorithm are fully implemented by using a high performance embedded controller. The controller is then used for active control of acoustic noise in a duct and, also, in free space. Different experiments show that the traditional active noise control algorithm is not stable when the setup is used in free space. However, the proposed algorithm is stable and converges at a high convergence rate until reaching steady state conditions.

1pSPc23. A detection of danger sounds based on variable-state hidden Markov models. Asako Okamoto, Kohei Hayashida, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Science and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is0009sv@ed.ritsumei.ac.jp)

To detect hazardous situations with danger sounds, the acoustic surveillance system is an ideal candidate. The conventional systems recognize environmental sounds with hidden Markov model (HMM) in order to detect danger sounds. It is however difficult to accurately recognize the environmental sounds, because the optimum HMM parameters for environmental sounds have not been identified. It is important factor for accurately recognizing them to ideally determine the number of states, one of the HMM parameter. On the other hand, environmental sounds which include danger sounds have a wider characteristic as the structure, the complexity, the length, etc. The variable states should be therefore an optimum HMM structure to detect the danger sounds. We thus propose the danger sound detection based on variable-state HMMs corresponding to a number of inflection points with the delta power of environmental sounds. We first investigate the recognition performance of environmental sounds including danger sounds with various states of HMM. We then investigate the relationship between the recognition performance and a number of inflection points with the delta power of various environmental sounds. As a result of evaluation experiments, we designed an optimum variable-state HMM for environmental sounds and confirmed the effectiveness of the proposed method.

1pSPc24. Sound source measurement of magnetic resonance imaging driving sound for feedforward active noise control system. Shohei Nakayama, Kenji Muto (Elec. Eng. and Comput. Sci., Shibaura Inst. of Technol., 3-7-5, toyosu, kouto-ku, Tokyo 135-8548, Japan, ma12077@shibaura-it.ac.jp), Kazuo Yagi (Dept. of Radiol. Sci., Tokyo Metropolitan Univ., Tokyo, Japan), and Guoyue Chen (Dept. of Electron. and Information Systems, Akita Prefectural Univ., Akita, Japan)

We proposed the active noise control (ANC) system reduce the loud MRI sound. It was important for performance improvement of the system. Therefore, we estimated the sound source of MRI driving sound. The position of the sound source of MRI driving sound was between the center and the edge in the gantry of MRI equipment. MRI equipment is important for the medical inspection, which gets the tomography of the body without x-ray. The patient of the MRI inspection needs to use the ear protector because the MRI equipment generated the loud sound, which the sound pressure level was around 100 dB. Here, our study was to make good acoustical environment using the ANC system for the MRI patient. The ANC system used the feedforward type because the MRI driving sound have unsteady pulsed sound. We made the ANC system using non-magnetic devices, ear protectors, and optical microphones. Because the MRI room had high magnetic environment. We measured the sound source of MRI driving sound to set the reference microphone. In this case, we showed the reduction effect of the ANC system of the sound by the computer simulation. As a result, the system reduced the MRI driving sound by around 50 dB.

1pSPc25. Parametric loudspeaker for speech signal based on the combination of amplitude and frequency modulations. Toru Iwasaki, Daisuke Ikefuji, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is0005ri@ed.ritsumei.ac.jp)

A parametric loudspeaker has been used for audio guidance to a specific area because it has a sharper directivity compared with the conventional electrodynamic loudspeakers. The parametric loudspeaker emits an ultrasound as a carrier wave, which is modulated with an audio signal and has large-amplitude. An audible sound is reproduced by the modulated ultrasound with large-amplitude distorted by the nonlinearity on the air. The conventional modulations have been proposed as the amplitude modulation and the frequency modulation. In the sound quality, the amplitude modulation is superior to the frequency modulation. However, in the sound pressure level, the frequency modulation is superior to the amplitude modulation. In the present paper, we especially focus on that the parametric loudspeaker will emit the speech signals for the audio guidance. Therefore, we propose new modulation method based on the combination of amplitude and frequency modulations, which are specialized for speech signals. More specifically, we apply the amplitude and frequency modulations to the divided frequency bands of speech signals, respectively. In order to confirm the effectiveness of the proposed method, we carried out evaluation experiments. As a result of evaluation experiments, we confirmed the effectiveness of the proposed method compared with conventional methods.

1pSPc26. Chain architecture: An efficient hardware solution for a large microphone array system. Dmitry N. Zotkin (Inst. for Adv. Comput. Studies (UMIACS), Univ. of Maryland, College Park, MD), Ross Adelman (Dept. of Comput. Sci., Univ. of Maryland, College Park, MD), Adam E. O'Donovan (VisiSonics Corp., College Park, MD), and Ramani Duraiswami (Inst. for Adv. Comput. Studies (UMIACS), Univ. of Maryland, Dept. of Comput. Sci., Univ. of Maryland, College Park, MD 20770, ramani@cs.umd.edu)

A typical microphone array system consists of a number of microphones connected to the digitization hardware and central processing unit in a parallel fashion. Such radial, hub-and-spoke architecture has multiple points of failure, suffers from electromagnetic interference, and does not scale well. In this paper, an alternative, chain-like architecture is described. In such setup, the microphones in a system are organized in a single chain. Each individual microphone board has an ADC chip and is connected to the previous and to the next microphones in the chain with short multi-wire cables carrying digital signals. A buffer board at the end of the chain converts the digital data stream into the industry-standard USB 2.0 format. In this way, the individual microphone board becomes the building block for quick and easy arbitrary-configuration microphone array assembly with minimal amount of wiring involved. A hardware implementation of the chain architecture was developed and is described. Accompanying drivers and software allow the user to perform on-the-fly data acquisition and processing in C and in MATLAB. As an example, a 64-microphone array was built, and several source localization and beamforming algorithms were implemented in MATLAB. Experimental results using the data gathered from the array are presented.

1pSPc27. A design of audio spot based on separating emission of the carrier and sideband waves. Tadashi Matsui, Daisuke Ikefuji, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is0039fx@ed.ritsumei.ac.jp)

Parametric loudspeaker, which utilizes an ultrasonic of non-linear interaction, is developed to achieve audio spot. The parametric loudspeaker has sharper directivity, but reflections and intercepts by emitted sounds become severe problems. This is because reflections and intercepts lead to an invasion of privacy, and become noise to other listeners except a target listener. Principle of the parametric loudspeaker can formulate as non-linear interaction of carrier and sideband waves in emitted ultrasonic sounds on air. This suggests that we can design audio spot by individually emitting the carrier and sideband waves. In the present paper, therefore, we propose the design method of audio spot with the separating emission of the carrier and sideband waves. More specifically, the audible sound is demodulated at an area where the carrier and sideband waves individually emitted from each parametric loudspeaker are overlapped. We carried out evaluation experiments to measure sound pressure level (SPL) of demodulated audible sound. In addition, we evaluated the speech articulation of the demodulated audible sound with the proposed method. As a result of evaluation experiments, we confirmed the effectiveness of the proposed method.

MONDAY AFTERNOON, 3 JUNE 2013 — 511AD, 1:00 P.M. TO 4:40 P.M.

Session 1pUW

Underwater Acoustics: Seabed Scattering: Measurements and Mechanisms II

Charles W. Holland, Cochair
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Contributed Papers

1:00

1pUW1. A study of the reflection coefficients and backscattering effects of one-dimensional rough poroelastic surfaces using the finite element method. Anthony L. Bonomo, Marcia J. Isakson, and Nicholas Chotiros (Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

Acoustic reflection and scattering effects of one-dimensional rough poroelastic surfaces are studied using the finite element method. The poroelastic sediment layer is modeled following the classical work of Biot as extended by Stoll, which assumes that two attenuating compressional waves and one attenuating shear wave propagate in the sediment. The rough surfaces are generated using power-law type spectra and the incident wave used is a Gaussian tapered plane wave. This work seeks to assess how the reflection coefficients and backscattering effects of a poroelastic bottom vary as a function of frequency, roughness, and sediment type. Special consideration is given to the mesh required to accurately resolve the effects of the slow compressional and shear waves, which often have wave speeds slower than the fast compressional wave by an order of magnitude or more. [Work sponsored by the Office of Naval Research, Ocean Acoustics.]

1:20

1pUW2. High frequency backscattering from sandy sediments: single or multiple scattering. Nicholas P. Chotiros and Marcia J. Isakson (Appl. Res. Lab., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, chotiros@arlut.utexas.edu)

As the sand grain size approaches the acoustic wavelength, the underwater backscattering strength increases rapidly. Laboratory measurements indicate that the shallow-grazing angle backscattering strength increases as the third power of the normalized grain diameter. In this regime, it has been shown that the attenuation of the sound in the sand increases as the fourth power of frequency and the speed of sound decreases with increasing frequency. The most likely explanation for the attenuation and speed dispersion is multiple scattering [Schwartz and Plona, J. Appl. Phys. **55** (1984) and Kimura, J. Acoust. Soc. Am. Express Lett. **129** (2011)]. Single and multiple scattering theories will be applied to the backscattering problem with the purpose of determining if it is a single or multiple scattering process. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

1:40

1pUW3. The impact of finite ensonified area on the scattering cross section. Derek R. Olson (Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, dro131@psu.edu) and Anthony P. Lyons (Appl. Res. Lab., The Penn State Univ., University Park, PA)

The definition of the scattering cross section for the pressure field scattered by a rough interface uses the underlying assumption that the ensonified area does not affect its shape. This assumption holds so long as the incident field is a good approximation of a plane wave, which situation occurs when the ensonified area is large compared to a wavelength. In the opposite situation, when the ensonified dimensions approaches a wavelength, the incident plane wave assumption does not hold and the cross section can depart from modeled behavior. This research uses the perturbation approximation to derive a model for the low grazing-angle behavior of the scattering cross section. The primary results are (1) the appearance of an additive Lambertian term and (2) a separation of scales imposed by the acoustic resolution. This separation can then be used as a criterion for the application of the composite roughness model. Model results are checked against direct numerical solution of the governing integral equations. Implications for inversion of seafloor parameters based on acoustic scattering experiments is discussed.

2:00

1pUW4. In situ calibration of seafloor acoustic backscatter data from swath mapping sonars. Christian de Moustier (HLS Res., Inc., 3366 N. Torrey Pines Court, Ste. 310, La Jolla, CA 92037, cpm@hlsresearch.com) and Barbara J. Kraft (HLS Research, Inc., Barrington, New Hampshire)

Geometric and radiometric corrections are necessary to convert raw seafloor acoustic backscatter data to imagery in which the intensity variations are due to the geoacoustical properties of the bottom and the angular dependence of the backscattering process. Radiometric corrections include removal of the combined effects of the sonar's transmit and receive beam patterns, and removal of the area ensonified. To this end, a method is presented to estimate the transmit/receive beam pattern of a multibeam swath mapping sonar using seafloor acoustic backscatter data collected over an entire survey area. This involves estimating the scattering area at each sounding based on local seafloor slopes. For a given sonar installation, results show that such beam pattern estimates remain stable to within $\pm$ 0.5 dB over different survey areas.

2:20

1pUW5. Non-invasive characterization of fluid mud from scalar and vector noise fields due to a small boat. Jean-Pierre Hermand and Qunyan Ren (Université libre de Bruxelles (U.L.B.), av. F.D. Roosevelt 50 - CP 165/57, B-1050 Brussels, Université libre de Bruxelles (U.L.B.), Brussels 1050, Belgium, jhermand@ulb.ac.be)

The passive geoacoustic characterization of sediment using the ratio between pressure and vector field that are measured by an easily deployable system is discussed. The ratio is very sensitive to environmental properties but independent of unknown source spectrogram, e.g., boat noise that exhibits its complex spectral shape. Noise data sets due to different runs with a small boat were recorded on two closely adjacent hydrophones offshore the Amazon Rio mouth and processed by a nonlinear inversion scheme. Global optimization based on genetic algorithms provides the geoacoustic parameters of the fluid mud and underlying mud sediment, and the marginal posterior probabilities. Good consistency among the respective inversion results demonstrates the feasibility of the method under far from ideal conditions for environmental characterization, in term of unknown range, source ship navigation data, unknown source spectra, uncertain receiver depth, tilt, etc.

2:40

1pUW6. Resolution analysis of the inverse problem for geoacoustic experiment design. Andrew A. Ganse (Appl. Phys. Lab., Univ. of WA, 1013 NE 40th St., Seattle, WA 98105, aganse@apl.washington.edu)

This work explores effects of experiment geometry and array configuration on the resolving power of a continuous ocean bottom geoacoustic inverse problem in a shallow water environment. The uncertainty and resolution of this problem, in which ocean bottom P-wave velocities as a function of depth are estimated from noisy acoustic pressure waveforms received on vertical and horizontal line arrays in the water, can be investigated before the experiment is conducted, allowing one to improve or optimize the problem parameters to best configure an experiment during its planning phase. In this work, the resolution results of complete synthetic geoacoustic inversions at varying geometries and array configurations are compared with resolution results at various candidate seabottom profiles, initially using standard techniques of linearized inverse theory. Singular value decomposition is used to interpret the tie between geometry and regularization in the inverse problem, which directly affects the resolution. Then, additional comparisons and analysis address the nonlinearity of the problem, which causes a dependence of the resolution results on the bottom profile being solved for—which is unknown, and Monte Carlo analysis is used to show where the linearity approximation breaks down in the resolution results. [Work partially funded by ONR.]

3:00

1pUW7. Inversion of seabed acoustic parameters in shallow water using the warping transform. Juan Zeng (Inst. of Acoust., CAS, No.21 BeiSi-Huan XiLu, Bei Jing, China, Bei Jing 100190, China, zengjuan01@yahoo.com.cn), N.Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Li Ma, and Yan Chen (Inst. of Acoust., CAS, Bei Jing, China)

In this paper, a method is described for inverting geoacoustic parameters of the seabed from short range field data recorded by single hydrophone. The original data in time domain are processed by a warping operator at first, and then, the dispersion curve and the mode amplitude ratios are extracted separately from the warped data. The velocity and the density in the bottom are inverted from the dispersion curve, and the attenuation from the mode amplitude ratios, respectively. The performance of the method is examined using simulated data and then experimental data from the North Sea of China. The source used in the experiment was a small explosive charge that provided good signal to noise ratio over the frequency band from 200 Hz to 1 kHz. The depth of the water was about 30 m, and the water sound speed was nearly constant with depth. The seabed geoacoustic parameters are inverted from the data received at different ranges from 2 to 14 km. The results from the different ranges are consistent with a simple half space model of the bottom. The seabed velocity is about 1600 m/s.

3:20

1pUW8. Determination of grain size distribution in water-saturated granular medium using p-wave attenuation dispersion. Haesang Yang, Keunhwa Lee, and Woojae Seong (Dept. of Ocean Eng., Seoul National Univ., Bd. 34, Rm. 306, Underwater Acoust. Lab.,1, Gwanak-ro, Gwanak-gu, Seoul 151-744, South Korea, coupon3@snu.ac.kr)

P-wave attenuation in the water-saturated granular medium depends on both the frequency and the grain size. In this study, the use of the attenuation dispersion for the determination of grain size distribution in the water-saturated granular medium is discussed. For the dense granular medium, mathematical model considering multiple scattering is used for regression algorithm by fitting model predictions to the measured attenuation data. Inversion of grain size distribution is carried out numerically, and the results are discussed and compared to measured data for the water-saturated glass beads with unimodal and bimodal distributions.

Invited Paper

3:40

1pUW9. Rough interface acoustic scattering from layered sediments using finite elements. Marcia J. Isakson and Nicholas Chotiros (Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Quantifying acoustic scattering from rough interfaces is important for reverberation modeling, acoustic sediment characterization, and propagation modeling. Most models of interface scattering on layered surfaces rely on approximations to the Helmholtz/Kirchhoff integral. These models generally make such assumptions as neglecting the local angle for reflection and disregarding multiple scattering between rough layers. In this study, a mixed boundary element/finite element model is used to calculate rough interface scattering from sediment layers including elastic solids. The finite element method, based on the Helmholtz equation, is exact within the limits of the discretization density; reflections are calculated locally and all orders of scattering among layers are included. Using this model, bottom loss and backscattering predictions will be calculated for several cases. These predictions will be compared with more traditional scattering models and finally to a full finite element model of a point source reflection from the same sediment bottom. [Work supported by ONR, Ocean Acoustics.]

Contributed Papers

4:00

1pUW10. Geo-acoustic parameter estimation using a multistep inversion technique based on normal mode method. Lin Wan, Mohsen Badiey (College of Earth, Ocean,and Environment, Univ. of Delaware, 003 Robinson Hall, Newark, DE 19716, wan@udel.edu), and David Knobles (Appl. Res. Lab., The Univ. of Texas at Austin, Austin, TX)

The geo-acoustic parameters are of great importance in determining how the sea bottom affects sound propagation in the ocean. Many inversion techniques have been developed to estimate geo-acoustic parameters. One-step inversion algorithms using a cost function defined only by energy loss may not result in a unique solution of geo-acoustic inversion problem because of the correlation between seabed sound speed and attenuation. The present paper utilizes different characteristics of normal modes, including modal dispersion, modal attenuation, and modal based spatial coherence, to define appropriate cost functions in a multistep inversion algorithm for geo-acoustic parameter estimation. This inversion scheme is applied to the long-range broadband acoustic data obtained from L-shaped arrays in the Shallow Water 2006 experiment. The seabed sound speed and attenuation are estimated by minimizing the cost function at each step. The results show a nonlinear frequency dependence of attenuation, which is similar to the seabed attenuation derived from measured time series and transmission loss data at the same experimental site [Knobles *et al*., J. Acoust. Soc. Am. **124**, 2008]. The uncertainties caused by the range dependent water column variability and bathymetry are discussed. [Work supported by ONR322OA.]

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1pUW11. Electrochemical basis of the card-house model of mud and its acoustical implications. Joseph O. Fayton (Mathematics, Rensselaer Polytechnic Inst., Amos Eaton, 110 Eighth St., Troy, NY 12180, faytoj@rpi.edu), Allan D. Pierce (Mech. Eng., Boston Univ., Boston, MA), and William L. Siegmann (Mathematics, Rensselaer Polytechnic Inst., Troy, NY)

A basic mud model contains thin mineral (kaolinite and smectite) particles, roughly hexagonally shaped platelets, with diameters typically 1 micron. Isomorphous substitution causes each platelet to carry a net negative charge per unit area. Because the ions in the surrounding water respond so that there is a net positive charge on both sides of the platelet, each platelet is modeled as a sheet of longitudinal electric quadrupoles aligned perpendicular to the surface. The electrical interaction between platelets is responsible for the card-house structure, whereby the edge of one platelet touches a central line along the surface of another platelet, with the platelets being at right angles to each other. When the perpendicular arrangement is perturbed, a restoring torque attempts to return the platelets to their original state. Electrostatic analysis is used to explain why the restoring torque is formally singular at the joining line when one platelet is slightly tilted from perpendicular. The singular behavior appears to arise when the corners of one platelet touch the edge of another. This singularity requires imposition of the cantilever boundary condition in order to consider the shear resistance of mud, with each platelet bending as an elastic plate. [Research supported by SMART Fellowship and ONR.]